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MODULATION AND CODING STUDY FOR THE ADVANCED NARROWBAND DIGITAL--ETC(U)

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Modulation and Coding Study for the Advanced Narrowband Digital Voice Terminal

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Systems Integration and Instrumentation Branch
Communications Sciences Division

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NAVAL RESEARCH LABORATORY
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A study was performed of modulation and coding techniques to support development of specifications for an advanced narrowband digital voice terminal. Emphasis was on the development of a signal design for HF radio transmissions. The results indicate that transmission channel effects can be significantly reduced by the application of forward acting error correction coding to the most sensitive bits of the digital voice signal. A recommendation is made to use the Golay (24, 12) code, with soft decision decoding, to protect 44 percent of each frame of data generated by a linear predictive encoder. This would result in a transmission rate of 3466.6 bps for an input of 2400 bps using multi-tone four phase differentially coherent phase shift keying.

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MODULATION AND CODING STUDY FOR THE ADVANCED NARROWBAND DIGITAL VOICE TERMINAL

1.0 INTRODUCTION

The Department of Defense is engaged in the development of an Advanced Narrowband Digital Voice Terminal (ANDVT). This system is to provide secure voice capability to military users who must operate over bandwidth limited circuits. The Joint Tactical Communication Office (TRI-TAC) is the executive agent for the ANDVT program. The Department of the Navy is responsible for the development and initial production of the non-cryptographic components. The Director, National Security Agency (NSA) has similar responsibility for the cryptographic components of ANDVT.

The TRI-TAC work plan for ANDVT identified six major tasks to be accomplished prior to procurement of advanced development models. These tasks were:

- Task 1 - Definition of System Requirements
- Task 2 - System Engineering Analysis
- Task 3 - Definition of Required Terminal Equipment
- Task 4 - Definition of Required Cryptographic Equipment
- Task 5 - Integrated Logistic Support Analysis
- Task 6 - Documentation.

The Naval Electronic Systems Command (Code 310) and the Naval Research Laboratory have been responsible for Task 3. That task has been further divided into five subtasks, as follows:

- Subtask 3.1 - Tradeoff Analysis of System Performance Requirements
- Subtask 3.2 - Assessment of QUINTRELL Processor Designs (QUINTRELL refers to two separate signal processor designs that were developed for NSA in support of their Executive Secure Voice Network (ESVN) program)
- Subtask 3.3 - Analyze and Define Terminal Architecture
- Subtask 3.4 - Refine Voice Processing Algorithms/Audio Processing Functions

Note: Manuscript submitted June 23, 1978.

- Subtask 3.5 - Evaluation of Modem Implementation Alternatives.

It is the purpose of this report to summarize the work performed under Subtask 3.5 and to describe the characteristics of the modems being specified for ANDVT.

2.0 OBJECTIVES

In the program plan for the development of the specifications for ANDVT, a task was established for a modem study (Task 3.5). This task has five broadly defined objectives. They were:

- Identification of the requirements for modems in ANDVT.
- Definition of the characteristics of the modems needed to satisfy the requirements.
- Definition of the intra-terminal and inter-terminal control and signaling problems relative to the modem designs.
- Assessment of the impact of the modem designs on the overall goals established for the terminal, relative to size, weight, power, cost, and reliability.
- Recommendations for technical specifications of the proposed modems.

The approaches taken to satisfy the objectives of the study were as follows:

- To itemize and assess the modem requirements that were stated in the Joint Operational Requirements and the TRI-TAC requirements documents.
- To develop a candidate HF modem specification to represent the best estimation of the modem characteristics necessary to satisfy the requirements.
- To develop a candidate functional description of the terminal to meet the stated requirements.
- To use the candidate modem specification and the functional description to size and cost out the modem functions for different approaches to the terminal architecture.
- To use the candidate modem specification and the functional description to identify the intra-terminal and inter-terminal control and signaling problems for each possible mode of operation.

. To determine the level of performance achievable with an integrated Linear Predictive Encoder (LPC) coder/HF modem on simulated HF channels. To compare this with an uncoded system.

. To determine the signal design for a robust modem preamble, for the transmission of the key generator preamble and for the transmission of digital messages rather than digitized voice.

The sizing and costing out of the modem functions, for various software and hardware implementations, were carried out as part of the contract work undertaken to look at the terminal architecture under Subtask 3.3. Contracts with the developers of the QUINTRELL processors included the development of software programs for a candidate HF modem for half duplex digital voice operation. A supplementary contract effort was aimed at the improvement of the HF modem signal design over that defined as the candidate modem. Work relative to the development of a wireline modem specification was confined to tracking the NSA development of a software wireline modem for their ESVN program, and a survey of commercially available devices.

3.0 REQUIREMENTS

The user's requirements that have an impact on the modem signal design may be grouped into the following general areas:

. Type of data
. Information rate
. Performance requirements
. Channel characteristics
. RF equipment characteristics
. Compatibility requirements
. Operational procedures and constraints
. Implementation considerations.

The known requirements for ANDVT are documented in the Joint Operational Requirement for ANDVT [1] and in the TRI-TAC Task 1 report [2]. Information regarding all of the above items were not found in the referenced documents. A summary of the modem requirements is presented:

3.1 Wireline Requirements

. One digital voice information rate. The rate was not specified.

. A digital message I/O was stated as an optional requirement. Rates were not given. The exception to this was the NATO requirement.

- The mode of operation was specified as full duplex, 4-wire. Again, the exception to this was the NATO requirement.

- Operation would be over voice grade lines.

- Standard nonencrypted analog loop signaling is required. A dial tone unit is required to be an integral part of the terminal.

- The terminal must be able to operate over switched lines.

- The wireline modem must be compatible with ESVN.

- The terminal must have the ability to transmit a DTMF signal in-band to ESVN after the circuit is established.

- The modem must be compatible with a L1 type of interface as defined in reference [2].

The above requirements dictate that the wireline modem for the Defense Communications System (DCS) ANDVT be compatible with ESVN. That establishes the modulation to be compatible with Federal Standard 1005. This is the federal government's specification for the Western Electric 201B modem. The modulation is four-phase DPSK of a single tone at a rate of 1200 baud. The signaling is selected to be compatible with AUTOVON DTMF. Thus, the decisions regarding the wireline modem that are left to be made within the ANDVT program are essentially reduced to only the issues of how to implement the selected modem. The alternatives are:

- Hardware vs software vs hybrid
- Internal vs external
- Adaptive equalization vs selectable equalization

3.2 HF Requirements

- Only half duplex operation is required.

- Operation in either nets or point-to-point.

- Digital message input at rates of 600, 1200 and 2400 bps. This requirement was confined to users operating from vehicles using portable digital message entry devices (DMED).

- RF equipments were identified as "standard tactical RF equipment".

- The interfaces are identified as types R1, R2 and L1 as defined in reference [2].

- The modem has a requirement to pass analog clear voice when operating in that mode.

- There is a special requirement for the HF modem to be able to pass a warning signal over the HF channel when the ANDVT is patched to the analog output of a digital nonsecure voice terminal.

The HF requirements are especially lacking in a definition of the user's performance requirements for data operation, operational constraints and identification of RF equipment characteristics. There were no requirements to be compatible with any presently designed modems.

3.3 VHF Requirements

- The Marines have a requirement to operate ANDVT into a four channel FDM with FM radio equipment. These RF links are essentially point-to-point, full duplex operation, although the ANDVT could function in a half duplex mode. The interface is type R1.

3.4 UHF Requirements

- The Air Force has a requirement to use ANDVT in a voice relay circuit. This operates full duplex, air-to-air over narrowband DSB-AM radio equipment.

4.0 ASSUMPTIONS

A number of assumptions were made in the conduction of the study relative to specifying an HF modem design. These assumptions served the purpose of defining the types of equipments with which the modem should interface, the operational constraints that affect the modem design and the range of channel conditions under which the modem must function. Some of the assumptions represented a distinct enlargement on the requirements for the HF modem. This step was taken in order to size, cost out and determine the architectural problems relative to a terminal that would satisfy more than the minimum requirements stated in the Tri-Tac document. The following is a listing of the assumptions made regarding the types of equipments with which the modem should interface:

- All HF radios to be used with ANDVT should be fully synthesized equipments capable of supporting DPSK signals.

- Optimistically, all HF radios would be designed to specifications at least as stringent as those classified as "High Performance Radios" in MIL-STD 188C; but, if possible, the modem should function with radios designed to specifications classified as "Low Performance Radios" in MIL-188C. The principal differences between these two classifications are listed in Table 1. Note that it is the frequency tolerance aspects of the low performance radios that have a strong impact on the doppler correction requirements of the modem. Likewise, it is the envelope delay distortion of the low performance equipments which impact on the bandwidth aspects of the modem signal design. The values listed in Table 1 are for each transmitter and receiver, respectively. Thus, they must be doubled to represent the combined effects of a communication set.

. Anti-jam capability is not a requirement for the ANDVT HF modem.

. The principal mode for ANDVT shall be half duplex digital voice with nondiversity operation. But, there may be some need for full duplex operation as well as some need for either out-of-band, space, or polarization diversity operation. These assumed requirements have a large effect on possible terminal architectures. The envisioned need for full duplex operation centers chiefly on providing the user with a back-to-back checkout capability for the terminal, and providing a fall-back mode of operation when interfacing to a full duplex wireline terminal. A full duplex to half duplex interface, as described in the Tri-Tac requirements, has never been implemented. It represents a definite risk. The assumption that there may be a need for diversity operation, that was not identified in the requirements, is a recognition of the difficulty that can be encountered on some HF channels. The Tri-Tac requirements were chiefly the response of analog users wanting to get a new secure voice capability. As analog users, they operate non-diversity by necessity. With a digital system, diversity operation is an option; at the same time, it may be a necessity in order to get satisfactory performance. This is especially true for digital message applications at 2400 bps.

. There were no stated requirements for the HF modem to be compatible with any other modem. Therefore, it was assumed that a new modem design could be considered that would optimize performance of the secure voice terminal.

. NSA plans to design a new key generator module for ANDVT. Because it will be a new design, it was assumed that the HF modem in the terminal could control the coding of the key generator preamble to protect it against transmission errors. This would lessen the amount of circuitry in the module, but increase the complexity of the HF modem.

There were a minimum of stated operational conditions in the ANDVT requirements. For this study, the following were assumed:

- The objective for the total length of the modem preamble and key generator preamble should be approximately one second. The maximum length should not exceed 100 frame times or 2.25 seconds.

- If it is found beneficial to time spread the HF signal over multiple frames to lessen the effects of bursts, that spreading should not exceed 10 frame periods or 0.225 seconds.

- The maximum permissible frequency offset is ± 75 Hz. This frequency error being the sum of all errors due to equipments, channel doppler, and doppler offsets due to the velocity of aircraft. The assumed requirement of ± 75 Hz was obtained from the present MIL-STD 188C. This limits operation to conditions when the relative velocity between users is approximately MACH 2 at 30 MHz. An implication here is that low performance radio equipments probably cannot be used with high velocity aircraft.

- The maximum doppler tracking rate requirement was assumed to be 3.5 Hz per second. This is the same as stated in MIL-STD 188C. This limit was assumed to have been established by a requirement to operate with high velocity aircraft. The exact value to specify is a function of both the rate of change in velocity and the operating frequency.

- The requirement to provide a secure voice capability over a half duplex, point-to-point HF link would permit consideration of a special mode of operation to lessen the turn-around time. This mode would require the ANDVT to have the equivalent of full duplex timing circuits and full duplex COMSEC module. Aircraft should not be considered as possible users of this mode of operation.

Some assumptions had to be made about the range of channel conditions over which the modem was expected to operate:

The beyond-line-of-sight HF transmission channel is the result of propagation of the transmitted wave between various layers of the ionosphere and the earth. It is a linear, time-varying channel. The variations in the channel produce multiplicative effects on the received signal. Combined with this is the additive noise that is either generated in the earth's atmosphere or is man-made. Each propagation from an ionospheric layer may be viewed as a complex time-varying signal to which can be attributed mean values associated with:

- Delay
- RMS amplitude
- Frequency offset
- Doppler spread

The combination of signals from different layers introduces the problems of multi-path and narrowband frequency selective fading. The ranges of differential delay and doppler spreading associated with non-auroral propagation paths have been experimentally examined by many researchers. Extreme values may be considered to be six milliseconds differential delay and five Hz doppler spread. More typical values that may be encountered are closer to two millisecond delay and two Hz of doppler spread.

5.0 STANDARD/SPECIAL-PURPOSE HF MODEM

According to the Tri-Tac requirements document, the principal use of the military ANDVT is for securing HF voice communication circuits. The HF use of ANDVT will be either to provide a new secure voice capability or to replace PARKHILL equipments. There are only a minimum number of identified HF users who plan to replace a present digital voice terminal with an ANDVT. Thus, the planned HF users of ANDVT are essentially analog users who will be switching to a digital system either to acquire a secure voice capability or to replace an analog scrambler.

Therefore, there is a need to procure a large number of HF modems to meet ANDVT requirements. Three general approaches were considered. They were:

- Utilize the present MIL-STD 188C 2400 bps HF modem specification.
- Propose some improvements to the present standard modem.
- Propose a new HF modem that is optimized for the ANDVT secure voice application.

5.1 Characteristics of Standard Modem

The MIL-STD 188C standard for a 2400 bps HF modem, for applications other than the Naval Tactical Data System, has the following general characteristics:

- 16 parallel data tones equally spaced 110 Hz apart between 935 Hz and 2585 Hz.
- 4-phase DPSK modulation.
- A signaling rate of 75 frames/second.
- A time guard of 4.24 milliseconds.
- Doppler tracking on a single unmodulated tone centered at 605 Hz. The amplitude of the doppler tone is 7 dB greater than the amplitude of each data tone.

- Slot synchronization tracking on an empty tone slot centered at 825 Hz.

- The modem preamble consists of 2 tones transmitted for 5 frame periods (66.67 milliseconds). One tone unmodulated at 605 Hz is used for doppler acquisition. The second tone at 1705 Hz is phase modulation 180 degrees at the signaling rate. It is used for synchronization.

5.2 Desired Characteristics for ANDVT Modem

The specification of a terminal that is optimized for operation in a half duplex, push-to-talk mode in an HF net is concerned with two general areas:

- A robust acquisition/synchronization scheme.

- Acceptable digital voice operation over a range of channel conditions comparable to analog voice.

The acquisition/synchronization of a half duplex HF terminal involves:

- Detection of signal presence.

- Doppler correction

- Modem synchronization.

- Key generator synchronization.

- Key generator initialization.

- Voice processor synchronization.

Three synchronization processes are involved when the modem, key generator and voice processor are treated as three separate functions. This is the approach used in some present HF digital voice systems. The synchronization problems in the terminal can be reduced from three separate processes to one process by designing the terminal to be frame synchronous. That is, the modem, key generator and voice processor would operate at a common frame rate. Then, acquisition of modem synchronization automatically establishes frame synchronization of the other functions in the terminal. The robustness of the entire acquisition/synchronization scheme is then established by the design of the modem preamble and the redundancy provided for the key generator preamble.

5.3 Sensitivity of LPC to Channel Errors

The LPC algorithm generates a digital signal that is a time division multiplex of several pulse code multiplexed narrowband signals. As a result, the individual bits in the data stream are of unequal weight. Some bits are far more sensitive to transmission errors than other bits. Listening tests indicate that the LPC excitation signal and the first two or three tap values become sensitive to transmission errors in the range of 0.1 to 1.0 percent error rate; while, the later tap values become sensitive in the 1.0 to 10.0 percent range of bit error rate. This one to two orders of magnitude difference in the error sensitivity of the LPC data bits is sufficient to make it a basic consideration in the development of the ANDVT.

5.4 Characterization of Errors on HF Channels

Bit errors occurring in an HF transmission system may be broadly classified in three groups as either random, deterministic or bursts. Random errors predominate at low signal to noise (S/N) conditions. Bursts are generally the result of wideband fading phenomena or wideband interference, which may be either natural or man-made. Those errors occurring in a somewhat deterministic fashion are usually associated with either narrowband frequency selective fading or narrowband interference. The latter is almost exclusively man-made. The modem design strongly influences the statistics of this class of errors. For example, a frequency division multiplex (FDM) four-phase differential phase shift keyed (DPSK) modem has a high probability of creating errors in two successive data bits. When the average bit error rate is worse than 1×10^{-3}, approximately 10 percent of the errors are associated with two bit clusters. Narrowband fading or interference can produce error patterns at multiples of the modem frame size when an individual subchannel has a high error rate for several successive frame periods. This means that for the 16 tone, four-phase DPSK standard modem there is a high probability of errors occurring at multiples of 32 bits $\pm$ 1 bit. Gaps between errors equal to the frame period can also be attributed to the differential detection process. A fade or interference during one frame period can cause additional errors in the following frame because the phase reference was corrupted. Wideband phenomena, which causes a very high probability of errors on all subchannels in a given frame, can be counteracted by coding over a time interval longer than the period of the burst; but, the interactive nature of voice communications limits the amount of time delay that can be used to combat this form of interference. Coding in the frequency domain, by confining each code word to a given modem frame period, combats the narrowband phenomena, but provides no protection against errors resulting from a wideband burst.

5.5 Application of Coding to LPC

Granted the premise that it may be beneficial to encode the voice processor data to provide protection against transmission errors, the first alternative is whether or not to provide equal coding on all bits. The value of encoding all data bits is a simpler system; but, the system simplicity is achieved with an excessive increase in transmission rate. It does not achieve the objective of "whitening" the data, but merely shifts the S/N condition that produces a degraded system performance.

The selective encoding of only the most sensitive information bits provides the protection where needed without excessive increase in the transmission rate. The alternatives then are concerned with whether the encoder should be RED or BLACK (at the LPC or at the modem).

The encoding of part of the data bits for protection against transmission errors can be most simply performed as a RED function at the voice processor, because at this point bit identity is known without the necessity to pass framing information from RED to BLACK areas. The penalty paid for RED encoding is that the digital bit rate will be greater than 2400 bps, and in order to maintain an acceptable standard rate (i.e., either 3600 or 4800 bps) the forward acting error correction code (FEC) must be selected from a limited set. A second objection to having the FEC in the RED area is that it would prevent the possibility of making a BLACK digital interconnection between an HF and a wireline terminal that did not use FEC.

If the FEC is in the BLACK area, the choice is between either making it an integral part of the HF modem or a separate unit. In either case, a clock signal at the voice processor frame rate is necessary in order to identify the data bits to be encoded. As a separate BLACK function, the encoder could operate at the voice processor frame rate. An advantage of shifting the FEC from the RED to BLACK side is that it would be used only prior to transmitting on HF, thus the intervening digital transmission system operates at 2400 bps. Furthermore the coding algorithms may then be used either for protection for the key generator preamble, for transmission of a new key generator variable or for transmission of external digital messages.

Incorporating the FEC as an integral part of the HF modem provides additional alternatives. One approach is to make the modem frame period equal to the voice processor frame period and use a block code constrained to the same period. This optimizes the system framing problem. Achieving modem sync automatically insures that the FEC and voice processor are correctly framed.

The FEC can be incorporated into the modem even when the modem frame rate is different from the voice processor, but this entails additional buffering and framing if block codes are used. Convolutional codes can be used to advantage when the modem frame rate and the voice processor frame rate are different. The FDM aspects of the HF modem may be used to isolate the uncoded bits, the coded information bits and the parity bits. The voice processor framing signal is necessary at the transmit modem to identify the bits to be encoded, but the receive modem need not provide frame sync for the receive voice processor. Incorporating the FEC in the modem provides the additional alternative of using soft decision decoding rather than hard decision decoding, but at the cost of additional complexity.

If coding were not used to protect a portion of the data, there are still some steps that may be taken to lessen the effect of errors. If the HF modem is a FDM four-phase DPSK, then there will be a high occurrence of two bit error clusters and errors at intervals equal to the modem frame size $\pm$ 1 bit. The effect of two bit error clusters on the synthesized speech can be lessened by formatting the voice processor data so that successive bit pairs have a nearly equal average rating on a "sensitivity to errors" scale. Likewise, the formatting of the data could attempt to minimize the effects of periodic errors at the modem frame rate.

5.6 Modem Design for Error Control Capability

If the HF modem were operating at the same frame rate as the voice processor (44.44 frames/second) then it is possible to significantly increase the number of parallel tones in the modem. This can be done while maintaining the same modulation technique (four-phase DPSK), while still providing an adequate time guard period (~4 milliseconds) and keeping the overall transmission bandwidth centered within the available bandwidth (2700 Hz). For example, using a sampling rate of 7200 Hz and an integration period equal to 128 samples, the tone spacing would be 56.25 Hz and the time guard would be 4.72 milliseconds. For a transmission rate of 2400 bps, it would require 27 data tones covering a bandwidth of only 1518.75 Hz. A bandwidth of 2400 Hz would provide for a transmission rate in excess of 3700 bps. Thus, it is feasible to consider a 50 percent increase in transmission rate to provide the redundancy needed to protect portions of the LPC data.

The primary disadvantage of considering a modem frame rate of 44.44 frames/second, rather than the standard 75 frames/second, is that the DPSK demodulator would be more sensitive to doppler. This would be so because with the longer frame period the phase reference is older. It turns out that the effects of doppler spreading is more significant at high S/N conditions than at low S/N where the additive noise is the controlling factor over the resultant bit error rate. Thus, for digital voice applications, there would be minimal difference in the effects of doppler spreading. Examples of this are shown in

Figures 1 and 2. These show the performance of four-phase DPSK over a time varying Rayleigh channel with RMS doppler bandwidth from zero to 5 Hz and for frame rates of either 44.44 or 75.0.

Another consideration in reducing the modem frame rate from 75.0 to 44.44 is that the number of tones must be increased from 16 to 27 just to maintain the 2400 bps transmission rate. Thus, the power per tone will be less. Multitone modems operating into single sideband HF transmitters must be concerned with peak power as well as average power. But, this is a primary concern only when the number of tones is very small. Increasing the number of tones from 16 to a larger number may be considered strictly on the basis of maintaining average power level. This is especially true for the digital voice application where a very infrequent occurrence of peak clipping in the transmitter would have insignificant effect on the digital voice transmission. The decreased power per tone is offset by the improvement in S/N due to the longer integration period (smaller noise bandwidth) at the receiver. For the signal design example used earlier, of a 27 tone modem at a tone spacing of 56.25 Hz, the 2.27 dB decrease in power per tone at the modulator is offset by a 2.91 dB increase in the demodulator S/N due to the longer integration period.

$$10 \log (27/16) = 2.27 \text{ dB}$$

$$10 \log (110/56.25) = 2.91 \text{ dB}.$$

This is equivalent to saying that a smaller percentage of the long frame period is used for time guard, even though the actual time guard is longer (4.72 milliseconds compared to 4.24 milliseconds). The number of tones can be increased to 32 before the loss in power per tone exceeds the improvement due to the decrease in noise bandwidth.

The conclusions reached were that there were two strong advantages to consideration of a new HF modem which was frame synchronous with the voice processor. Those advantages were:

- The terminal synchronization process could be made less subject to failure by designing a frame synchronous system.
- A more efficient signaling format could be used which would provide for error control on the most sensitive bits of the LPC.

6.0 CANDIDATE HF MODEM

In order to have a starting point for the HF modem signal design studies, for sizing out of software and hardware implementations, and for the terminal architectural studies, it was necessary to generate a strawman specification of the best estimate of the HF modem. This estimate was based on the work performed on the use of coding in HF modems [3, 4, 5, 6]. The essential parameters of this candidate modem are given below for operation in the digital voice mode:

- 39 data tones spaced 56.25 Hz apart between 843.75 Hz and 2981.25 Hz.
- Four-phase DPSK modulation.
- Frame rate of 44.44 frames/second.
- Doppler tracking tone at 562.5 Hz.
- Slot sync tracking centered at 787.5 Hz.
- Input information rate of 2400 bps.
- FEC on 24 data bits, consisting of two code words of extended Golay code (24, 12).
- Soft decision decoding.
- Modem preamble consists of two parts with a maximum duration of 32 frames each. First part consists of 3 unmodulated tones for doppler acquisition. Second part consists of three phase modulated tones for synchronization.
- Provision for encoding a system preamble of maximum length of 256 bits. Encoding consists of coding all bits with (24, 12) code and transmitting five times.
- Sampling rate of 7200 samples per second.
- 128 point Fast Fourier Transform (FFT).

In the digital message mode of operation, the candidate modem had the following general parameters:

- 16 data tones spaced 112.5 Hz apart between 900 Hz and 2587.5 Hz.
- Four-phase DPSK modulation.
- Frame rate of 75.0.
- Two doppler tracking tones at 562.5 Hz and 2925.0 Hz.
- Slot sync tracking centered at 787.5 Hz and 2700 Hz.
- Input information rate of 1200 bps.
- Transmission rate of 2400 bps.

- All data bits encoded with (24, 12) code and interleaved over N code words, where N is a multiple of four, and with a goal of N = 200.
- Modem preamble is the same as in the voice mode.
- In the data mode the modem treats the system preamble the same as digital data.
- Sampling rate of 7200 samples per second.
- 64 point FFT.

The detailed specifications for the candidate HF modem are presented in Appendix A. These include flow charts of the modem operation for half duplex operation in both the voice and data mode. Figures 3 through 6 are functional block diagrams of the candidate modem.

The control and signaling aspects of the HF modem, as well as for any other mode of transmission, are highly dependent upon the terminal and system architecture. Again, in order to have a starting point for the ultimate solution of these problems, it was necessary to generate a proposed functional description of the terminal. A copy of that description is presented in Appendix B.

The functional description of the terminal described a special mode for operating point-to-point. It is not just a two-party net. The concept of point-to-point operation was defined as one in which each party transmitted the full modem and key generator preambles after the first push-to-talk after going off hook. Subsequent push-to-talk signals would initiate an abbreviated modem preamble with no key generator preamble. The objectives of this mode of operation were to:

- Reduce the turn around time on a half duplex link when only two parties were involved.
- Reduce the probability of failure to establish a link due to missing the modem and/or key generator preamble.
- Provide a capability for a BLACK digital patch with a full duplex wireline terminal.

The disadvantage of this mode of operation is that portions of the modem and key generator must at least appear to function like full duplex equipments, because both transmit and receive clocks must be maintained at both ends in order to get the maximum advantage of point-to-point operation.

The clear voice bypass mode described in the functional description of the terminal was a digital bypass approach. It was selected on the assumption that the need was to provide a communication capability when one terminal did not have the proper key variable. The TRI-TAC requirements document indicates that the user's principle concern is to have the ability to bypass the terminal when it is not functioning.

That is, to provide an analog bypass capability. Such an approach would obviously not provide a bypass capability for those links where ANDVT operate into a digital interface, such as FLTSATCOM.

7.0 SIZING OF MODEMS

The candidate specification for the HF modem included in Appendix A is representative of the HF modem that will be implemented in ANDVT.

This type of modem can be implemented using high speed processors with read-only memory programs or in dedicated hardware. The two QUINTRELL contractors have determined the memory and timing requirements to implement the candidate HF modem in their respective processors. The QUINTRELL contractors have also suggested dedicated hardware approaches for implementing the candidate HF modem. While the dedicated hardware approaches result in savings in size, power, and cost over a single modem processor, much of the modem flexibility and serviceability are lost with dedicated hardware. Since a processor implementation will be used for the voice processor, a similar or identical processor assembly could be used for the modem which has similar speed and processing requirements.

The size, weight, and power required for the modem portion of the terminal is a function of the architecture chosen. The architecture used may be decided more on security aspects necessary to maintain RED/BLACK isolation than to minimize the size, weight, power, and cost of the terminal.

Some of the possible combinations of voice processor and modem function are listed below:

1. Single processor for voice processor and modem.
2. Separate processor for voice processor and modem.
3. Single processor for voice processor with hardware modem.
4. Single processor for voice processor and modem receiver with modem transmitter in hardware.

Note that each of these implementations includes a processor for the voice function. A processor is a relatively costly complex assembly even if its performance is relatively low. The least expensive way to add additional functions is to incorporate them in the processor by adding or changing components to increase the speed, and by adding the necessary program and data memory. This fact would suggest the first implementation, but it might not be possible to maintain the required RED/BLACK isolation without adding a prohibitive amount of hardware.

The second and third implementations allow easy separation of RED/BLACK functions. These could either be combined in a single enclosure to minimize size, weight, power, and cost or the modem could reside in a separate enclosure. While such a modem could be used for other purposes utilizing its data mode, the modem is optimized for use with the voice processor.

The fourth approach takes advantage of the fact that a modem transmitter can be implemented in dedicated hardware using much fewer components than required for a processor and with an even larger power saving. This technique could meet the RED/BLACK isolation problem while avoiding a second costly processor in the terminal.

In each implementation the terminal sizing is dependent on the total number of functions implemented such as half duplex versus half/full duplex, data mode, etc. Adding these functions to a modem processor would impact the sizing very little compared to adding functions to dedicated hardware.

The modem functions alone can be implemented in a 3/8 ATR short. This packaging was demonstrated by TRW in their QUINTRELL PROM model. In production either of the QUINTRELL processors could be produced in this size with adequate room for a single processor, power supplies, and memory to incorporate all proposed modem functions.

It is desirable for the power required by the terminal to be low enough to allow a sealed ATR enclosure. The power required will be very dependent on the semiconductor technology used to implement each function.

In general, the cost of components to implement the HF modem is lower for dedicated hardware than for a separate processor, if the implementation provides for only half duplex, non-diversity operation. While the initial cost saving may seem attractive, the life cycle cost may favor the modem implementation in a processor. Another advantage of a software modem would be the possibility of including the ESVN wire-line modem code by adding the small additional memory necessary to store the program. This WE 201B compatible wireline modem could be used for the VHF and UHF channels as well as wireline applications.

8.0 SUPPLEMENTARY HF SIGNAL DESIGN

A contract effort was established with CNR of Needham, Massachusetts to aid in the study of the signal design for the HF modem. Reference [8] is the final report on that effort. A brief summary of the basic objectives follows:

- To develop by analysis and simulations the signal format for:
  - A robust modem preamble
  - An integrated LPC/HF modem
  - A digital message mode
  - Transmission of key generator preamble.
- To assess the software processor requirements to implement the recommended designs.

8.1 Modem Preamble Design

The HF modem preamble serves four functions in a half duplex operation. They are:

- Establishment of receiver AGC level
- Detection of signal presence
- Doppler acquisition
- Modem frame sync acquisition.

The rapid, accurate and reliable performance of the modem preamble functions is most important in half duplex operation. This is because the most critical information in each transmission sequence is the key generator preamble, which is the very first data transmitted. Thus, one cannot rely on a modem preamble that obtains only a rough estimate of the frequency offset and synchronization and depends on slow-acting tracking loops to refine those estimates. The preamble must be of sufficient duration and the signal design must have enough redundancy that a good estimate of the preamble functions may be made <u>before</u> the reception of the key generator preamble. This would maximize the probability of detecting the key generator preamble correctly.

The preamble design in the present MIL-STD 188C modem has several deficiencies:

- The duration is only five frame periods (66.67 milliseconds), which is too short.
- Both the doppler and the sync signals are single tones, that in a frequency selective fading channel can be very unreliable.

The doppler problem was looked at in this study to determine the precision with which a frequency offset can be estimated as a function of the time allowed and the signal design. The maximum offset assumed was ± 75.0 Hz. This was based on the requirements in MIL-STD 188C for the standard HF modem. The signal design consisted of four unmodulated tones evenly spaced in the band. The phasing of the tones were selected to minimize the peak/average power in the resultant waveform. Selection of four tones was a compromise between factors requiring the transmission of a large number of tones and the factors favoring the use of as few tones as possible. Some of these factors were:

- Maintenance of the RF receiver AGC in a frequency selective fading channel favors transmission of many tones.
- Maximization of diversity requires many tones.
- The number of tones is limited by the practical requirement to have several hundred Hertz separation between tones in order to get uncorrelated fading. This implies a large bandwidth if many tones are transmitted. But, the total bandwidth must not exceed that available for the data tones.
- The power per tone should be maximized, which favors the transmission of as few tones as necessary.

The doppler estimation algorithm was based on the rate of change of the sampled in-phase and quadrature components of each filtered tone.

$$\hat{d} = \frac{1}{2\pi} \frac{\sum_{\ell=1}^{N} \sum_{k=1}^{K} \left(X_{k\ell} \dot{Y}_{k\ell} - Y_{k\ell} \dot{X}_{k\ell} \right)}{\sum_{\ell=1}^{N} \sum_{k=1}^{K} \left(X_{k\ell}^2 + Y_{k\ell}^2 \right)}$$

where x and y are the in-phase and quadrature components, N is the number of time samples in a given stage and K is the number of tone filters. The results indicate that in order to arrive, in a reasonable processing time, at a doppler estimation error of 0.5 Hz or less, it is necessary to go to a multistage estimation approach. In such an approach the bandwidth of the filter associated with each tone is successively narrowed from the original bandwidth of 150 Hz. One to five stages of estimation were looked at where a doppler correction is made at the end of each stage with a corresponding decrease in the filter bandwidth. The bandwidth of the stages after the first were set to three times the standard deviation of the estimate at the output of the previous stage. Characteristics of the multistage estimator are given in Table 2 for the condition that the design value for the standard deviation was 0.5 Hz. The bandwidth and processing time per stage are given as well as the final RMS error achieved and the total processing time. The first line of data in the table indicates that a single stage estimator, which keeps the bandwidth constant at 150 Hz, would require a processing time of 7.7 second. This is entirely unreasonable. The processing time must be reduced significantly. The data in line two shows that by use of a two-stage estimator the time can be reduced to approximately 0.3 seconds. Use of more than two stages provides a small improvement in the RMS error and processing time, but not significant enough to warrant implementation. A two stage doppler estimator is recommended for the ANDVT based on performance and minimization of implementation problems.

The unkeyed tones used for doppler acquisition are also used initially for signal presence detection. The approach preferred is to compare the sum of the energy in the four tone filters (150 Hz bandwidth) with the sum of the noise energy in four equal bandwidths. Signal presence is declared when the ratio of these two sums exceed a threshold. The important point to note about this structure is that the effect of the receiver AGC is cancelled out in the detector because the signal and the noise filters are affected equally. Figure 7 is a diagram of this form of signal presence detector. The threshold for declaring signal presence should be chosen based on a compromise between setting a high threshold to minimize the probability of false alarm (P_{FA}) and setting a low threshold to minimize the probability of an incorrect dismissal (P_{ID}). Figure 8 is a plot of the P_{ID} versus processing time for several P_{FA} for a SNDR of 40 dB with independent fading on the four tones. A P_{ID} of 2.5 X 10^{-3} was obtained with a P_{FA} of 1 X 10^{-5} for a processing time of 50 msecond and a threshold of 2.65. A P_{FA} of 1 X 10^{-5} for a processing time of 50 mseconds represents one false alarm in a time period of 83.3 minutes.

The fourth function to be performed by the modem preamble is to obtain terminal synchronization. This consists of obtaining modem frame synchronization and identification of the beginning of the key generator preamble. This may be done in either one or two steps. CNR examined the one-step approach. This consists of transmitting a wideband pulse train designed both for high crest factor and low autocorrelation side-lobes. A length 15 Carley sequence was chosen because of its tolerance to doppler shift. The proposed sequence would be transmitted in 15 mseconds within an available bandwidth of 2 kHz. An alternate approach is to transmit multiple narrowband signals to obtain modem frame synchronization and then to transmit a framing pattern to identify the beginning of the key generator preamble. The factors which influence which approach to use are:

- The range of time uncertainty that must be searched.
- The processing time required.
- Relative performance.
- Implementation problem.

8.2 COMSEC Synchronization

One of the major aspects of the terminal synchronization problem is the correct reception of the key generator preamble. The problem was examined from the viewpoint of ascertaining the signal design necessary to insure correct reception of the preamble under channel conditions considerably worse than the threshold for acceptable digital voice operation. For the purpose of this study three lengths of preambles were considered, 100 bits, 128 bits, and 150 bits. Assuming a 40 tone 4 phase DPSK modem, the number of frames of transmissions was calculated to reach a decoded bit error rate of 1×10^{-7} when the preamble was encoded in a shortened BCH code and transmitted repeatedly. The codes examined were:

$$(255,155) \rightarrow (250,150;27)$$

$$(255,131) \rightarrow (252,128;37)$$

$$(255,105) \rightarrow (250,100;46)$$

The channel simulated was a time varying Rayleigh channel with a RMS doppler spread of 5 Hz. A family of performance curves were obtained relating S/N versus transmission time for each code. The results indicated that for the strongest code (250,100), a required S/N ratio per tone of -0.2 dB would be sufficient for a preamble length of 25 frames (0.5625 seconds). This was for soft decision decoding. Hard decision decoding required a 2 dB greater S/N. In comparison, the (252,128) code required either +0.6 dB or 2.9 dB depending on whether soft decision or hard decision decoding were used, and the S/N values were either +1.7 dB or 3.9 dB for the (250,150) code.

The final selection of the particular BCH code to recommend for ANDVT was based on the following:

- The number of information bits to encode shall consist of the message indicator plus a mode control word.
- The mode control word shall be at least 5 bits long.
- The encoded word shall be transmitted with the 39 tone, four-phase DPSK modem with eighth order diversity.
- The total number of bits transmitted shall include a minimum number of dummy bits necessary to keep the transmission equal to an integer number of frames.
- The total number of information bits shall be a multiple of eight in order to maximize throughput if this coding scheme were to be used for transmission of signaling information (8-bit ASCII characters).

- The sum of information and parity bits shall be an even number in order that the encoded bits may be grouped into two bit pairs (dibits) to simplify the diversity combining.

The decoding of a long BCH code is very time consuming. Estimates of the decoding time for QUINTRELL type processors is approximately 50 mseconds. This will require that the ANDVT transmission sequence allow for extra decoding time if software implementation is used.

8.3 Coding of LPC Parameters

The performance of a combined LPC and HF modem with coding has been determined using computer simulations of the algorithms and Rayleigh fading channel. Data has been obtained for the Golay (24, 12) code compared to an uncoded modem. Both hard and soft decision decoding algorithms have been used. Figure 9 is a plot of the bit error rate as a function of energy per bit/noise density ratio (E_b/N_o) for the coded and uncoded system. The points of greatest interest are, of course, those at the lowest usable E_b/N_o conditions. These indicate that soft decision decoding provides up to 9 dB gain over an uncoded system. Listening tests on the synthesized speech confirm this. The data in Figure 9 is for a six-path channel model representing a condition most beneficial for frequency domain coding. Results for a two-path channel model are shown in Figure 10. The coding gain is not as significant for this channel, which does not have as much frequency selective fading. Minimum coding gain would be expected for a non-fading Gaussian channel. Performance of the Golay code with four-phase DPSK modulation is shown in Figure 11 for a Gaussian channel.

A limited amount of work was performed in simulation of a combined LPC and HF modem using codes spread over more than one frame period. A maximum spread of ten frame periods was used. The results favored the use of spreading for E_b/N_o conditions greater than 11 dB. Tests were not performed with impulse type noise which would probably favor the use of spreading the code over multiple frames. The simulations performed used an NRL version of the LPC-10. The bits selected to be encoded for the (24, 12) simulations are shown in Table 3. A relative ranking of all bits as to their sensitivity to transmission errors is shown in Table 4.

The performance of a soft decision decoder is 2 to 3 dB better than a hard decision decoder for Gaussian noise conditions. But for a non-Gaussian channel, such as the HF channel, the improvement obtained by using soft, rather than hard, decision decoding can be significant, as was shown in the simulations in Figure 9. The cost of obtaining that capability in a software implementation is largely a significant increase in processing time. In most soft decision decoding algorithms the decoding time is a variable. But, if in real-time implementation information must be processed frame by frame with minimum buffering, then it is the worst case decoding time that must be accounted for.

In the candidate modem specification, two (24, 12) Golay code words must be decoded per frame. Soft decision decoding was specified. An algorithm for soft decision decoding of the Golay code is shown in Figure 12. Using this algorithm the maximum decoding time is controlled by the possibility of having to perform the binary decoding 16 separate times. A table look-up type of binary decoder that trades off memory requirements for speed of execution would seem to be the best approach.

Early in the study of the HF signal format for digital voice, the decision was made not to investigate the advantages of data labeling. Data labeling refers to the use of error correction coding to label code words that have a high probability of being uncorrectable. The voice synthesizer would use this information to interpolate over the labeled frames. The arguments against considering data labeling within the scope of the present work were:

- Most of the past efforts to determine the advantages of data labeling, as reported on in the FRANKLIN, CODEM I and CODEM II [3, 4, 5] work, were concerned with quantitative evaluations. Speech systems are still being rated, compared and evaluated by means of subjective criteria. Extensive testing would be required to determine how and when to use labeling and the advantages gained. This would be especially true if the voice synthesizer incorporated some form of statistical interpolation on the digital parameters.

- The present voice processor frame rate for ANDVT (44.44 F/S) is a minimum. Thus, there is often minimum correlation from frame to frame on some parameters. Often speech may be voiced or unvoiced for only one or two frames. It is at these transition points where the parameter values are changing rapidly and their value are most critical to the determination of the synthesized sound. Thus, interpolation over a transition point may turn a noise burst into a distinct but incorrect sound.

- If the voice processor employed a different bit assignment depending on the voiced or unvoiced decision, then this would further complicate the use of data labeling.

- Error correction coding is considered a BLACK function in the ANDVT. The application of data labeling must be a RED function if interpolation is used. Thus, this would require the passing of data labeling information from BLACK to RED areas. This is a further complication to the terminal, especially if the BLACK function is implemented as a separate module.

NSA is giving serious consideration to specifying a dual format LPC. It would be a tenth order predictor in the voiced mode and a fourth order predictor in the unvoiced mode. Table 5 is a list of the proposed encoding scheme for both voiced and unvoiced frames. In the unvoiced mode, pitch, sync, and the first four coefficients are transmitted once per frame. The remaining 16 bits are used for forward acting error correction. NSA is considering four code words of an (8, 4) code. Table 6 lists a coding scheme being considered by NSA.

The proposed dual format LPC would impact in two ways on an HF modem which itself applied error correction coding to a portion of the input data. It would:

- Affect the selection of the bits to be encoded in the HF modem and, because the modem must operate without knowledge of whether a frame is voiced or unvoiced, it would affect the bit placement scheme in the data format.

- The concatenation of a weak error correction code (8, 4) in the voice synthesizer with a strong code (24, 12) in the modem may negate the effects of the strong code.

The concatenation of the two codes are the principle concern. In particular, the concern is for the condition when the coding in the modem encodes only the information bits and not the parity bits of the code word that was generated in the LPC. Under these conditions the LPC (8, 4) decoder will see different average bit error rates on the four information bits than on the four parity bits. Figure 13 is a diagram of such a concatenated coding scheme for one four bit data symbol. These four bits are encoded by the (8, 4) encoder in the LPC analyzer. This code is a single error correction, double error detection Hamming code. The eight bits out of the encoder are divided into four information bits and four parity bits. The four information bits are further encoded in the HF modem. The four parity bits from the (8, 4) encoder are not encoded in the HF modem. The modulator and demodulator treats all bits equally so that all demodulated bits are assumed to have an average bit error rate of Q1. The modem decoder has an average error rate output of Q2 for an input of Q1. Thus, the input word to the (8, 4) decoder is four bits with an average error rate of Q1 and four bits with an average error rate of Q2.

NSA's selection of the augmented version of the (7, 4) Hamming code was based on achieving a double error detection capability. This implies that data labeling would be used in the voice synthesizer. Thus, there are two conditions of interest when this code is concatenated with a coding scheme in the modem. They are:

- The (8, 4) decoded bit error rate as a function of Q1 and Q2. Decoded bit error rate being the undetected symbol error rate divided by two. And, the undetected symbol error rate being equal to the probability of 3 or more errors in the input word. Assumption is that data labeling of detected two errors in the input word can be used effectively.

- The (8, 4) decoded bit error rate as a function of Q1 and Q2 assuming no data labeling (two or more errors in the input word).

Figures 14 and 15 show the (8, 4) decoder performance as a function of the error rate on the information bits (Q2) for fixed error rates on the parity bits (Q1) both with and without data labeling. Figure 16 shows the results when the coder in the HF modem is selected to be a (24, 12) code. The straight line on these figures indicates no coding gain. Note that most of the curves for the data labeling case lie in the region of coding gain, while essentially all of the data points for the case of no labeling lie in the region of coding loss. Since the feasibility of using data labeling to improve the quality of LPC speech is highly questionable, it appears that the best results on an HF channel will be obtained by simply bypassing the Hamming decoder.

8.4 DPSK Modulation

MIL-STD-188C defines DPSK modulation for the MIL-STD HF modem as the phase change on each tone between the beginning of one baud and the end of the previous baud. Such a definition makes it compatible with an implementation where each tone is generated by a continuously running oscillator. This was the method by which early DPSK modems were implemented. All recently designed HF modems are digitally implemented. Tones are generated digitally by scanning cosine tables or by FFT routines. Phase changes are introduced by entering the tables at a different point or by changing the real and imaginary input values to the FFT.

For digital designs to be compatible with the MIL-STD definition of DPSK, the phase shift must include the amount of phase change that occurred during the time guard period. This is a variable that is a function of each tone frequency. Thus, a table of values would have to be stored. The phase of the signal at the beginning of a baud would then be equal to the phase of the signal at the beginning of the previous baud plus the phase change during the time guard period, plus the phase change imparted by the information to be transmitted. That is,

$$\phi \text{ new} = \phi \text{ old} + \phi \text{ tg} + \phi \text{ inf}$$

It is not necessary that DPSK modulation be defined as if each tone were a CW signal. The modulation can be defined as the phase change between the beginning of two orthogonal periods, which may or may not be separated by a time guard. Thus,

$$\phi \text{ new} = \phi \text{ old} + \phi \text{ inf}$$

This is the definition of DPSK that was used in the standard modem programmed on the Sylvania PSP [6], and it is the definition used in the specification for the candidate ANDVT modem (Appendix A), which was provided to the ANDVT architecture contractors under Subtask 3.3.

Utilizing this definition of DPSK modulation for the ANDVT HF modem will eliminate the need to store a table of values for Ø tg (approximately 40 values), and it will reduce the number of operations to perform in both the modulator and the demodulator. The performance of the demodulator would not be changed.

The one area where this definition of DPSK modulation might introduce some change is in the performance of slot synchronization. Slot sync is a technique for sync tracking by performing early and late integration on an empty tone slot. Energy will build up in these integrators when the timing is not perfect. The amount of energy is a function of the timing error, the distance between the sync slot and the first data tone and the phase shift on the data tone. In actuality, all data tones contribute to the sync energy, but their contribution is inversely proportional to their distance from the sync slot; so, the first tone is the principal contributor. One of the reasons that the phase change for four phase modulation is defined as odd multiples of 45 degrees is to insure that some phase change would always occur independent of the information being transmitted, thus improving slot sync performance. This new definition of DPSK modulation does not insure this. Depending on the tone library and the time guard selected, the phase change during the time guard period may be equal and opposite to the phase shift for a fixed data pattern, thus resulting in no phase change between bauds. With encrypted data this could occur no more than ¼ of the time for four phase modulation.

Table 7 shows how this pertains to the candidate modem tone format. The sampling frequency for this modem is 7200 Hz, with 162 samples per baud. Thus, with an 128 point FFT, the time guard is 34 samples. Column two lists the phase change during the time guard period for all 63 possible tones. These values are all multiples of 95.625 degrees (module 360 degrees). Columns 3 through 6 lists the resultant phase change between bauds for Ø inf equal to odd multiples of 45 degrees. The selection of a tone library may be made to minimize the effect in slot sync.

8.5 Soft Decision Decoding for the Golay Code

The Golay code with soft decision decoding is recommended for the digital voice mode for ANDVT. The decoding algorithm corresponds to version two of the Chase Generalized Minimum Distance Decoder (GMD) described in reference [7]. The following is an interpretation of the algorithm as applied to the Golay code (24, 12).

The inputs to the GMD decoder are 24 bits of demodulated data and 24 confidence values for the data bits. The specification for the candidate modem stipulated that each 24 bit code word would be transmitted as either the in-phase or quadrature-phase bit of 24 separate 4-phase DPSK tones. Thus, the confidence value for each bit will be the absolute value of either A cos θ or A sin θ, where A is the amplitude of the received vector and θ is the detected phase change. The data bits for a given code word are demodulated as either the sign of $A \cos \theta$ or the sign of $A \sin \theta$.

The first step in the soft decision decoding algorithm is to identify the four bits in a received code word that have the lowest confidence values. These will be the bits that are most likely to be in error. The basic approach of the GDM algorithm is to go through all $2^4 = 16$ possible combinations of these four bits and decode the resulting test word with a binary decoder. Each resultant bit error pattern from the binary decoder is modulo two added to the test error pattern that was used to generate the test word. The result of this addition is an estimate of the true error pattern. This pattern is used to obtain an equivalent analog weight by summation of the confidence values associated with the bit errors. The error pattern that produces the lowest analog weight is used to correct the received data.

Once the four bits with the lowest confidence values have been identified, they may be used to generate four 24-bit masks. Each mask is an all zero word with the exception that there is a one-bit at one of the four-bit positions identified as having the lowest confidence values. The four masks are used to generate the 16 possible test error patterns, the first of which is the all zero pattern.

Each of the 16 test error patterns are used sequentially to generate a test word for the binary decoder. This is done by modulo two addition of the test error pattern to the received word from the demodulator.

The output of the Golay binary decoder is the decoder error pattern. This may be from zero to three bits. If no errors were detected by the binary decoder then the assumption is that the test word is the best estimate of the transmitted word. If the binary decoder output identified some bits in error then this error pattern is modulo two added to the test error pattern that was used to generate the test word.

The analog weight of the final error pattern is the summation of the confidence values of the bits identified in error. This may be from one to seven bits.

This analog weight is tested against previously obtained weights. The error pattern with the lowest analog weight is retained.

If the binary decoder does not identify an error-free test word, then the procedure described above is repeated until all 16 possible test patterns have been decoded. If the decoder runs until it has examined all 16 possible patterns, then the output is the modulo two sum of the received word and the error pattern that was identified as having the lowest analog weight. If the decoder terminates before it has looked at all 16 possible test patterns, it will have done so because the binary decoder identified an error-free test pattern. In that case, the decoder output is the test pattern that was identified as being error-free. Figure 12 is a flow chart of the GMD decoder as described.

8.6 Data Interleaving

Data interleaving is a recognized technique for combating burst errors with short block codes. It is essentially a simple technique for spreading the bits from a given code word over a large time interval. The selection of an interleaver size is a two-step procedure that must be followed if the modem has a multitone format. The procedure is to:

- Select an approximate size for the interleaver based on the transmission rate and either the amount of time spreading desired or the maximum time delay permissible or the maximum buffer size.

- Select an exact size for the interleaver that will cause the individual bits of a given code word to be transmitted on a maximum number of the data tones in the modem, in order to provide frequency diversity as well as time diversity.

An interleaver of degree 226 is proposed for 1200 bps data transmission for ANDVT. This is based on providing approximately two seconds of time spreading for a (24, 12) Golay code word transmitted at 2400 bps. The modem format would be 16 tones with four phase DPSK. This modem transmits 32 bits per frame. Thus, with degree 226 of interleaving, individual bits in a given code word will be transmitted seven frames plus two bits apart. That is, if bit one of a code word is transmitted as the in-phase bit on tone one, then bit two of that same code word will be transmitted seven frames later as the in-phase bit of tone two. This shall provide maximum frequency diversity for each code word with time spreading greater than two seconds.

For 600 bps data the degree of interleaving would be reduced to 114 to provide the same time spreading with dual inband frequency diversity. Interleaving to degree 58 with quadruple inband diversity would provide the same protection for 300 bps data. The transmission rate would be 2400 bps for all cases.

The principal data requirement for ANDVT was related to the use of DMED. The design of these devices are still in an early stage. There is some indication that at least one design will use some form of coding and time interleaving. There is still a large advantage to the use of coding in the modem even when the source is already coded. This is in regard to reducing the number of random errors before decryption, if the decryption process has an error extension property. Error extension is less of a problem for the burst error condition.

8.7 Tone Library

With a digital implementation, the tone library is essentially controlled by the selection of the sampling frequency. The other factor that influences the tone spacing is the integration period, and from an implementation standpoint, it is highly desirable to make this period equal to a power of two divided by the sampling rate. The choice of a sampling rate is also influenced by the desire to make it an integer multiple of the frame rate and at least 2.5 times the frequency of the highest data tone.

The frame rate for the digital voice mode would be 44.44 frames per second. In the data mode it would be desirable to use a frame rate of 75.0 frames per second, in order to be able to recommend this data format as a standard format to be implemented in other stand alone modem configurations. Therefore, the sampling rate must be an integer multiple of both 44.44 and 75.0. A sampling rate of 7200 Hz satisfies this criteria.

When the integration period is selected to be equal to a power of two divided by the sampling rate, and when the frame rate is established by other constraints, then the sampling rate determines both the tone spacing and the time guard period. A sampling rate of 7200 Hz will establish the following characteristics for the voice and data modes:

Mode	Frame Rate	Samples/ Frame	Integration Period	Tone Spacing	Time Guard
Voice	44.44	162	128/7200	56.25	34/7200
Data	75.0	96	64/7200	112.5	32/7200

The selection of a 7200 Hz sampling rate establishes the Nyquist frequency as 3600 Hz. In order to have at least 2.5 samples per cycle at the highest frequency data tone, the tone library must be kept below 2880.0 Hz. If allowance is made for a maximum doppler error of +75 Hz then 2805 Hz represent an approximate upper bound on the tone library. The tone library recommended for the digital voice mode has the highest tone frequency at 2812.5 Hz. The lowest frequency of the 39 tone assignment is at 675.0 Hz. The constraint here is to keep it as high as possible to avoid differential time delay problems with low performance radio equipments. The frequency band is, therefore, from 675 Hz to 2812.5 Hz with ±75 Hz tolerance.

In the data mode the 16 tone library is on frequencies spaced 112.5 Hz apart between 900.0 Hz and 2587.5 Hz. This very closely approximates the transmission bandwidth for the present MIL-STD-188C modem. The present standard modem requires a sampling rate of 7040 Hz for a 64 point digital FFT implementation. This permits detection of orthogonal tones spaced 110.0 Hz apart. Unfortunately, the specified channel assignments are odd harmonics of 55.0 Hz rather than 110.0 Hz. This requires a frequency translation of 55.0 Hz at both the transmitter and the receiver when FFT techniques are used. For the 39 tone voice modem it is beneficial in processing time to use an FFT rather than a DFT both to generate and detect the tones. Therefore, implementation of the standard modem format as the data modem would not be as easily or economically accomplished as the proposed data modem. In addition, the 7040 Hz sampling rate is not an integer multiple of the frame rate (75.0). The average number of samples in a frame is 93.8667. This requires that in each 15 frame sequence, 13 frames must have 94 samples and two frames must have 93 samples, in order to minimize the synchronization tracking problems.

9.0 PROPOSED MODEMS

The proposed HF modem is a modified version of the candidate modem that was developed early in the study program. The principal changes in the design are in the following areas:

- Better definition of the coding requirements to protect the key generator preamble.
- Incorporation of a mode control word into the system preamble.
- Use of the mode control word to switch the receiver between voice and data.
- Provision for a point-to-point mode of operation.
- Data inputs at 300, 600, 1200, and 2400 bps.
- Doppler tracking on the data tones.
- Use of multi-stage doppler estimation on the modem preamble.
- Transmission of a PN sequence for system framing.
- Half-duplex operation with no out-of-band or space diversity.

The coding for the key generator preamble needs to be more robust than that which was specified in the candidate modem. The recommendation is to use a moderately long BCH code (approximately 255 bits) and encode the system preamble as one word. The encoded word would need to be transmitted at least eight times. Diversity combining followed by hard decision decoding would be used at the receiver. The system preamble would include a mode control word in addition to the key generator preamble. Mode control would be used to automatically switch the receiver between voice and data mode. The remote control unit VOICE/DATA switch would control only the transmitter function. Digital data input rates in the candidate modem were limited to 1200 bps. It is proposed to provide for four data rates of 300, 600, 1200, and 2400 bps. Data at 600 bps would be encoded at ½ rate by the FEC coding used for voice and transmitted with dual inband frequency diversity with a transmission rate of 2400 bps. Data at 300 bps would be encoded at ½ rate and transmitted with quadruple inband diversity. Data input at 1200 bps would be encoded with FEC and transmitted without diversity. Data input at 2400 bps would be transmitted with neither coding nor diversity. Time interleaving of the encoded data would be used in the terminal. The degree of interleaving would be 226 for 1200 bps, 114 for 600 bps, and 58 for 300 bps.

The signal format being proposed for the HF modem is identical to the candidate modem with the exception that no tones would be transmitted for doppler tracking. Instead, tracking information would be obtained from the data tones. The prime reason for eliminating the doppler tones is to use the available bandwidth for error correction coding.

The modem being proposed for the wireline implementation is the FED STD 1005. This is dictated by the requirements. Implementation of the ESVN software adaptive demodulator is recommended if the architecture permits. This same signal format is recommended to meet those requirements to operate ANDVT on VHF and UHF. Both of these requirements are for full duplex point-to-point operation. Use of this modem signal on the VHF link would provide for the possibility of doing either a BLACK analog or a BLACK digital interconnect between either a wireline and the FDM/VHF equipment or a separate wireline modem and a wireline modem interfaced with the FDM/VHF equipment.

10.0 RECOMMENDATIONS

The general characteristics of the HF modem that is recommended for the voice mode of ANDVT are:

- 39 data tones spaced 56.25 Hz apart between 675.0 Hz and 2812.5 Hz.
- Four phase DPSK modulation.
- Frame rate of 44.44 frames/second.
- Doppler tracking on the data tones.
- Slot sync tracking centered at 618.75 Hz.
- Input information rate of 2400 bps.
- Transmission rate of 3466.6 bps.

- FEC on 24 data bits consisting of two code words of (24, 12) Golay code.

- Soft decision decoding.

- Permutation of the bit assignment to data tones.

- Modem preamble consists of three parts. First part consists of four unmodulated tones for doppler acquisition. Second part consists of three phase modulated tones for synchronization. Third part consists of PN sequence transmitted on the data tones for system frame timing.

- Use of two stage doppler acquisition technique.

- Use of a moderately long BCH code (approximately 255 bits) for encoding the system preamble.

- Inclusion of a mode control word in the system preamble.

- Transmission of the BCH encoded system preamble with eighth order diversity (combination of time diversity and inband frequency diversity).

- Hard decision decoding of BCH code.

- Sampling rate of 7200 samples per second.

- 128 point FFT.

- Provision for point-to-point mode of operation.

- Half-duplex, non-diversity, net operation.

The characteristics of the recommended HF modem for the data mode are:

- 16 data tones spaced 112.5 Hz apart between 900.0 Hz and 2587.5 Hz.

- Four phase DPSK.

- Frame rate of 75.0 frames per second.

- Doppler tracking on the data tones.

- Slot sync tracking centered at 787.5 Hz.

- Input information rates of 300, 600, 1200, and 2400 bps.

- Transmission rate of 2400 bps for all input data rates.
- FEC on all data bits for input rates of 300, 600, and 1200 bps.
- (24, 12) Golay code with hard decision decoding.
- Interleaving on coded data to degree 226 for 1200 bps input, 114 for 600 bps input, and 58 for 300 bps input.
- Modem preamble identical to that for voice mode.
- BCH encoded system preamble identical to voice mode using 39 tone format.
- Sampling rate of 7200 samples per second.
- 64 point FFT.
- Half-duplex, non-diversity net operation.

To satisfy the presently defined requirement for a DCS terminal, it is recommended that FED STD 1005 be used. The implementation will depend on the architecture selected for the terminal. Use of the ESVN software demodulator is not cost-effective if the architecture does not permit a processor approach.

A single tone, four-phase DPSK modulation technique is recommended for those limited VHF/UHF requirements that have been identified. This is a signal design that is compatible with the wireline modem.

ACKNOWLEDGEMENTS

The authors are indebted to D. Himes and E. Kline for their guidance during the conduction of the study. The authors gratefully acknowledge the technical assistance and support provided by other NRL personnel including J. Linnehan, J. Morris, H. Beck, J. Ihnat, T. McChesney, B. Shay, G. Kang, D. Coulter, and L. Blocker. The contract work at CNR was performed by D. Chase, P. Bello, L. Pickering, R. Pinto, L. Weng, and E. Bell. Technical assistance by the staff at Booz-Allen Applied Research included J. Oetting and R. Scher.

REFERENCES

1. Joint Operational Requirement for ANDVT, SM-869-76, 26 Oct 1976.

2. "System Requirements for ANDVT", TRI-TAC Task 1 Report, 29 Apr 1977.

3. Franklin Study, CNR Final Report, Oct 1975.

4. Codem I Data Transmission System, General Atronics Final Report, 1972.

5. Codem II Final Report, General Atronics, June 1974.

6. HF Programmable Modem, Final Technical Report, GTE Sylvania, Oct 1974.

7. "A Class of Algorithms for Decoding Block Codes with Channel Measurement Information", D. Chase, IEEE Trans. Inform. Theory, IT-18,170-182, Jan. 1972.

8. HF Modem Design for the Advanced Narrowband Digital Voice Terminal, CNR Final Report, November 1977.

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Table 1. Comparison of High performance and Low Performance HF Radio Equipment (From MIL-STD 188C)

Parameter	High Performance	Low Performance
Type of Service	May be used in links connecting into other communication systems.	Used essentially in self-supporting systems. Rarely interconnect into other systems.
Frequency Coverage	2.0000 MHz to 29.9999 MHz in steps of 100 Hz at integral 100 Hz frequency increments.	Same as high performance equipment. In addition, steps of 1 kHz shall be optimal in lieu of 100 Hz
Frequency Calibration	May be calibrated within 1 part in 10^9	May be calibrated within 5 parts in 10^8
Frequency Stability	1 part in 10^8 per day, 5 parts in 10^8 during first 30 days after calibration, not more than 4 parts in 10^8 per each 30-day period thereafter.	5 parts in 10^7 per day, 1 part in 10^6 during the first 30 days after calibration, not more than 2 parts in 10^7 per each 30-day period thereafter.
Audio Frequency Response (Tx and Rx, respectively	Maximum of 1 dB variation between 250 Hz and 3100 Hz, not less than 40 dB attenuation at 50 Hz and 3250 Hz, not less than 60 dB attenuation at 3550 Hz and higher frequencies and at f_0-250 Hz and lower frequencies.	Maximum of 2 dB variation between 530 Hz and 3050 Hz and 3 dB between 350 Hz and 3500 Hz, not less than 30 dB attenuation from 0 Hz to f_0-300 Hz, not less than 60 dB attenuation at 400 Hz and higher frequencies and at f_0-300 Hz and lower frequencies.
Envelope Delay Distortion (Tx and Rx, respectively	Maximum of 500 microseconds from 300 Hz to 3020 Hz, maximum of 100 microseconds for any 100 Hz frequency increment between 300 Hz and 3100 Hz.	Delay relative to 825 Hz, not more than -300 microseconds to +500 microseconds from 825 Hz to 2750 Hz, maximum of 150 microseconds for any 100 Hz increment between 750 Hz and 2750 Hz, maximum of 250 microseconds from 2750 Hz to 2850 Hz, maximum of 350 microseconds from 600 Hz to 750 Hz.

Table 2. Multi-Stage Doppler Estimator Characteristics, for Design Value of Estimator Standard Deviation, = 0.5 Hz

Stages	Bandwidth/Processing Time Per Stage										RMS Error Achieved	Total Time (Sec)
	Stage #1		Stage #2		Stage #3		Stage #4		Stage #5			
	Hz	Sec.	Hz	Sec.	Hz	Sec.	Hz	Sec.	Hz	Sec.		
1	150	7.7									.5	7.7
2	150	.156	21.	.145							.512	.3
3	150	.041	41.	.045	11.	.056					.419	.142
4	150	.02	56.	.022	26.	.029	8.	.077			.237	.148
5	150	.016	68.	.018	31.	.02	14.	.043	6.6	.093	.086	.191

Table 3. Golay Code Protection Pattern on LPC-10 Parameters (NRL's LPC-10)

Parameter	Bits Quantized	Bits Protected
Pitch	6	5
V/uV	1	1
Amplitude	5	4
R1	5	4
R2	5	4
R3	4	3
R4	4	3
R5	4	0
R6	4	0
R7	4	0
R8	4	0
R9	4	0
R10	3	0
Sync	1	0
Total	54	24

Table 4. Relative Sensitivity of Data Bits in LPC Format (NRL's LPC-10)

1-	P-1	19-	P-5	37-	R6-2
2-	V-1	20-	A-4	38-	R7-2
3-	P-2	21-	R1-4	39-	R8-2
4-	A-1	22-	R2-4	40-	R9-2
5-	R1-1	23-	R3-3	41-	R10-1
6-	R2-1	24-	R4-3	42-	R5-3
7-	P-3	25-	P-6	43-	R6-3
8-	A-2	26-	A-5	44-	R7-3
9-	R1-2	27-	R1-5	45-	R8-3
10-	R2-2	28-	R2-5	46-	R9-3
11-	R3-1	29-	R3-4	47-	R10-2
12-	R4-1	30-	R4-4	48-	R5-4
13-	P-4	31-	R5-1	49-	R6-4
14-	A-3	32-	R6-1	50-	R7-4
15-	R1-3	33-	R7-1	51-	R8-4
16-	R2-3	34-	R8-1	52-	R9-4
17-	R3-2	35-	R9-1	53-	R10-3
18-	R4-2	36-	R5-2	54-	S-1

Table 5. Dual Format LPC Coding Scheme

Voiced	Unvoiced
Pitch - 7 bits	Pitch - 7
Amp - 5	Amp #1 - 5
Sync - 1	Sync - 1
K1 - 5	K1 - 5
K2 - 5	K2 - 5
K3 - 5	K3 - 5
K4 - 5	K4 - 5
K5 - 4	Amp #2 - 5
K6 - 4	Parity bits - 16
K7 - 4	
K8 - 4	
K9 - 3	
K10 - 2	
Total 54	54

Table 6. NSA Coding for LPC Unvoiced Frames

Code Word	Code	Information Bits
1	8, 4	4 bits of K1
2	8, 4	4 bits of K2
3	8, 4	2 bits of A1 +2 bits of A2
4	8, 4	2 bits of K3 +2 bits of K4

Tone No.	Φ_{tg}	ΦNew - ΦOld for Given ΦInf				Average of Absolute Φ Change
		$\pi/4$	$3\pi/4$	$-3\pi/4$	$-\pi/4$	
1	55.625	-50.624	39.375	129.375	-140.624	89.999
2	-168.749	-146.249	-56.249	33.750	123.750	89.999
3	73.124	118.125	-151.874	-61.874	28.125	89.999
4	22.501	22.500	112.500	-157.499	-67.499	89.999
5	118.126	-73.123	16.875	106.875	-163.123	89.999
6	-146.248	-168.748	-78.748	11.250	101.250	89.999
7	-50.623	95.626	-174.373	-84.373	5.626	89.999
8	45.002	0.001	90.001	-179.998	-89.998	89.999
9	140.627	95.623	-5.623	84.376	174.376	89.999
10	-123.747	168.751	-101.248	-11.248	78.751	89.999
11	-28.122	73.126	163.126	-106.873	-16.873	89.999
12	67.503	-22.498	67.501	157.501	-112.498	89.999
13	163.128	-118.123	-28.123	61.876	151.875	89.999
14	-101.247	146.251	-123.748	-33.748	56.251	89.999
15	-5.622	50.626	140.626	-129.373	39.373	89.999
16	90.003	-44.998	45.001	135.001	-134.998	89.999
17	-174.371	-140.623	-50.623	39.376	129.376	89.999
18	-78.746	123.751	-146.248	-56.247	33.751	89.999
19	16.879	28.127	118.127	-151.872	-61.872	89.999
20	112.504	-67.497	22.502	112.502	-157.497	89.999
21	-151.870	-173.122	-73.122	16.877	106.877	89.999
22	-56.245	101.252	-168.747	-78.747	11.252	89.999
23	30.380	5.627	95.627	-174.372	-84.372	89.999
24	135.005	-89.997	0.002	90.002	-179.997	89.999
25	-129.369	174.377	-95.622	-5.622	84.377	89.999
26	-33.744	78.752	168.752	-101.247	-11.247	89.999
27	61.881	-16.872	73.127	163.127	-106.872	89.999
28	157.506	-112.437	-22.497	67.502	157.502	89.999
29	-106.868	151.977	-118.122	-28.122	61.877	89.999
30	-11.243	56.252	146.252	-123.747	-33.747	89.999
31	84.382	-39.372	50.627	140.627	-129.372	89.999
32	-179.992	-134.997	-44.997	45.002	135.002	89.999
33	-84.367	129.377	-140.621	-50.621	39.378	89.999
34	11.258	33.752	123.752	-146.246	-56.246	89.999
35	106.883	-61.871	28.128	118.128	-151.871	89.999
36	-157.491	-157.496	-67.496	22.503	112.503	89.999
37	-61.866	106.878	-163.121	-73.121	16.878	89.999
38	33.759	11.253	101.253	-168.746	-78.746	89.999
39	129.334	-84.371	5.628	95.628	-174.371	89.999
40	-134.990	-179.996	-89.996	0.003	90.003	89.999
41	-39.366	84.378	174.378	-95.621	-5.621	89.999
42	56.260	-11.246	78.753	168.753	-101.246	89.999
43	151.885	-106.871	-16.871	73.128	163.128	89.999
44	-112.499	157.503	-112.496	-22.496	67.503	89.999
45	-16.865	61.879	151.878	-118.120	-28.121	89.999
46	78.760	-33.745	56.254	146.254	-123.745	89.999
47	174.386	-129.370	-39.371	50.629	140.628	89.999
48	-89.989	135.004	-134.995	-44.995	45.004	89.999
49	5.636	39.379	129.379	-140.620	-50.620	89.999
50	101.261	-56.245	33.754	123.754	-146.245	89.999
51	-163.113	-151.870	-61.870	28.129	118.129	89.999
52	-67.488	112.504	-157.495	-67.495	22.504	89.999
53	28.137	16.879	106.879	-163.120	-73.120	89.999
54	123.762	-78.744	11.254	101.255	-168.745	89.999
55	-140.612	-174.369	-84.369	5.630	95.630	89.999
56	-44.967	90.005	-179.984	-89.994	0.005	89.999
57	50.638	-5.619	84.380	174.380	-95.613	89.999
58	146.263	-101.244	-11.244	78.755	168.755	89.999
59	-118.111	163.130	-106.869	-16.869	73.130	89.999
60	-22.485	67.505	157.505	-112.494	-22.494	89.999
61	73.139	-28.118	61.880	151.881	-118.119	89.999
62	168.764	-123.744	-33.744	56.255	146.355	89.999
63	-95.610	140.831	-129.369	-39.368	50.630	89.999

Table 7

Phase Change Between Bauds for Condition that
Φ new = Φ old + Φ inf
for
Candidate HF Modem
F_S = 7200 Hz
NBD = 162 Samples
NFFT - 128 Samples
NTG - 34 Samples

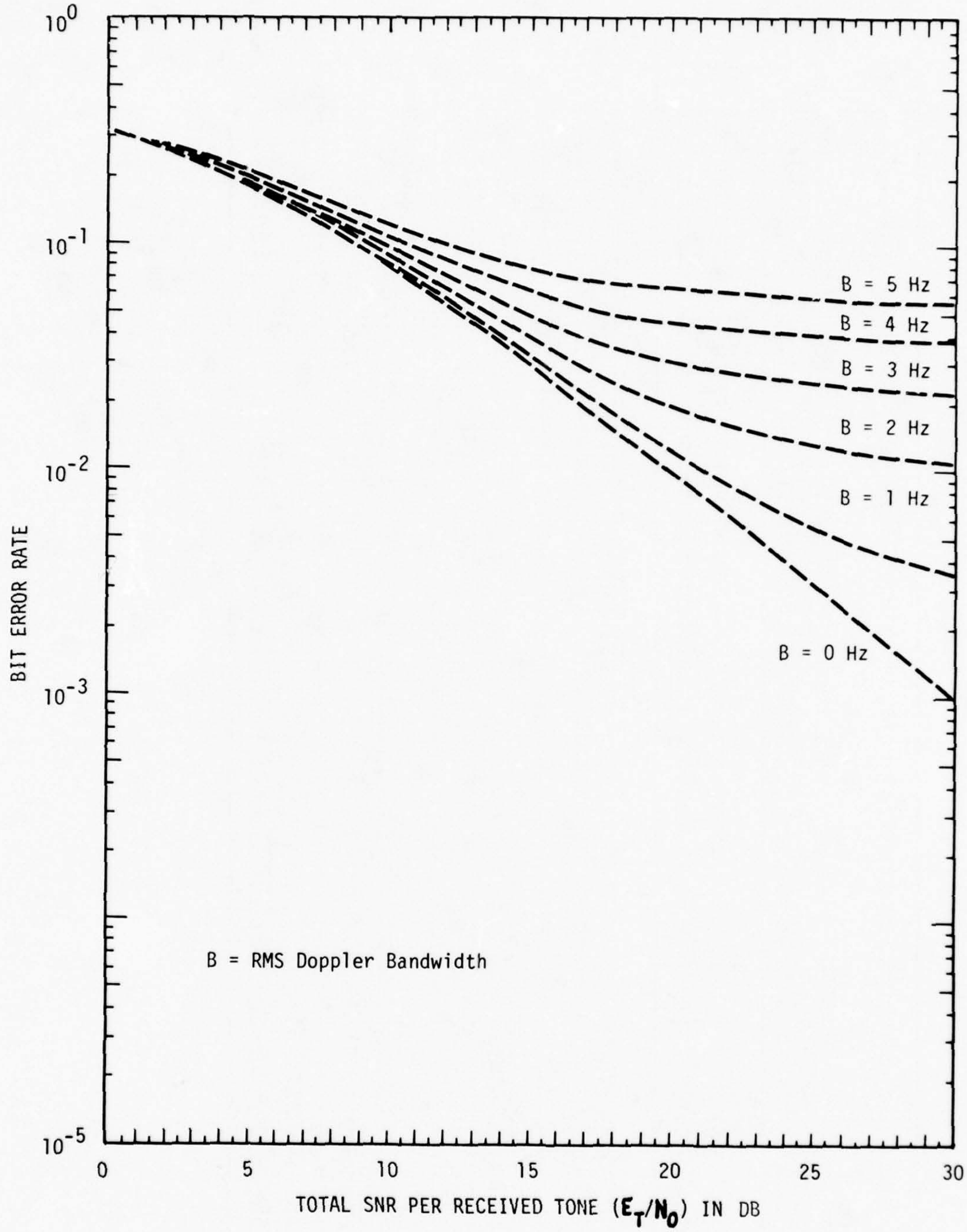


Fig. 1 - Performance of 4-phase DPSK modulation over a time-varying Rayleigh channel (Frame rate - 44.44 frames/Sec.)

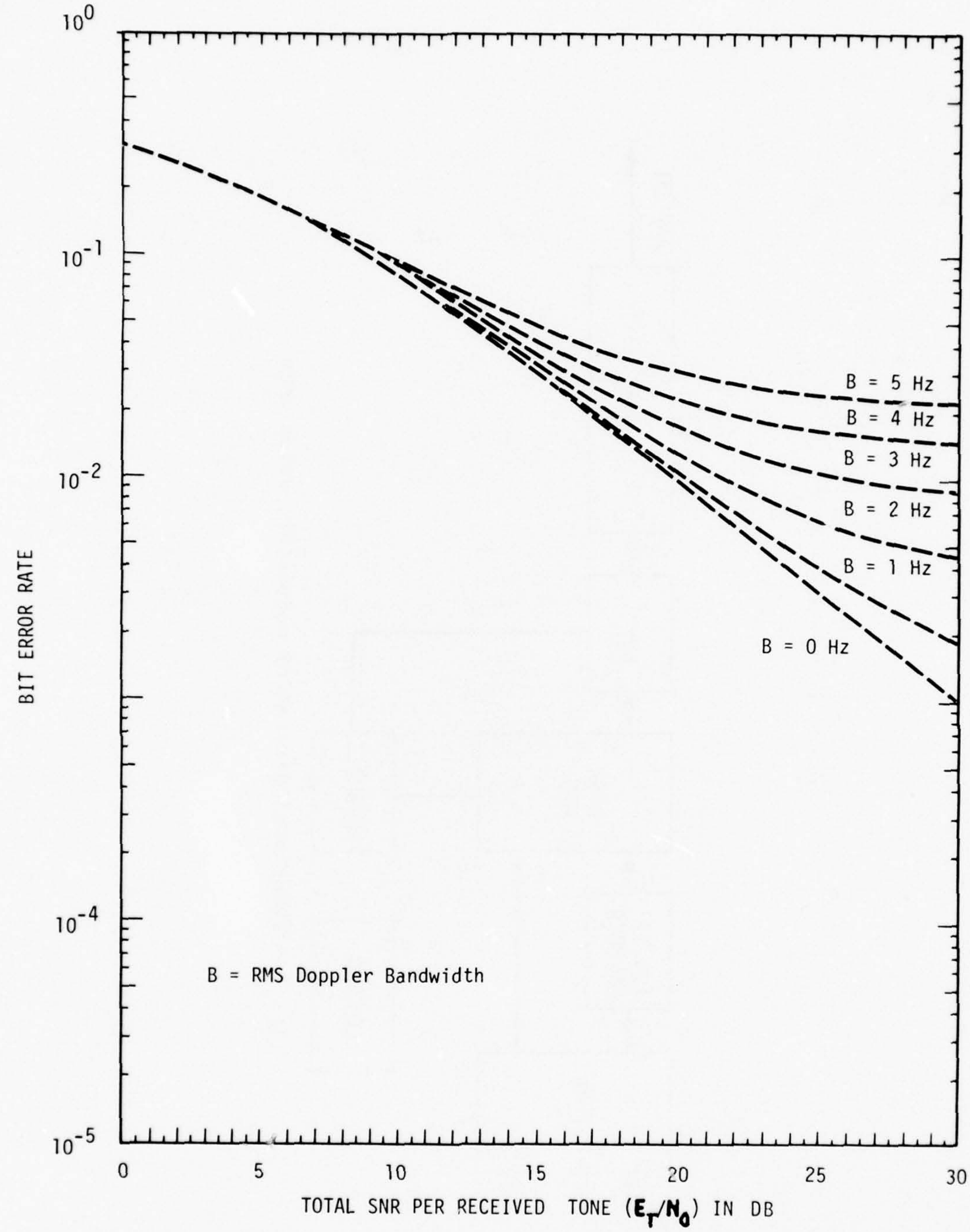


Fig. 2 - Performance of 4-phase DPSK modulation over a time-varying Rayleigh channel (Frame rate = 75 frames/Sec.)

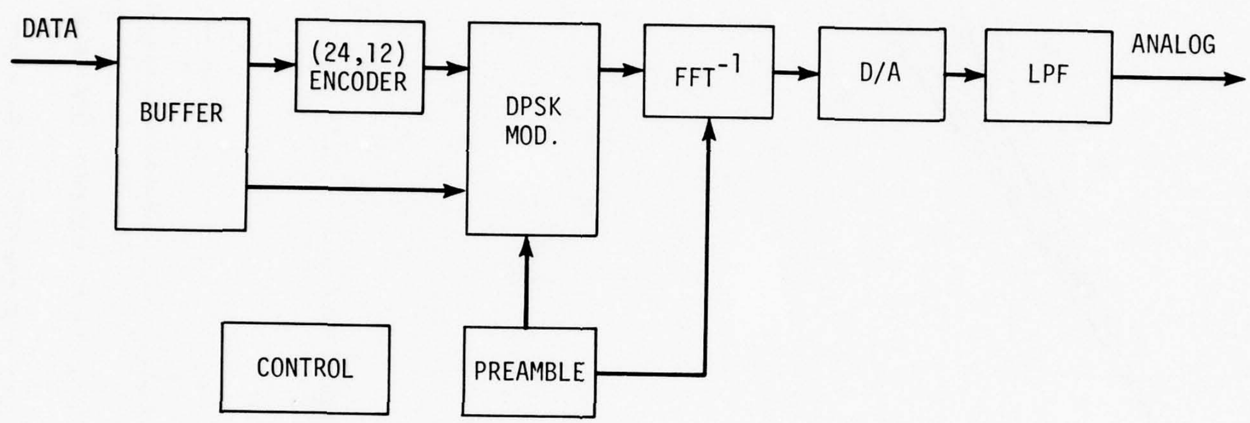


Fig. 3 - Functional diagram of modulator, voice mode

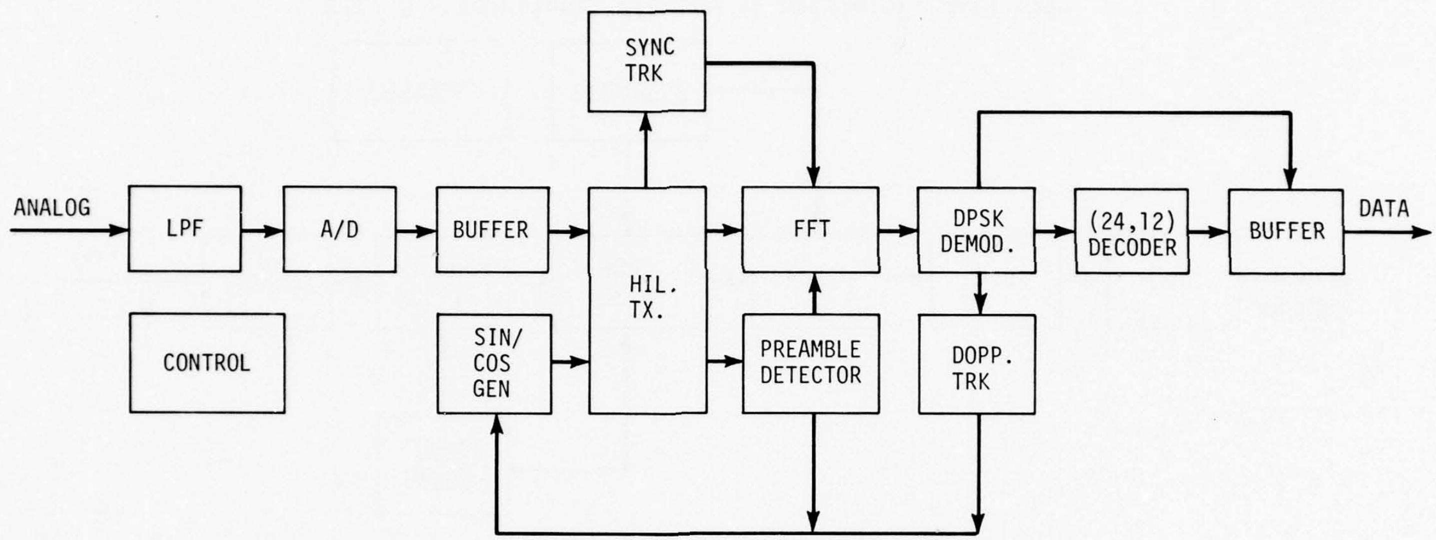


Fig. 4 - Functional diagram of demodulator, voice mode

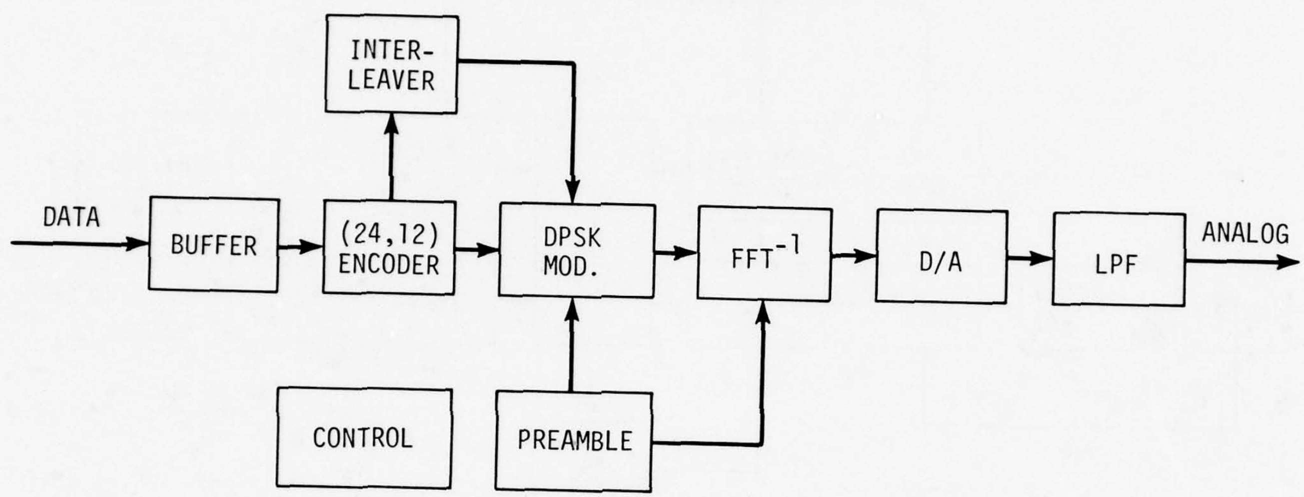


Fig. 5 - Functional diagram of modulator, data mode

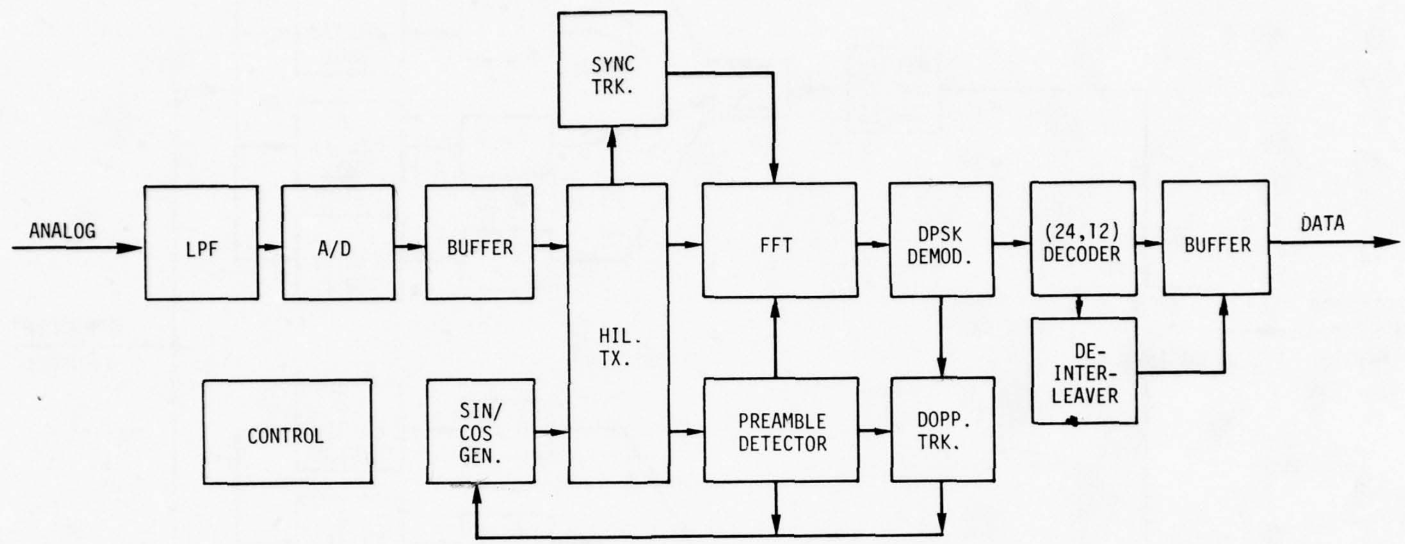


Fig. 6 - Functional diagram of demodulator, data mode

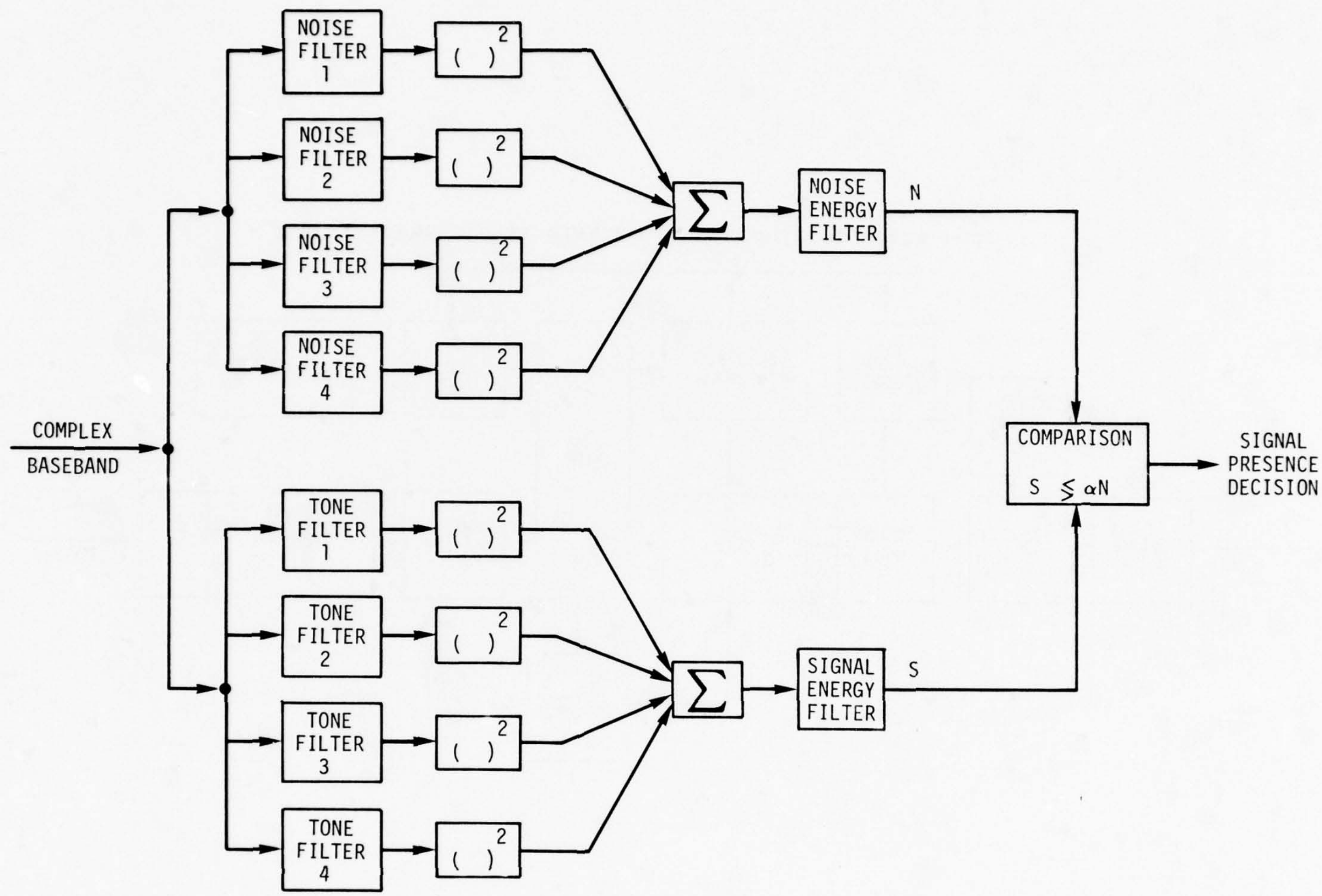


Fig. 7 - Signal presence detection

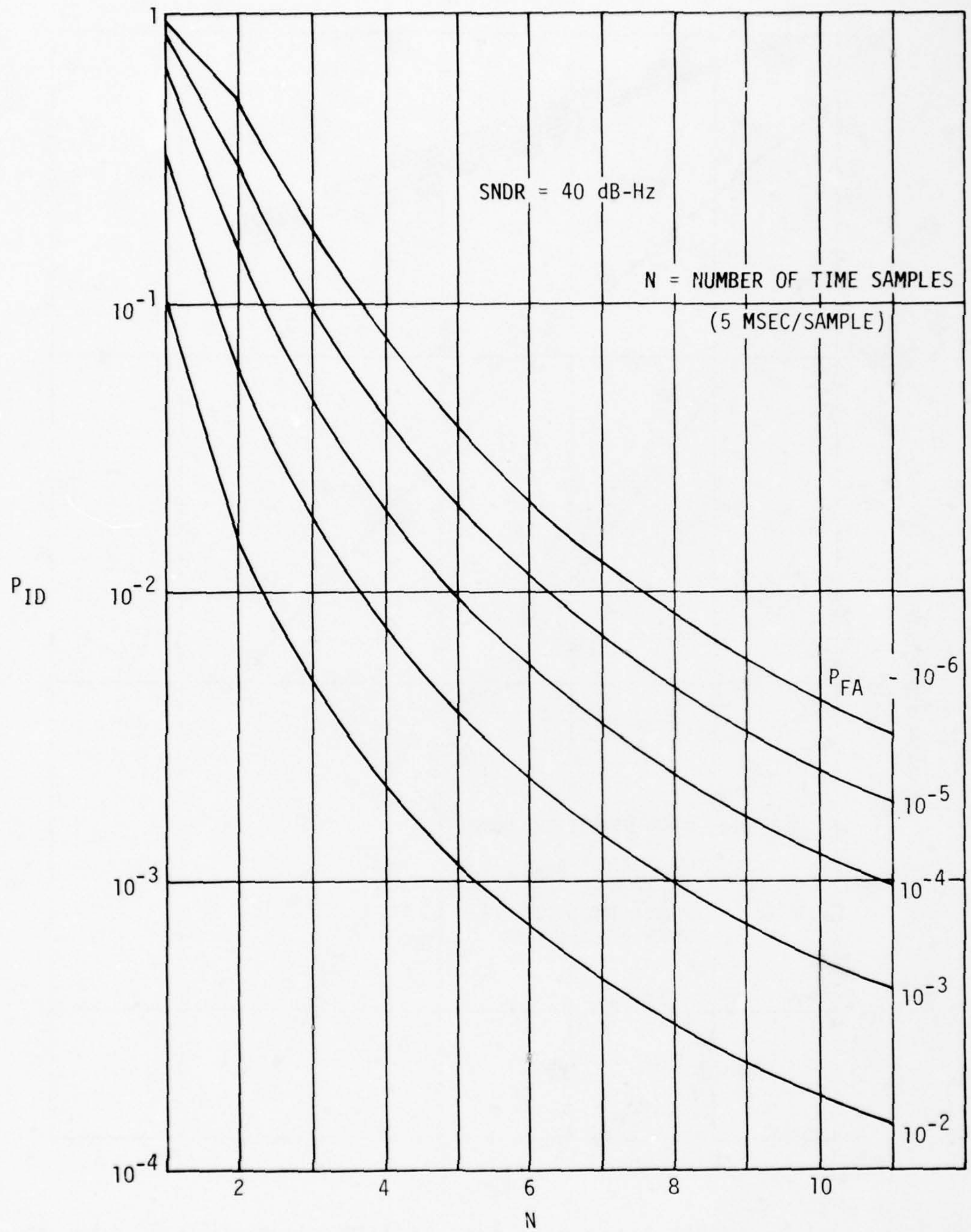


Fig. 8 - Probability of incorrect dismissal
(Four independently fading tones)

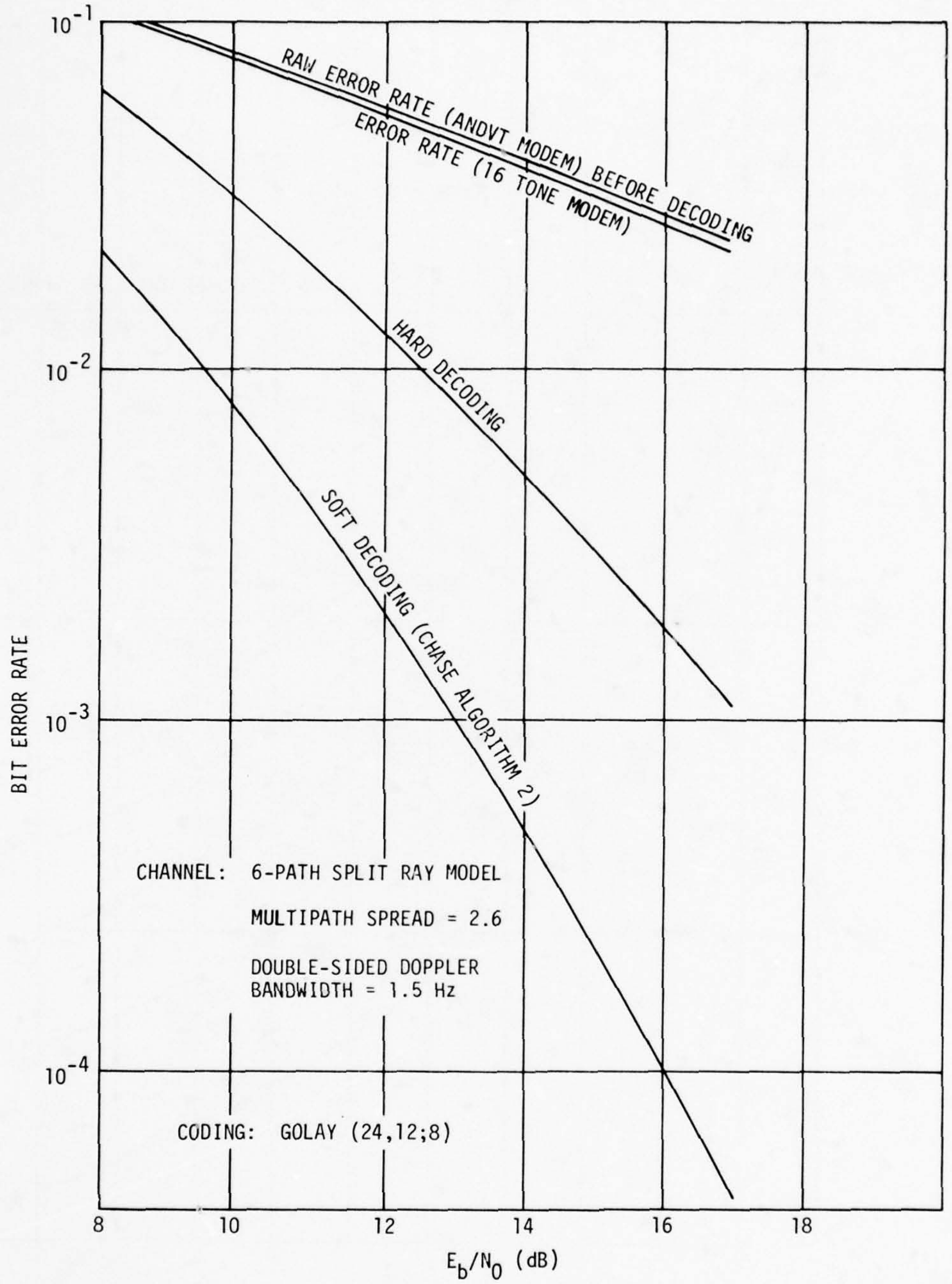


Fig. 9 - Simulated bit error rate for the ANDT modem and a 16 tone modem for a 6 path channel model

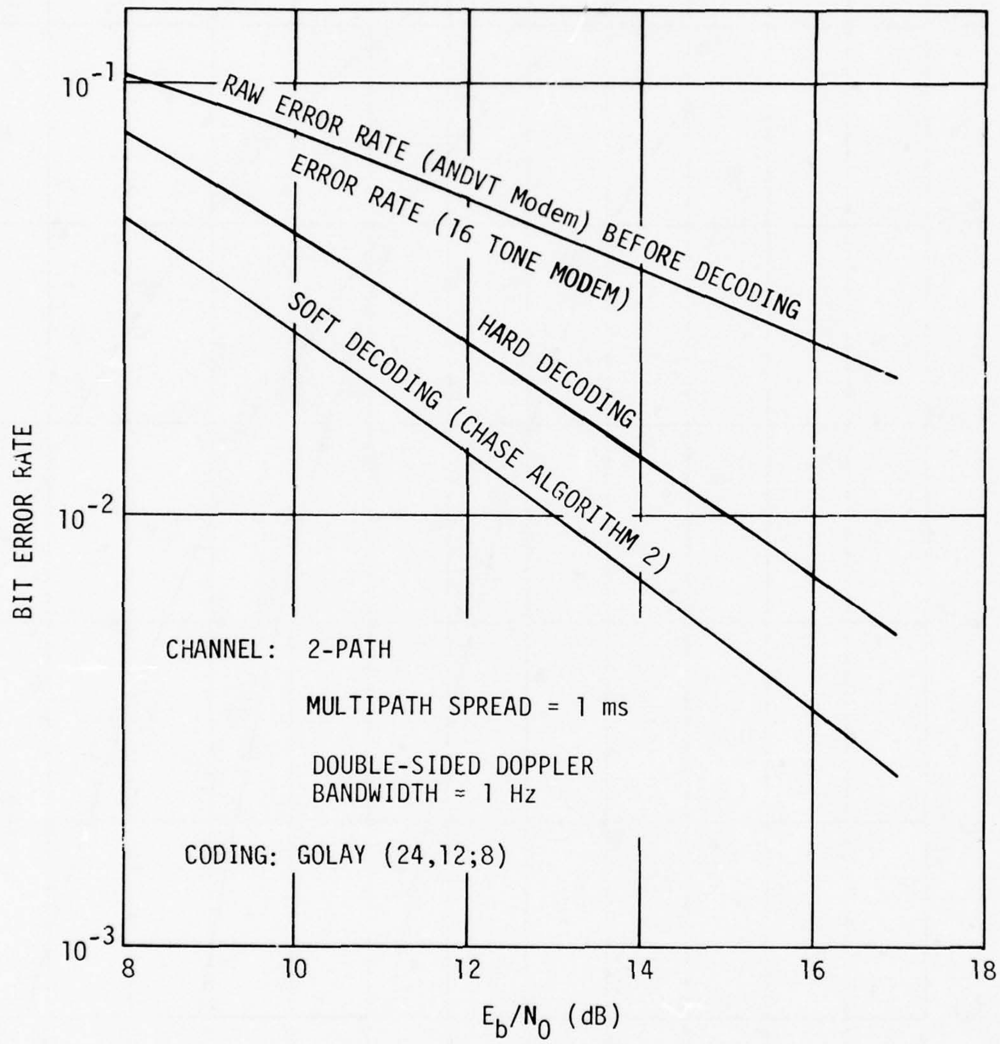


Fig. 10 - Simulated bit error rates for the ANDVT modem and a 16 parallel tone modem (2-path model, Golay code)

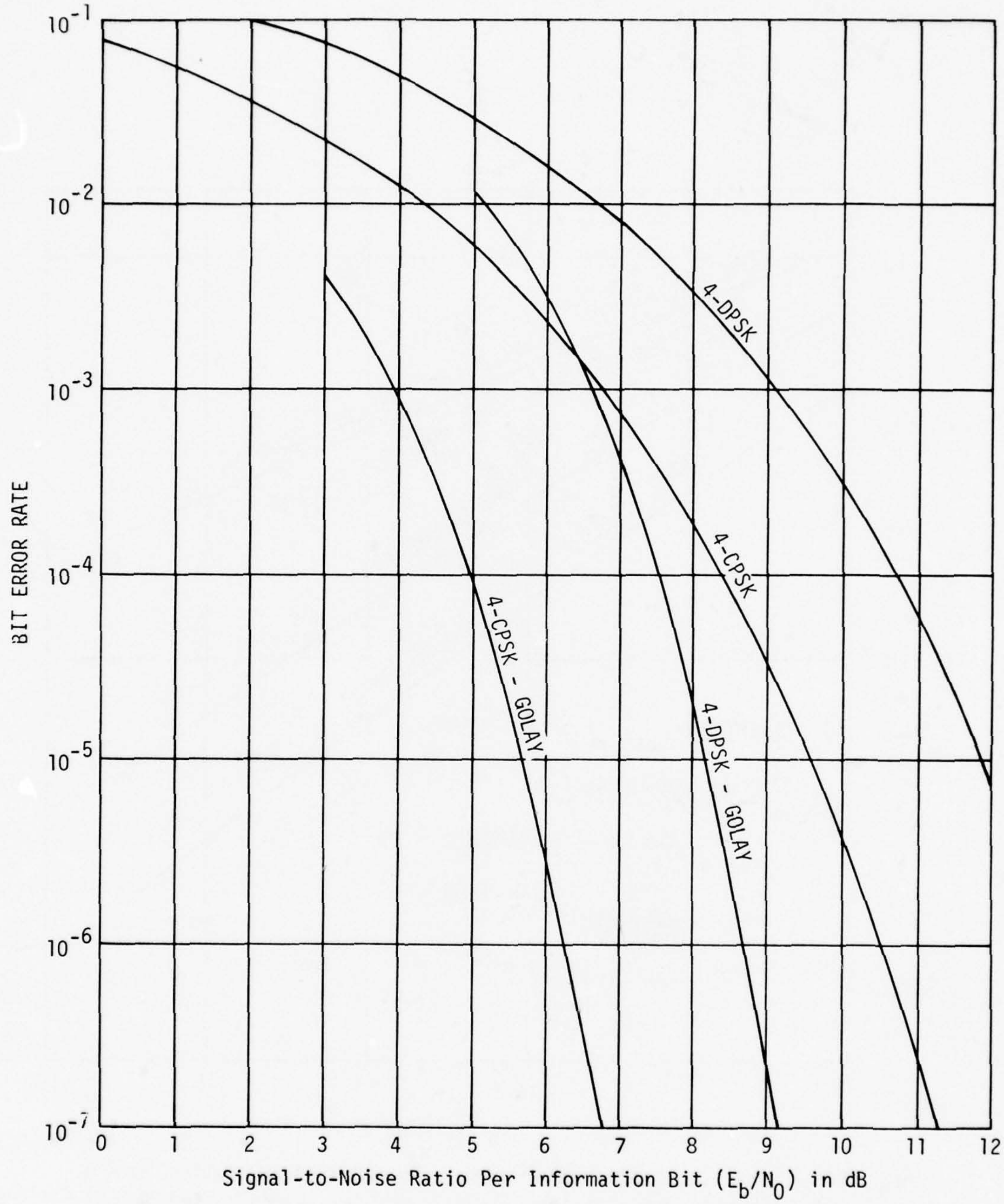


Fig. 11 - Performance of Golay (24,12;8) code over Gaussian channel (soft-decoding, 4-phase modulation)

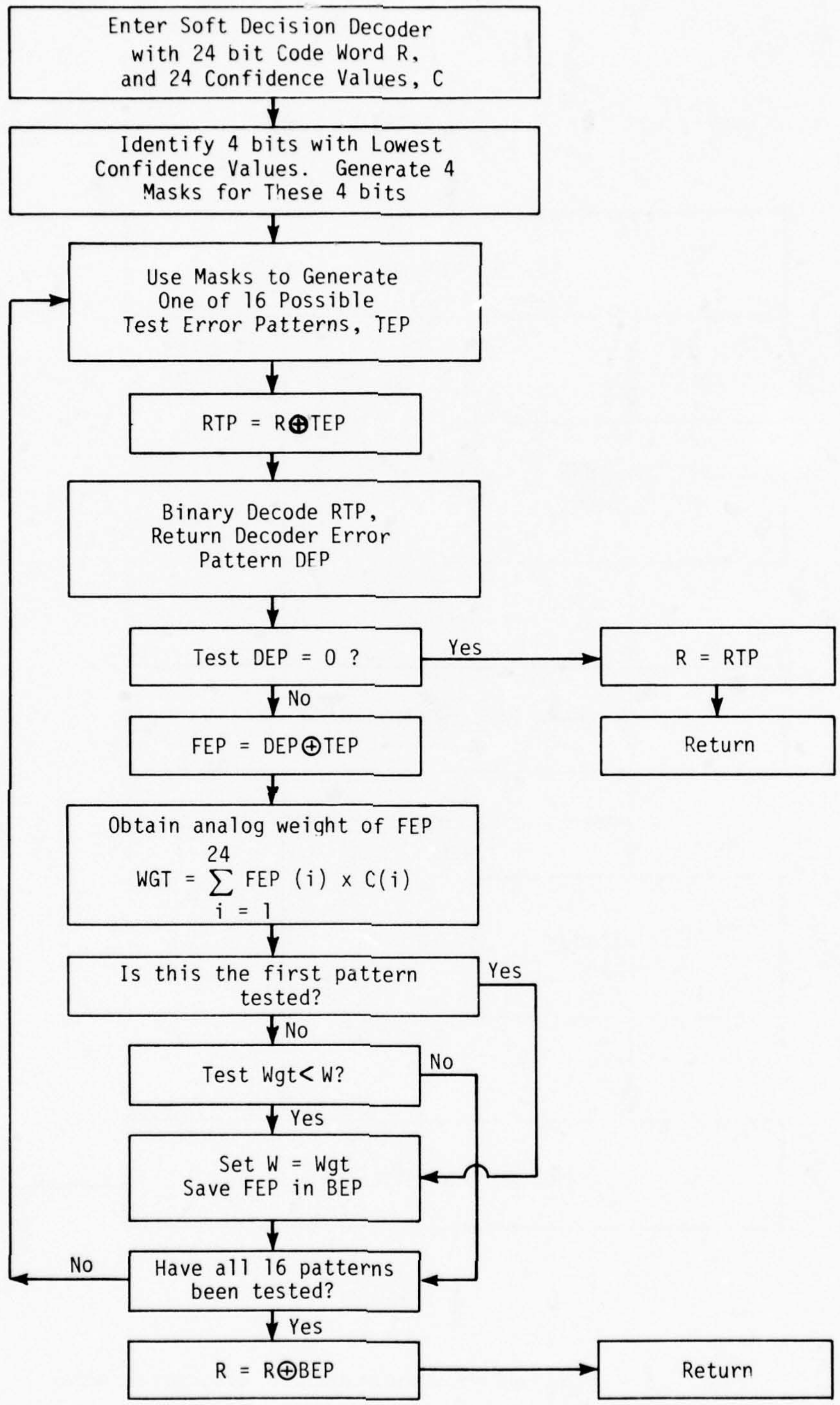


Fig. 12 - Soft decision decoding Algorithm for (24, 12) code

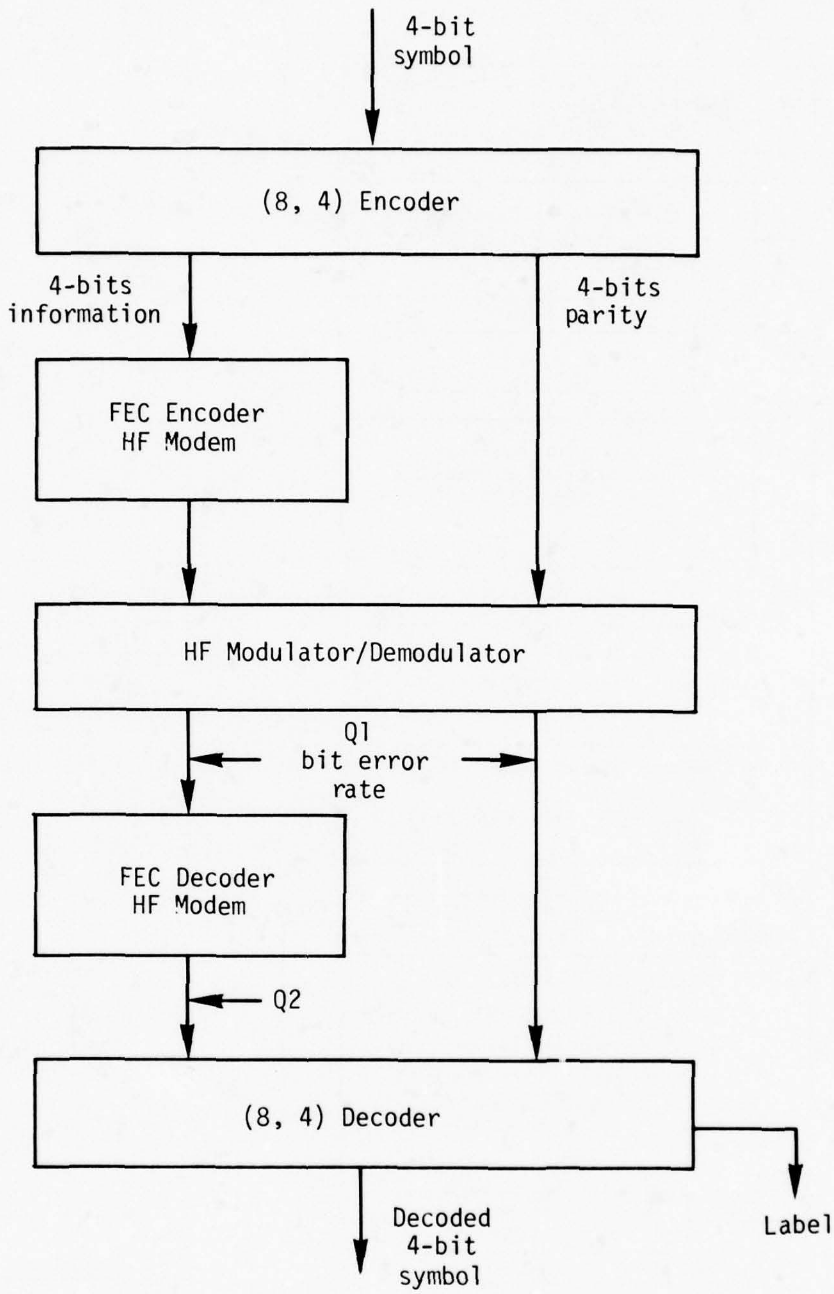


Fig. 13 - Diagram of concatenated coding scheme

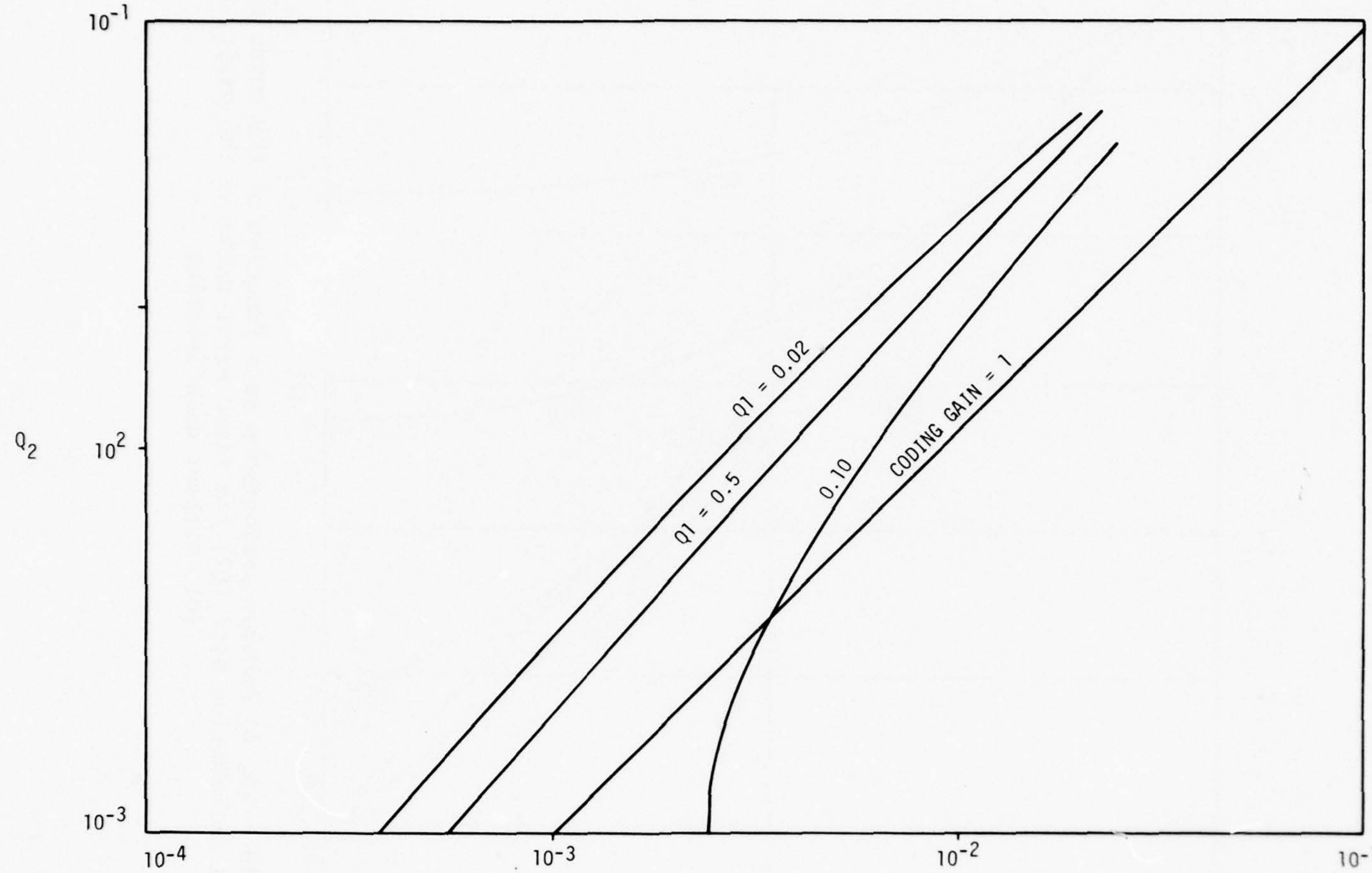


Fig. 14 - (8, 4) decoder performance as a function of the error rate in the information bits (Q2) for fixed error rates on the parity bits (Q1) with data labeling

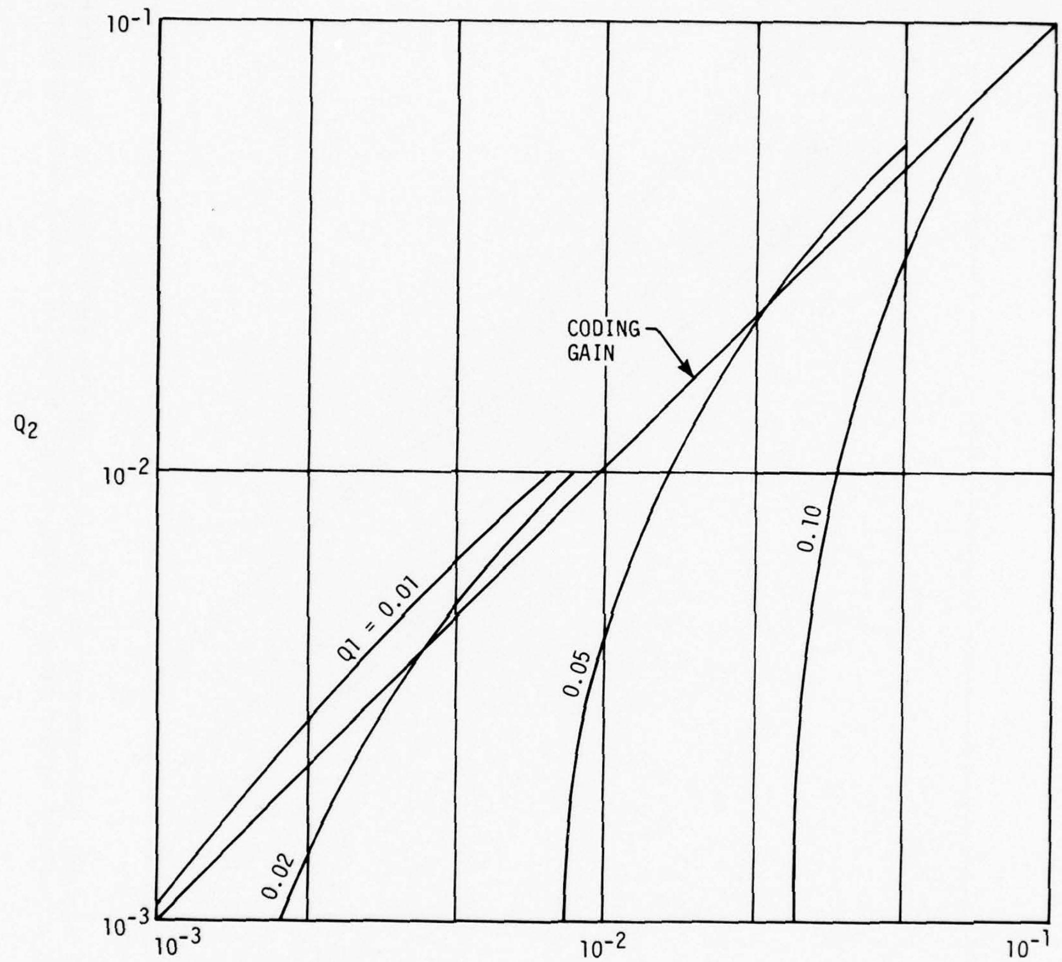


Fig. 15 - (8, 4) decoder performance as a function of the error rate on the information bits (Q2) for fixed error rates on the parity bits (Q1) without data labeling

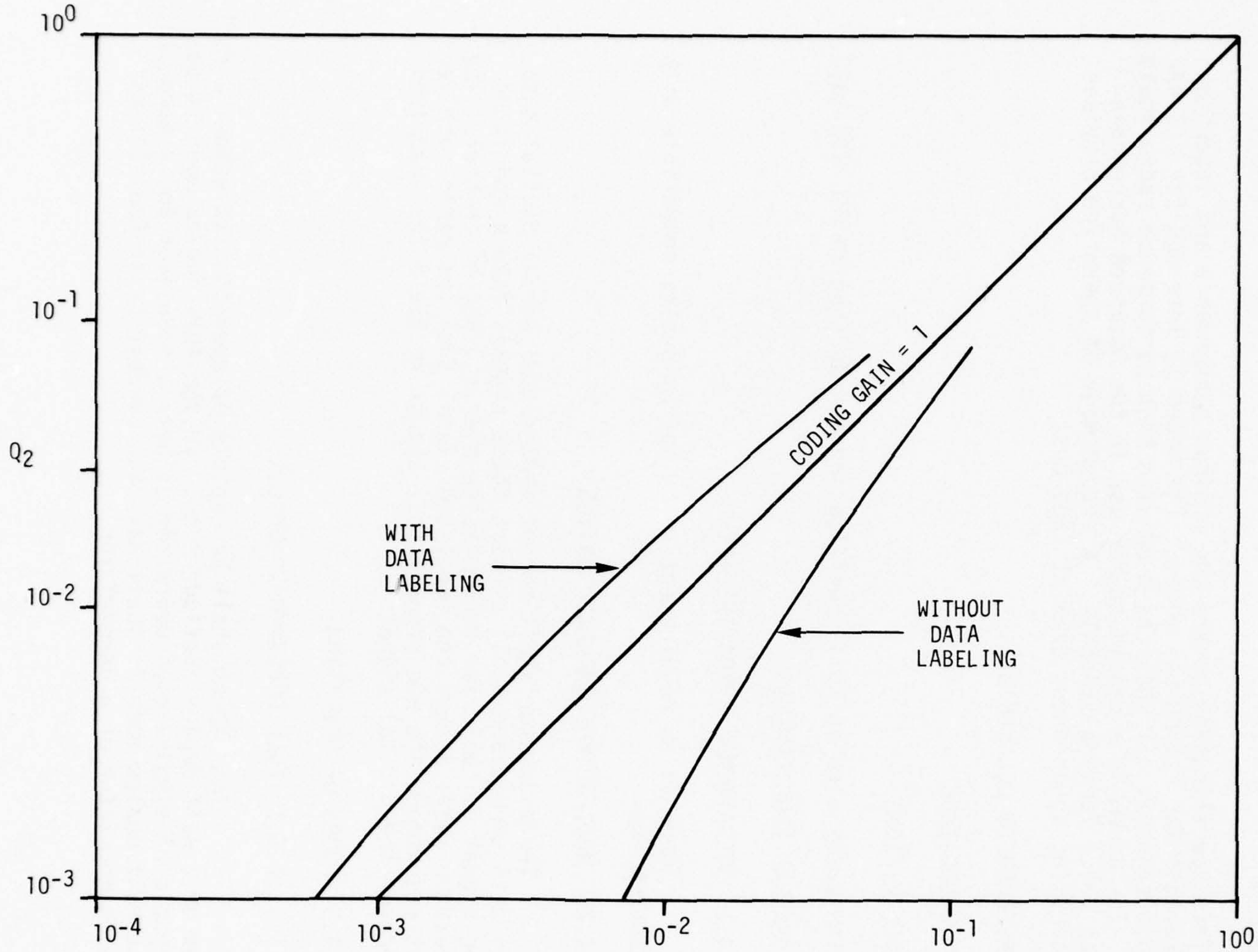


Fig. 16 - (8, 4) decoder performance as a function of the error rate on the information bits (Q2) for Golay decoding with soft decisions on a simulated HF channel

APPENDIX A

CANDIDATE SPECIFICATION FOR HF SECURE VOICE MODEM

1.0 SCOPE

This specification covers the minimum performance and interface requirements for a 2400 bps modem. The modem is intended for a 3 kHz HF radio channel. It shall be used with high performance radio equipment. The modem is a candidate for use in the Advanced Narrowband Digital Voice Terminal (ANDVT). A second mode of operation provides for accepting synchronous data at 1200 bps.

2.0 REFERENCED DOCUMENTS

MIL-STD-188C

3.0 DEFINITIONS

The terms used in this specification are defined in MIL-STD-188C.

4.0 GENERAL REQUIREMENTS

4.1 OPERATIONAL COMPATIBILITY

There is no requirement to be operationally compatible with any other modem.

4.2 INPUT/OUTPUT CHARACTERISTICS

The modulator shall accept synchronous serial digital data from a data source and shall convert those signals into a modulated analog signal suitable for transmission over a 3 kHz HF channel. The demodulator shall accept the modulated signal from the radio receiver output and demodulate and process this signal to the extent required to recover the digital signal.

4.3 MODES OF OPERATION

4.3.1 FULL DUPLEX/HALF DUPLEX

This modem shall be capable of operation in either a full duplex or a half duplex configuration. In the full duplex mode it shall be capable of simultaneous operation as both a modulator and a demodulator. In the half duplex mode it shall have the capability to function as either a modulator or a demodulator.

4.3.2 DIGITAL VOICE/DATA

In the digital voice mode the modem shall be capable of accepting digital information at 2400 bps at a frame rate of 44.44 frames per second. In the data mode it shall accept one synchronous serial digital signal at a rate of 1200 bps.

4.3.3 DIVERSITY/NON-DIVERSITY

In the diversity mode the demodulator shall accept inputs from two 3 kHz channels. This may be a form of RF diversity or antenna diversity. In the non-diversity mode the demodulator interfaces to a single 3 kHz channel.

4.3.4 INTERNAL TIMING/EXTERNAL TIMING

The modulator shall be capable of using digital clock signals that are either generated in the modem or are provided by the data source.

5.0 DETAILED REQUIREMENTS

5.1 MODULATOR CHARACTERISTICS

5.1.1 DIGITAL DATA INPUT

In the digital voice mode the modulator shall be capable of accepting data serially at 2400 bps in frames of 54 bits. The modulator shall process this data frame by frame. In the data mode, the modulator shall be capable of accepting one digital data signal serially at 1200 bps. This shall not be frame synchronous data. The digital input signal and the input circuits shall conform to MIL-STD-188C low level interface specifications.

5.1.2 ERROR ENCODING

5.1.2.1 ENCODING FOR VOICE MODE

In the digital voice mode the modulator shall provide for the encoding of 24 bits out of each 54 bit data frame with the Golay (24, 12) error correction code. Two words of 12 bits shall be encoded into two 24 bit words. The information to be encoded shall be selected by a table look-up procedure which shall permit bit identity to be established after programming is complete. The 48 bits of encoded data and the 30 bits of unencoded data shall make up the total of 78 bits of data to transmit each modem frame period. The assignment of bits to the modulator data tones shall utilize a table look-up procedure which insures that:

- No two bits from a given code word are assigned to the same data tone.
- That the encoded bits of each code word are dispersed over the entire frequency band of the tone library.
- That the tone assignments are continuously changed from frame to frame to minimize the effects of narrowband interference.

5.1.2.2 ENCODING FOR DATA MODE

In the data mode the modulator shall accept digital data at 1200 bps and encode all bits using the Golay (24, 12) code. The encoded data shall be interleaved over N code words, where N is a multiple of four. The amount of interleaving shall be based on the capabilities of the processor. A goal would be interleaving over 200 code words.

5.1.3 MODULATION TECHNIQUE

The modulation technique shall be differentially coherent phase shift keying (DPSK) for each tone as identified in 5.1.4 and shall employ four phase encoding. The transmitted phase at the beginning of a given signalling interval shall be defined as:

$$\theta_n = \theta_{n-1} + \theta_s$$

where

θ_n = signal phase at beginning of present interval

θ_{n-1} = signal phase at beginning of previous interval

θ_s = phase shift due to data.

This definition departs from the MIL-STD-188C definition for DPSK by removing the necessity to update each phase reference by the phase shift that occurs during the time guard period. The phase shift due to the data shall be determined as follows:

dibit		phase shift (degrees)
odd bit	even bit	
0	0	-225
0	1	-135
1	0	-315
1	1	-45

5.1.4 TONE LIBRARY

5.1.4.1 VOICE MODE LIBRARY

In the digital voice mode the modulator shall transmit 39 DPSK data tones plus one continuous tone for use in automatic frequency correction. The tone frequencies shall be spaced at intervals of 56.25 Hz. The data tones shall consist of the 39 frequency assignments between 843.75 Hz and 2981.25 Hz. The AFC slot shall be centered at 562.5 Hz. No signals shall be transmitted in the four frequency slots between the AFC tone and the lowest frequency data tone. All frequency assignments correspond to those obtained using an 128 point integration period with a sampling frequency of 7200 Hz. The data tones shall be simultaneously keyed at 0.0225 second intervals. This interval is equivalent to 162 periods of the 7200 Hz sampling frequency.

5.1.4.2 DATA MODE LIBRARY

In the data mode the modulator shall transmit 16 DPSK data tones plus two continuous tones for use in automatic frequency correction. The tone frequencies shall be spaced at frequency intervals of 112.5 Hz. The data tones shall consist of the 16 frequency assignments between 900.00 Hz and 2587.5 Hz. The AFC slots shall be centered at 562.5 Hz and 2925.0 Hz. No signals shall be transmitted on the two frequency slots between the lowest AFC slot and the lowest frequency data tone and on the two frequency slots between the highest AFC slot and the highest frequency data tone. The frequency assignments correspond to those obtained using a 64 point integration period with a sampling frequency of 7200 Hz. The data tones shall be simultaneously keyed at 0.0133 second intervals. This interval is equivalent to 96 periods of the 7200 Hz sampling frequency.

5.1.5 MODULATOR SIGNAL OUTPUT LEVEL

The composite output signal level shall be zero dBm for all modes of operation. It shall be zero dBm during the modem preamble period. The individual modulated tones shall have a maximum spread, between the highest and the lowest tone level, of not more than 1.5 dB. The level of the AFC tones shall be 6 dB above the average level of the individual data tones.

5.1.6 ANALOG OUTPUT CIRCUITS

The digitally generated signals shall be converted to analog using a 12 bit converter and a sampling frequency of 7200 Hz. The output signal shall be passed through a bandpass filter to remove any dc component and to provide at least 40 dB attenuation at 3600 Hz. The envelope delay distortion with reference to the delay at 900.0 Hz shall not exceed 0.25 milliseconds over the range 600 Hz to 3100 Hz. The output circuit shall conform to that specified in MIL-STD-188C (7.3.5.5).

5.2 DEMODULATOR CHARACTERISTICS

5.2.1 ANALOG INPUT SIGNALS

The demodulator input signals shall be the DPSK signals generated by the modulator and modified by passage through the transmission channel. The demodulator shall accept this signal, synchronize the internal timing to the received signal, demodulate and decode the signal, and produce digital signals at the corresponding information rate.

5.2.2 ANALOG INPUT CIRCUITS

The analog input circuits shall conform to the specifications given in MIL-STD-188C (7.3.5.5). The input signals shall be bandpass filtered prior to digitalization. The filter shall meet the requirements given in 5.1.6. A 12 bit A/D with a sampling frequency of 7200 Hz shall be used to convert the analog signal to digital form. It shall accept without distortion the modulator output signal at zero dBm composite signal level.

5.2.3 DOPPLER CORRECTION

5.2.3.1 DOPPLER CORRECTION IN THE HALF DUPLEX DIGITAL VOICE MODE

In the half duplex digital voice mode the demodulator shall have the capability of using the 562.5 Hz doppler tone to correct frequency translations up to ± 75 Hz. The AFC algorithm shall utilize the estimation of the frequency error obtained during the modem preamble period as described in 5.4.2. It shall have the capability of tracking the frequency offset at a maximum rate of 3.5 Hz per second. The algorithm shall provide for not using the information obtained when the doppler tone is faded below an adaptable threshold.

5.2.3.2 DOPPLER CORRECTION IN ALL OTHER MODES

In all modes other than half duplex digital voice it is assumed that the transmission period will be relatively long. Thus, the doppler correction technique shall be adaptable to maximize system performance. It shall include use of performance data obtained on the data tones. In addition, in the data mode a second doppler tone at 2925 Hz is also transmitted. The doppler correction algorithm in the data mode shall make optimum use of both tones in a frequency selective fading condition.

5.2.4 SYNCHRONIZATION

In the half duplex mode the modem shall initiate a synchronization preamble as described in 5.4 During the reception of the data tones the modem shall, in the voice mode, maintain/acquire synchronization by sampling the signal energy in the first frequency slot below the lowest data tone. This corresponds to the 787.5 Hz slot. In the data mode, the synchronization algorithm shall cause the modem to maintain/acquire synchronization by sampling the signal energy in either one or both of the first frequency slots below the lowest data tone and above the highest data tone. This corresponds to the 787.5 Hz slot and the 2700 Hz slot. In all modes of operation the demodulator synchronization tracking shall be inhibited when the data tones signal to noise level drops below a threshold. The synchronization tracking rate shall not be faster than that equivalent to a shift of 1/7200 second every two frame periods.

5.2.5 FREQUENCY DETECTION

The detection of the data and the doppler tones shall be performed using a digital FFT. A 64 point transform shall be used for the data mode and a 128 point transform shall be used for the digital voice mode.

5.2.6 DIVERSITY COMBINING

The signal combining shall be accomplished by a method that provides optimum results with fading DPSK signals.

5.2.7 DIFFERENTIAL PHASE DETECTION

The differential phase detection shall be based on a measurement of the phase shift, or its equivalent, between two successive signalling intervals. The phase reference shall not be updated by the phase shift that occurs during the time guard period.

5.2.8 ERROR DECODING

5.2.8.1 ERROR DECODING FOR VOICE MODE

The two Golay (24, 12) code words that are received each frame period shall be decoded using a soft decision algorithm. Identity of the bit positions for the coded and uncoded data shall be determined by a table look-up procedure similar to that described for the encoder in 5.1.2.1.

5.2.8.2 ERROR DECODING FOR DATA MODE

The detected signal shall be buffered through a de-interleaver to recover the encoded Golay code words. The size of the de-interleaver shall correspond to the size of the interleaver used in the modulator. The choice of using either hard or soft decision decoding shall be based on implementation implications.

5.2.9 DIGITAL DATA OUTPUT

The digital output signal and the output circuits shall conform to MIL-STD-188C low level interface specifications.

5.3 TIMING REQUIREMENTS

The modem shall provide timing at the following rates depending on the mode of operation:

Mode	Modulator		Demodulator	
	Tx. Data	Tx. Frame	Rx. Data	Rx. Frame
HD DV	2400 Hz	44.44 Hz	2400 Hz	44.44 Hz
FD DV	2400 Hz	44.44 Hz	2400 Hz	44.44 Hz
HD DATA	1200 Hz	--	1200 Hz	--
FD DATA	1200 Hz	--	1200 Hz	--

In addition, the modulator shall be capable of accepting an external serial data clock signal in all modes of operation and an external frame clock signal in the digital voice mode. The demodulator shall always supply the clock signals for the digital output signal. The digital timing circuits shall conform to MIL-STD-188C low level interface specifications. The synchronization timing changes made in the demodulator framing shall be proportionally divided over all cycles of the serial data clock output. The stability of the basic oscillator from which the digital clock signals are derived shall be at least 1 part in 10^6 per day.

5.4 MODEM PREAMBLE

5.4.1 MODULATOR PREAMBLE SIGNAL

The modulator shall be capable of generating a two-part modem preamble for the purpose of establishing a communication link. The shifting between the transmission of the normal modem signal to the transmission of the modem preamble shall be in accordance with the requirements outlined in 5.6. The first part of the modem preamble shall consist of three unmodulated tones centered at frequencies of 562.5 Hz, 1462.5 Hz and 2250.0 Hz. The amplitude of the tones shall be selected to give a composite output level of zero dBm. The unmodulated tones shall be transmitted for an integer number of frame periods up to 32 frames. The second part of the modem preamble shall consist of three modulated tones centered at frequencies of 787.5 Hz, 1800.0 Hz and 2700.0 Hz. These tones shall be phase shifted 180 degrees at the signalling rate. The amplitude of the tones should result in a composite output level of zero dBm. The modulated tones shall be transmitted for an integer number of frame periods up to 32.

5.4.2 DEMODULATOR MODEM PREAMBLE FUNCTIONS

The demodulator shall use the first part of the modem preamble to recognize signal presence and to estimate the frequency translation error. The second part of the modem preamble shall be used to establish frame synchronization.

5.5 SYSTEM PREAMBLE

Following the transmission of the modem preamble, the modem shall transmit a single phase reference frame. This signal shall consist of all data and AFC tones. The phases of the tones shall be randomized to minimize the peak level of the output signal. Following the reference frame the modulator shall transmit a system synchronization preamble. The length of the preamble shall be a variable from zero to 256 bits. The manner in which this preamble shall be encoded depends on whether the modem is in the digital voice mode or the data mode.

5.5.1 ENCODING OF SYSTEM PREAMBLE IN DIGITAL VOICE MODE

In the digital voice mode the modem shall encode the entire system preamble with the Golay (24, 12) code. If the length of the preamble is not a multiple of twelve, the modem shall stuff zeros into the encoder to obtain an integer number of code words. The encoded data repeated five times does not fill an integer number of modem frames, the remaining bit positions shall be stuffed with zeros.

5.5.2 DECODING OF THE SYSTEM PREAMBLE IN DIGITAL VOICE MODE

In the digital voice mode the modem shall have special provisions for the decoding of the system preamble because it is encoded differently than the digital voice signal. Following the phase reference frame the modem shall demodulate an integer number of modem frames. A fifth order diversity combining technique shall be used to derive the encoded data which shall be decoded using a soft decision decoder. If the system preamble is not an integer number of frames of 54 bits then it must be preceded with zeros to make it so.

5.5.3 ENCODING AND DECODING OF SYSTEM PREAMBLE IN DATA MODE

In the data mode the modem encodes and decodes the system preamble the same as data.

5.6 CONTROL FUNCTIONS

Functional diagrams are provided representing the control requirements for each mode of operation.

5.6.1 CONTROL FOR HALF DUPLEX DIGITAL VOICE

The functional control requirements for half duplex digital voice are shown in Figures 1 and 2. The basic requirements are:

- A push-to-talk (PTT) signal is used to transfer from a receive mode to a transmit mode.
- Once data is detected, PTT cannot be used to interrupt reception.
- Once PTT is activated, the entire preamble sequence must be transmitted.
- PTT is tested once per frame of data.
- As long as PTT is held down, test for receipt of a signal is not made.
- The receive system clock is disabled when in the transmit mode and the transmit system clock is disabled when in the receive mode.
- The transmit system clock is disabled for a short period after reading in the system preamble to permit it to be transmitted with redundancy.
- In the receive mode, the modem must detect both the modem preamble and the reference frame in order to progress into the data loop.
- Data may be detected either one or two frames at a time.
- The receive system clock is enabled when the decoding of the system preamble has progressed to the point where a continuous digital output can be maintained at 2400 bps.
- A prep signal is given to the transmit system prior to enabling the transmit system clock.
- A prep signal is given to the receiver system prior to enabling the receive system clock.

5.6.2 CONTROL FOR FULL DUPLEX DIGITAL VOICE

The functional control requirements for full duplex digital voice are shown in Figures 3 and 4. The basic requirements are:

- A restart flag, which is set by the operator, is used to initiate a transmission of the system preamble.
- Once the restart flag is detected, the transmit modem goes "off the air" briefly and then the transmit cycle is restarted in same manner as in the PTT half duplex mode.
- The loss of modem signal is used by the receive modem to indicate that a system synchronization preamble shall follow. If the receive modem signal is lost temporarily and a modem preamble is not detected, then the return of the modem signal will permit the system to continue because receive system clock was not inhibited.
- Receive system clock is inhibited if modem preamble is detected.

5.6.3 CONTROL FOR HALF DUPLEX DATA MODE

The functional control requirements for half duplex data mode operation are shown in Figures 5 and 6. The basic requirements are:

- Operation is essentially the same as in the half duplex digital voice mode except that the system preamble is treated the same as data.

5.6.4 CONTROL FOR FULL DUPLEX DATA MODE

The functional control requirements for full duplex data mode operation are shown in Figures 7 and 8. The basic requirements are:

- A restart flag signal is used for the same purpose described for full duplex digital voice.
- The system preamble is treated the same as data.

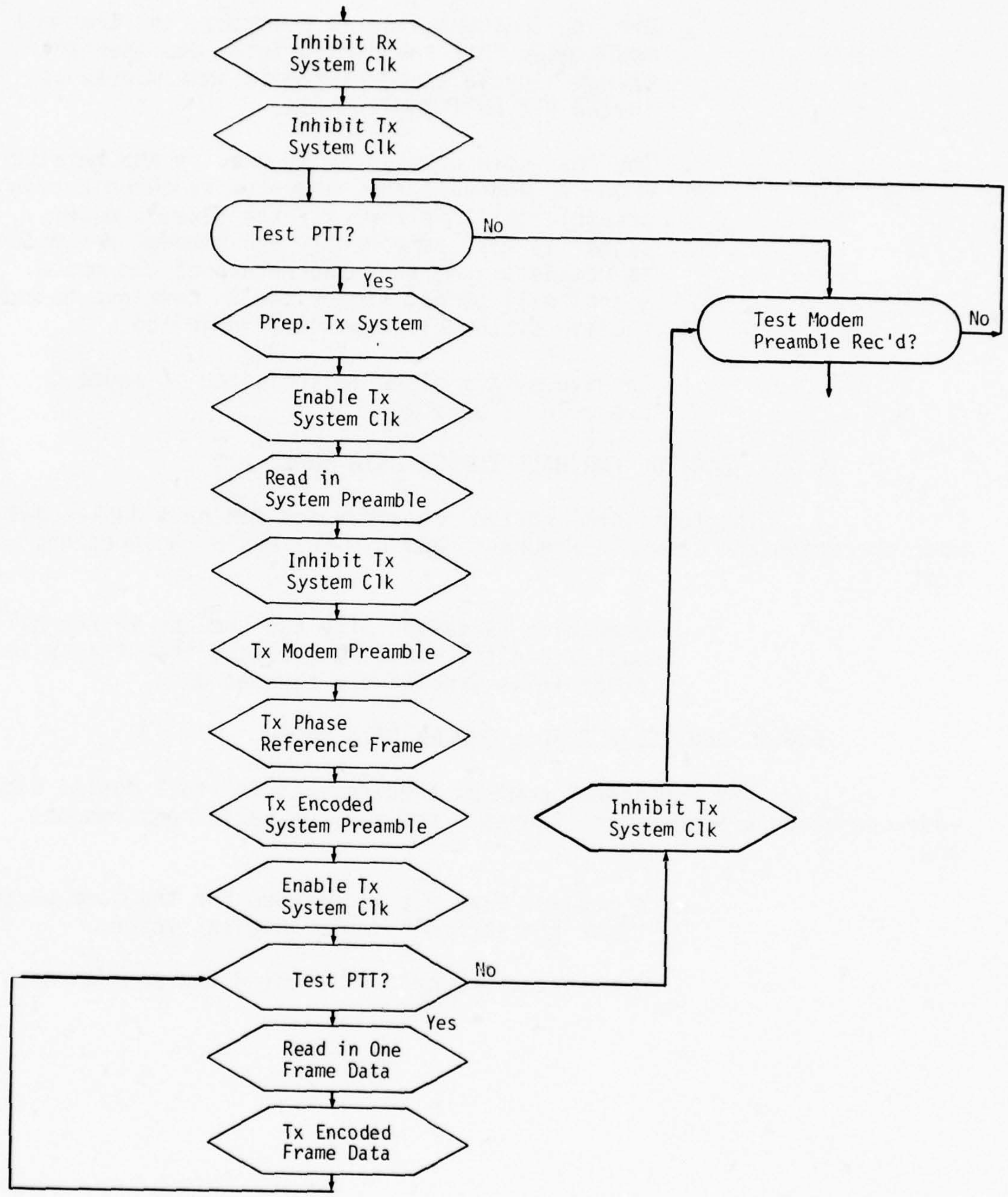
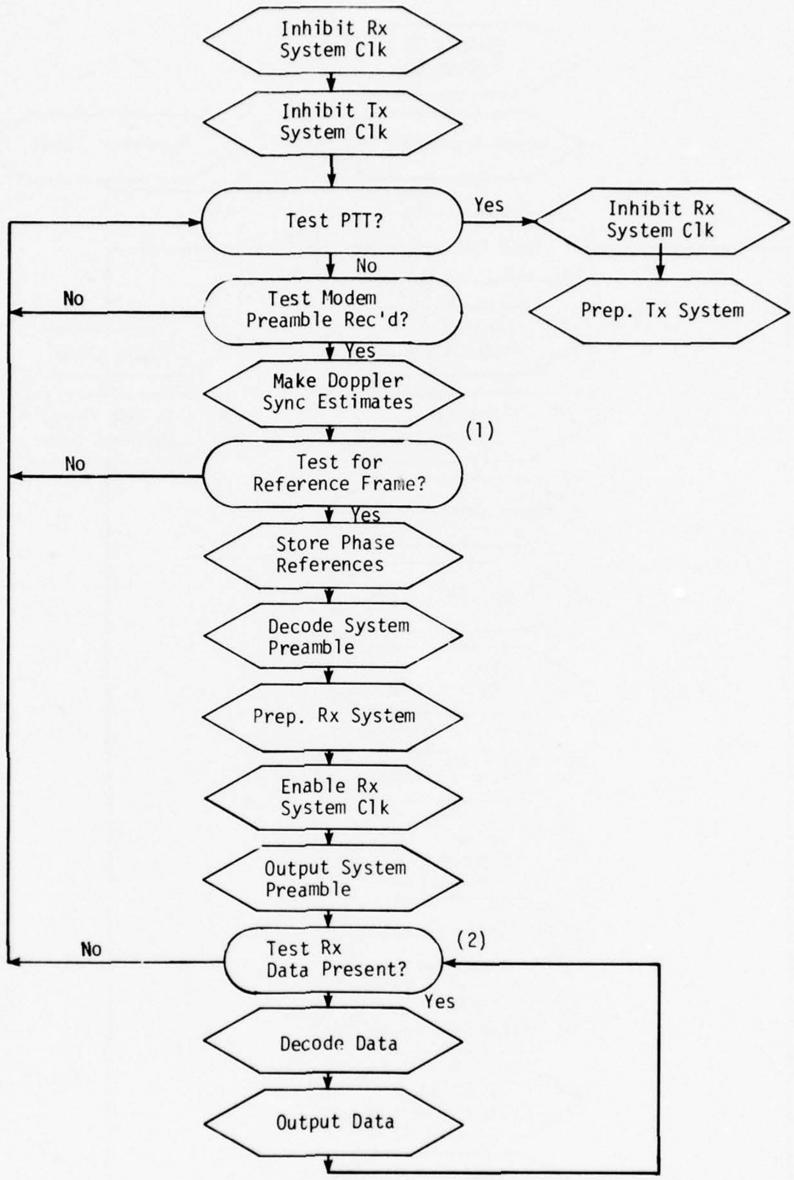


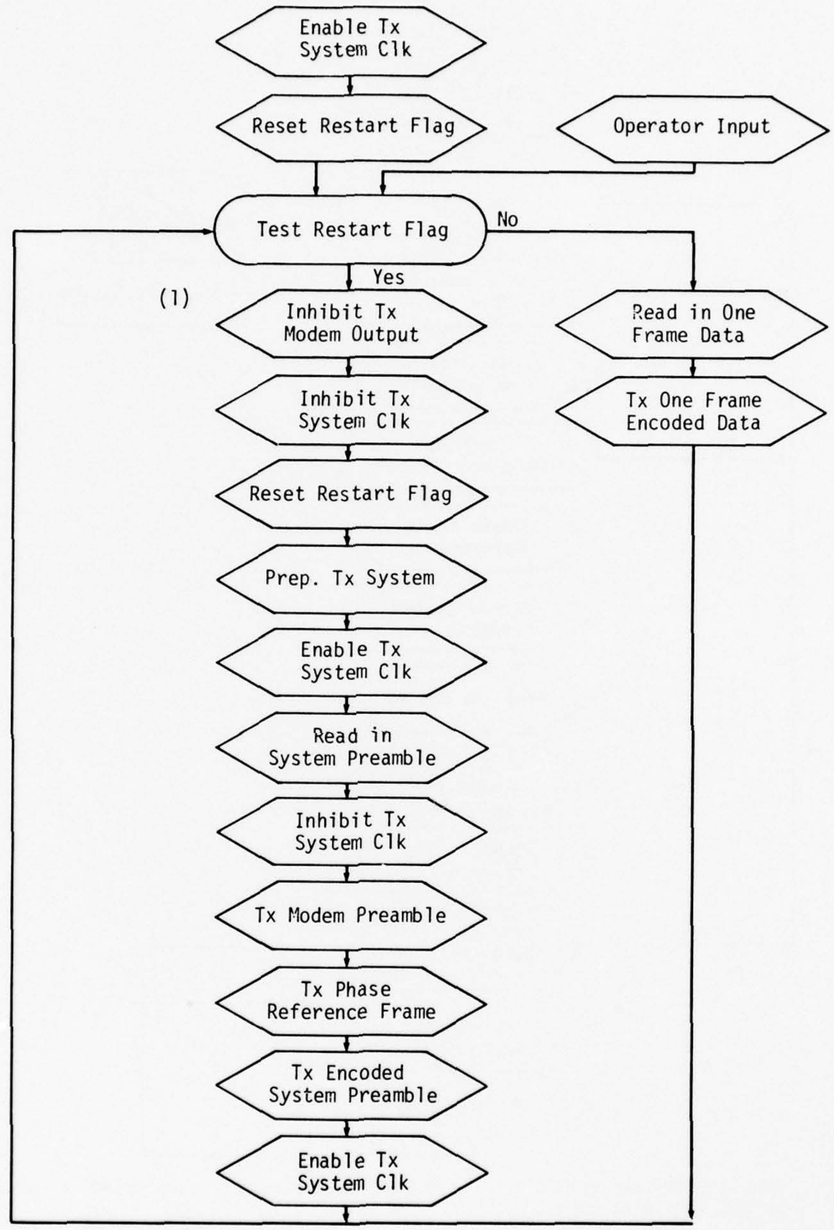
Fig. 1 - Preamble/data functional diagram half duplex, digital voice, transmit



(1) Both the modem preamble and the reference frame must be detected in order to get into data loop.

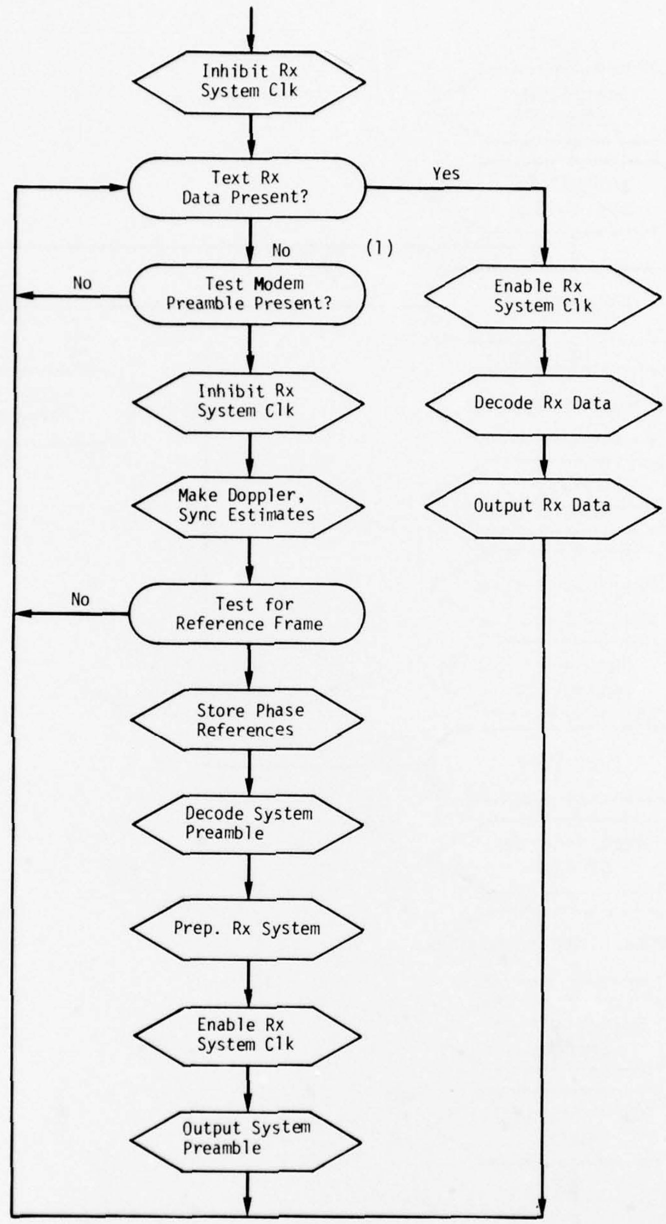
(2) Once data is detected, PTT cannot interrupt reception.

Fig. 2 - Preamble/data functional diagram half duplex, digital voice, receive



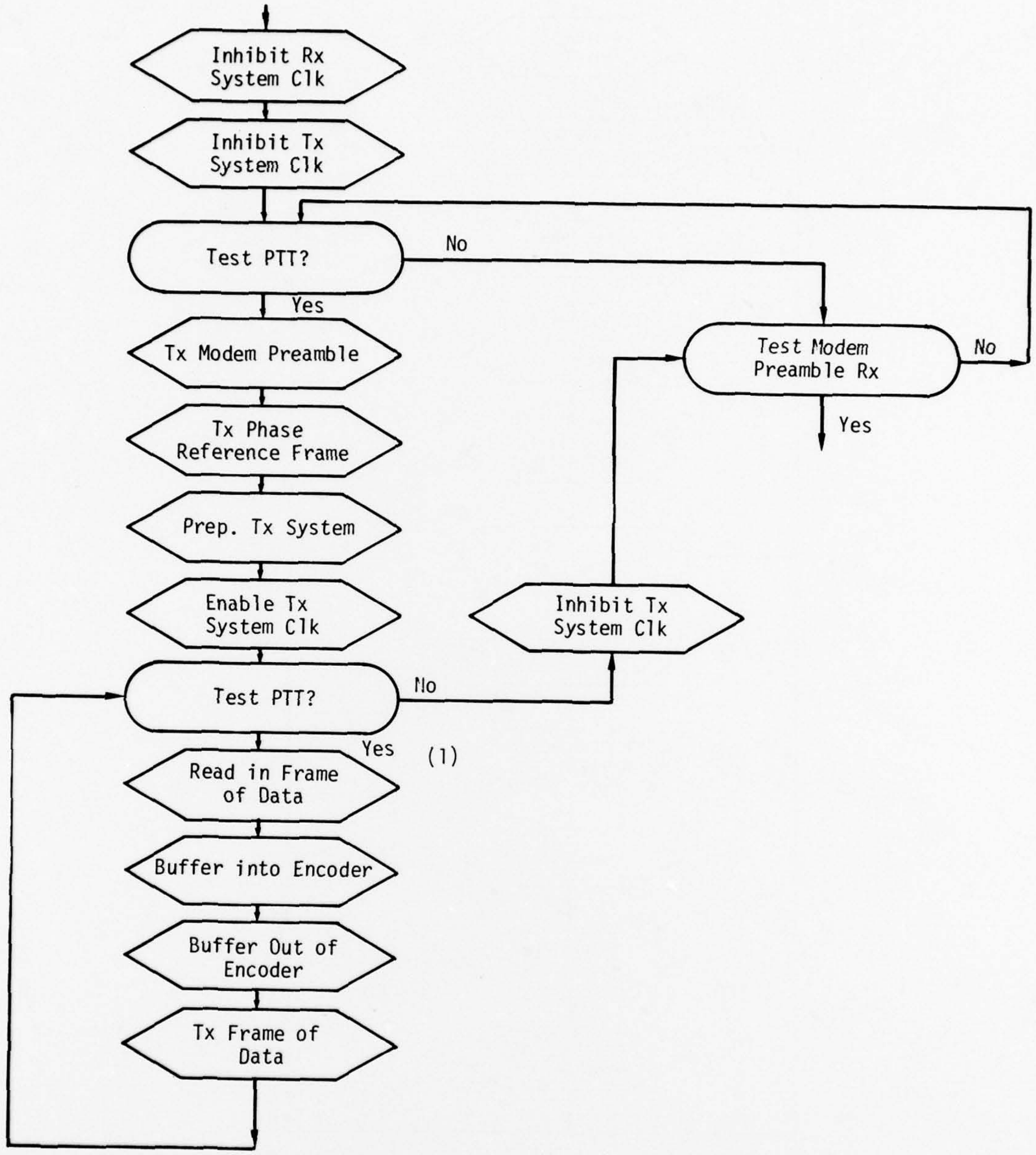
(1) Loss of received signal is used by receiver to identify system preamble. Modem treats system preamble differently than digital voice.

Fig. 3 - Preamble/data functional diagram full duplex, digital voice, transmit



(1) Loss of modem signal used to indicate system sync restart. If signal is lost temporarily and a modem preamble is not detected, then return of modem signal will permit system to continue because system clock was not inhibited.

Fig. 4 - Preamble/data functional diagram full duplex, digital voice, receive



(1) System preamble treated the same as data.

Fig. 5 - Preamble/data functional diagram half duplex, data mode, transmit

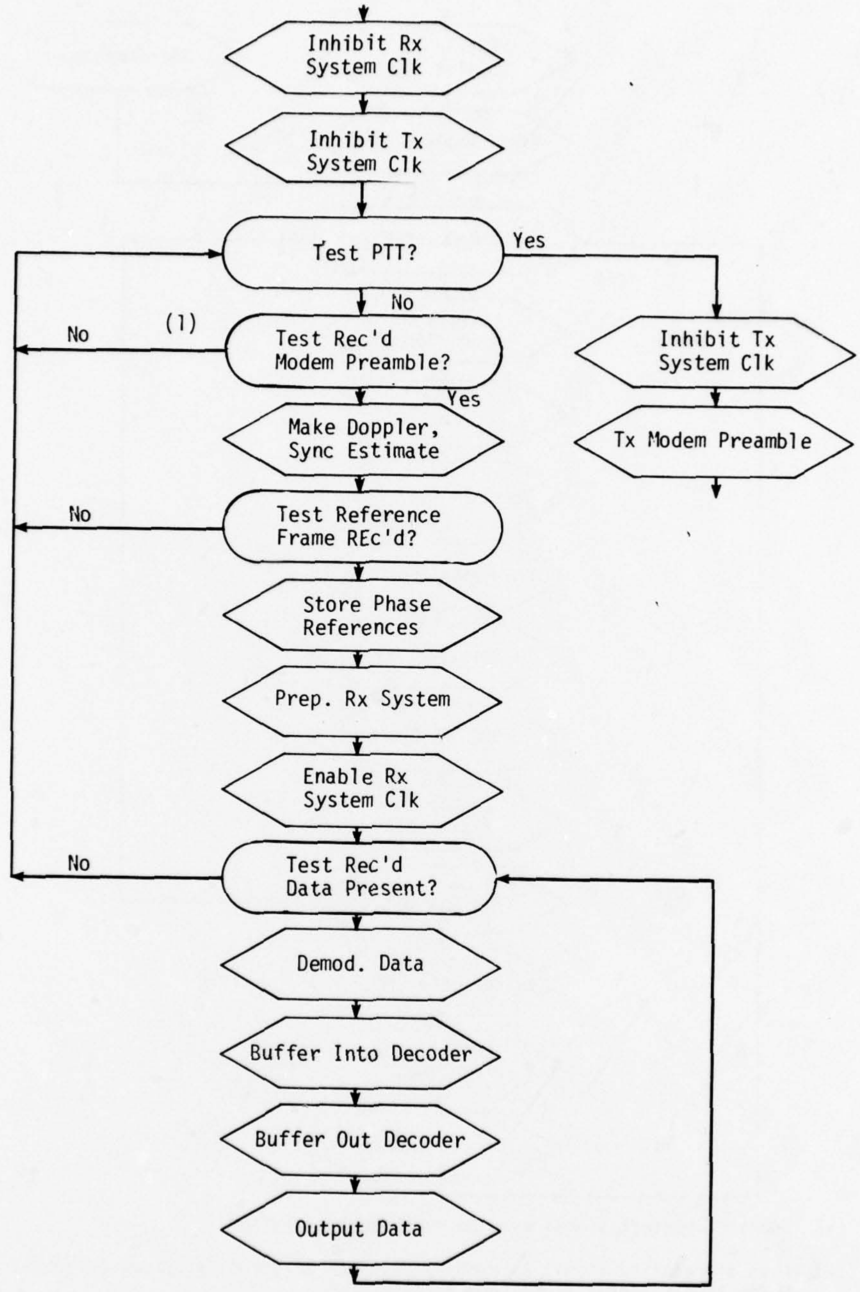
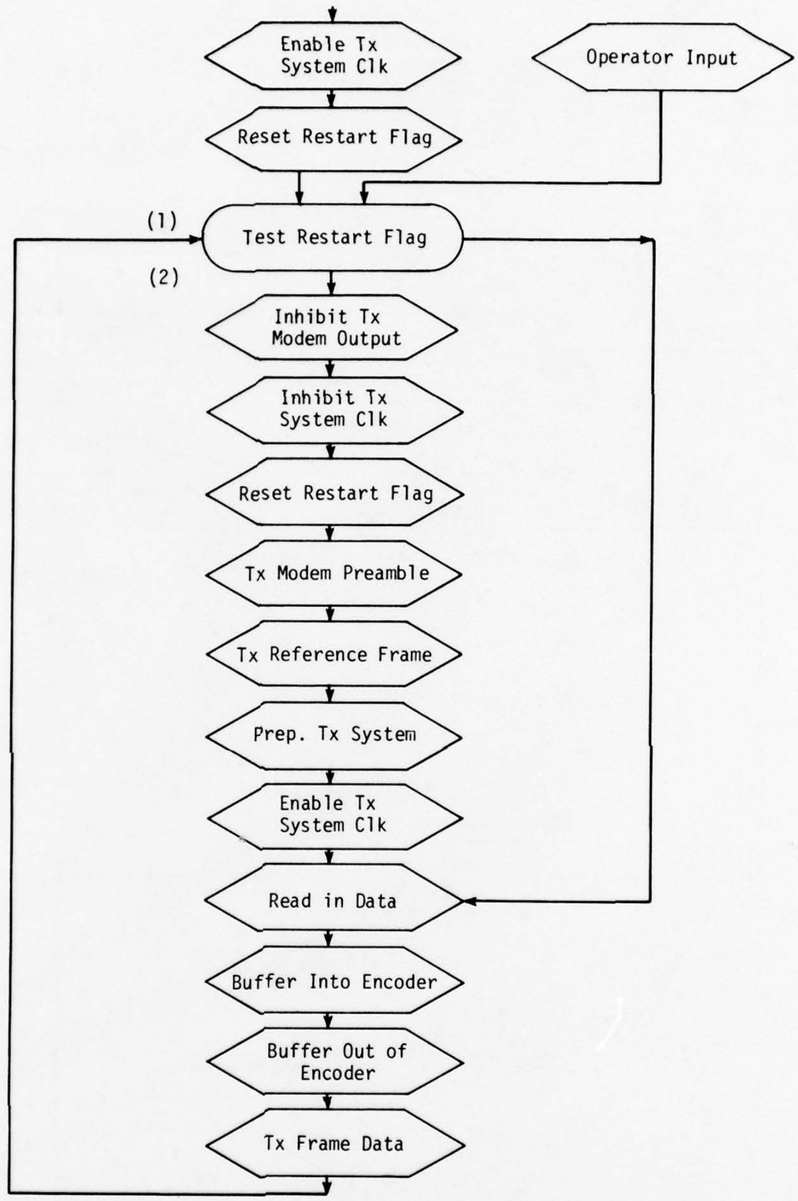


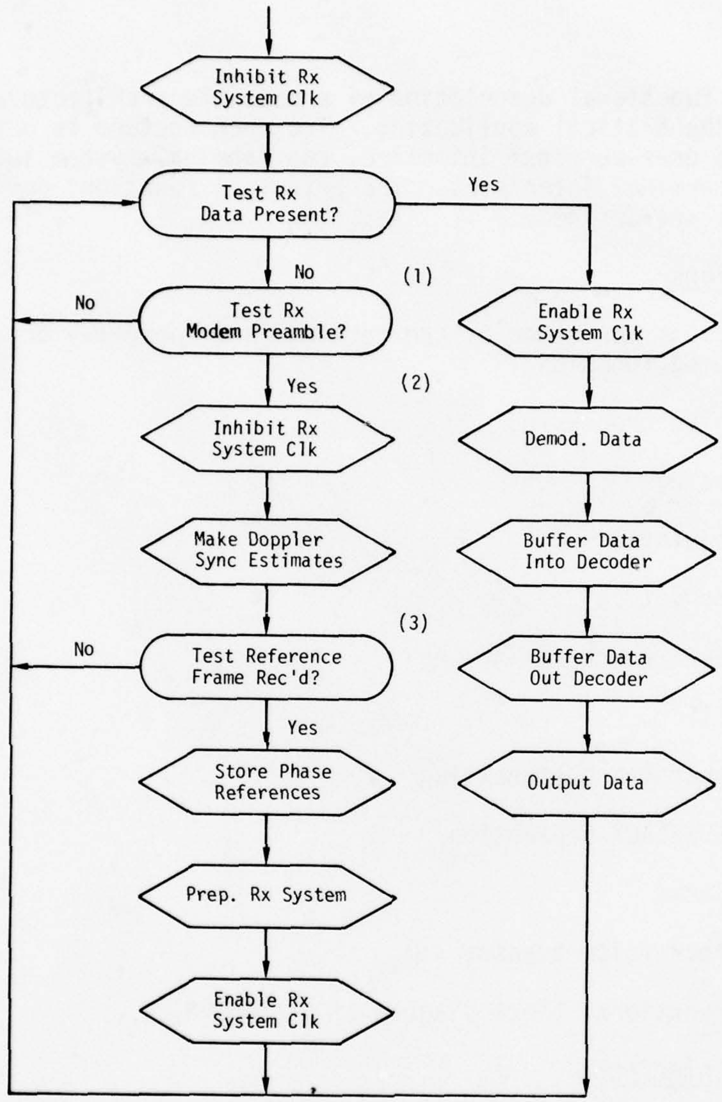
Fig. 6 - Preamble/data functional diagram half duplex, data mode, receive



(1) Operator restart signal used to resynchronize system.

(2) Loss of received signal is used by receiver to identify system preamble. Modem treats preamble the same as data.

Fig. 7 - Preamble/data functional diagram full duplex, data mode, transmit



(1) Loss of received signal used to indicate restart. Test for modem preamble occurs only after loss of signal

(2) Temporary loss of received signal without modem preamble being detected does not cause receive system clock to be disabled.

(3) Reference frame must be detected on first frame in which preamble signal is lost.

Fig. 8 - Preamble/data functional diagram full duplex, data mode, receive

APPENDIX B

CANDIDATE TERMINAL FUNCTIONAL DESCRIPTION

Scope

This is a functional description of a possible architecture for the ANDVT for the tactical application. The architecture is described in terms of the user-terminal interface, the terminal-system interfaces and the intra-terminal interfaces. The principal functions performed by the terminal are defined.

Terminal Functions

The operations performed by the terminal are logically organized into the following functions:

- voice processing
- coding
- modulation
- encryption
- performance monitoring
- BITE
- control and signalling
- RED/BLACK separation
- alarms
- clear voice bypass.

Figure 1 is a functional block diagram of the ANDVT.

User-Terminal Interface

Two persons have normal accesss to the terminal: the user and the operator. These may be the same person in some installations. The handset or remote telephone is the user's principal interface to the terminal. In addition, the user has a minimum of three signals with which he can control the operation of the terminal. They are:

- off-hook
- push-to-talk (PTT)
- clear voice bypass.

The off-hook signal is used in the tactical terminal to control and conserve dc power to transmit functions. The PTT signal is used to transfer from the receive mode to the transmit mode. Both of these control signals function only when the terminal is operating in the half-duplex mode. Activation of the clear voice bypass permits the transmission of unencrypted digital speech.

Some remote telephone terminals have a selective calling capability. Up to eight distinct calls may be placed. The audio lines between the handset and the terminal are used for RED analog voice and for the transmission of performance monitoring audio signals from the terminal to the user. The normal sidetone signal to the user is undelayed RED analog voice.

The operator's interface to the terminal is used to activate the equipment, provide mode control and to control the encryption variables. In some installations these operations are performed by the operator at the terminal. In other installations, a higher order control system resides between the operator and the terminal (i.e., the Navy's ACCS or CSS).

Terminal-System Interface

In the tactical application there are four interfaces between the terminal and the communication system. They are classed as:

- BLACK analog
- BLACK digital
- RED digital
- RED analog.

The BLACK analog interfaces provide connections between the modems in the terminal and the RF equipment. The terminal also provides a keying signal to the RF transmitter.

A BLACK digital interface is provided principally for connection to communication systems which have their own modems. In addition, a BLACK digital interface is required to provide for a digital tandem between a DCA ANDVT and a tactical terminal. For this condition, the terminal must accept digital transmit clock signals from the receive terminal with which it is tandemed.

A RED digital interface satisfies two requirements. It is the normal interface to the terminal in the data mode. In addition, it provides for using the terminal with a RED digital conference bridge in the voice mode.

The final system interface is RED analog. It provides for a tandem capability at the voice level. It also provides for a RED analog conference bridge.

In all conference configurations some of the terminal control functions may need to be remoted to the conference control unit (i.e., the selective calling function).

Intra-Terminal Interface

In the digital voice mode, the ANDVT functions of voice processing, COMSEC, coding and modulation are all performed frame synchronously. The coding for the HF channel provides error detection and correction on that portion of the encrypted data most sensitive to transmission errors. All digital voice data bits are protected with coding for the VHF/UHF radio channel. Coding is applied only to the system preamble when the terminal is interfaced to an external modem.

The voice processing function is not performed in the data mode. The data input is RED digital, with the terminal providing the clocking signals. There are no frame signals associated with the input data. Coding is provided to cover all data bits equally. Modem and coding functions differ in the data and digital voice modes, and depend upon the information rate of the source data.

The sequence of events performed in the establishment of a communication link depend upon the mode of operation. In the half-duplex digital voice mode, PTT is used to transfer from the receive mode to the transmit mode. This transfer is blocked if the terminal is receiving a valid signal. The acknowledgement of a PTT signal causes the terminal to initiate a keying signal to the transmitter. A "prep" signal is passed to the transmit COMSEC module which generates the system preamble for that call. The preamble is encoded to protect it from transmission errors. After a time sufficient for the RF transmitter to be ready, the modem preamble is transmitted. If the transmission is via HF, a modem phase reference frame must be transmitted. This is then followed by the encoded system preamble. The terminal must inhibit the transmit clock to the COMSEC module for a brief period after reading in the system preamble. This permits the preamble to be encoded and transmitted with sufficient redundancy to minimize synchronization failure. Figure 2 is a diagram of the sequence of transmissions for the HF-ANDVT.

In the half-duplex data mode, PTT is an operator function and not a user function. The system preamble is encoded in the same manner as the digital data in this mode of operation.

In the full-duplex digital voice mode, PTT is not a control signal. Resynchronization of the communication link is performed by the operator at the transmit terminal and is referred to as a restart. The initiation of a restart causes the transmit modem output to be momentarily inhibited, the COMSEC transmit module is "prepped" and a new system preamble generated. The HF communication link is reestablished by the transmission of a modem preamble, a modem phase reference frame and the encoded system preamble. The loss of the HF modem signal is used by the receiver terminal to identify the system preamble. This protocol is performed because the terminal encodes and transmits the system preamble differently than it does the digital voice signal. If a receive terminal loses the HF modem signal temporarily and a valid modem preamble is not detected, then the return of the modem signal will permit the terminal to continue in sync because the clock to the receive COMSEC module was not inhibited.

In the full duplex data mode the basic control is the same as in the full duplex digital voice except that the system preamble is encoded in the same manner as the digital data.

In all HF modes of operation, the receiver terminal requires the receipt of a valid modem preamble followed by the phase reference frame in order to "prep" the receive COMSEC module. In addition, in the half-duplex mode, these signals must be received before the demodulator will output digital data to the decoder. In the full duplex mode, this is not a requirement.

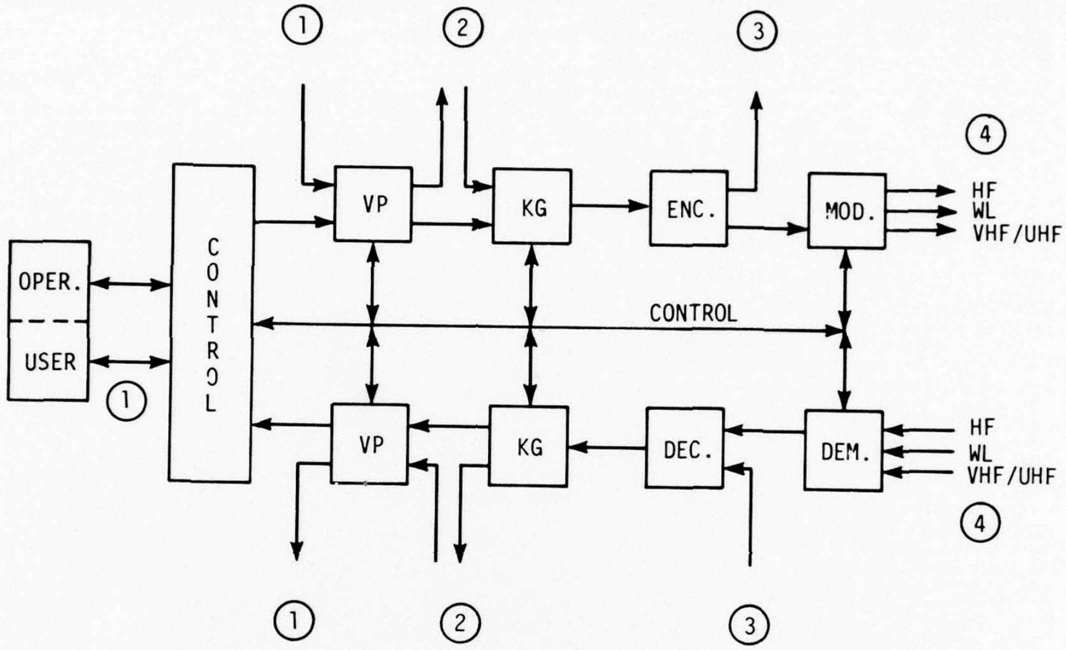
The clear voice bypass is achieved by a control signal from the user to the COMSEC module. This control signal causes the RED digital signal to be passed through the module unencrypted. In addition, it causes a change in the system preamble. This permits the receiving terminal to determine whether the following transmission is encrypted. Coding on the system preamble protects this aspect of the preamble function as well as its normal function.

In the digital voice mode, the terminal provides user related performance monitoring functions. These functions permit the user to obtain information about the performance of the terminal and the condition of the transmission channel. In both the half-duplex and full-duplex modes, the radio receiver audio output is routed to the user when the modem does not have a valid signal present. In the half-duplex HF mode, the initiation of a PTT causes the modulator output to be passed back to the user during the period that the modem preamble and the system preamble are being transmitted. This permits the user to make some assumptions about the performance of the transmit terminal, and to provide an indicator as to when voice transmissions may begin. The voice synthesizer audio output is suppressed by the terminal whenever the receive COMSEC module is in a resynchronization mode, or whenever the demodulator declares that a valid signal is not present, or whenever the RED digital data to the voice synthesizer is not proper. Coded audio alarms may be passed to the user indicating the nature of the problem. The normal sidetone signal to the user is undelayed RED analog voice.

There are two requirements for a BLACK digital interface to the terminal. They each have a different effect on the architecture of the terminal. When the ANDVT is used with a communication system with its own modem (i.e., satellite), then voice processing and COMSEC are the principal functions performed by the terminal. The coding function is applied only to the system preamble. In this mode the ANDVT will provide transmit clock signals to the external modem and will accept receive clock signals from the modem. There are no frame signals from or to the external modem. Consequently, in the transmit mode, the terminal generates framing signals for only the voice processor and the COMSEC module. In order for the receive ANDVT to derive frame sync from the BLACK digital data, the terminal will transmit a system preamble from which a framing signal can be derived. The receive ANDVT must identify this framing signal when operating in this mode. The sequence of transmissions is shown in Figure 3. When the ANDVT is interfaced to the SHF satellite modem, it is possible that the modem sync signal could be used to indicate the initial frame boundary and to "prep" the receive COMSEC module. A BLACK digital interface to the SHF satellite modem is the requirement for a proposed voice conferencing technique.

The second requirement for a BLACK digital interface to the ANDVT is derived from the need to tandem a DCA ANDVT and a tactical ANDVT. A single thread diagram of this requirement is shown in Figure 5. The transmitting functions of the tactical ANDVT, at the tandem point, are restricted to the coding and HF modulator. Frame information for these functions must be derived from the BLACK digital data in a manner similar to that previously described. Clock signals for the HF modulator must be accepted from the DCA terminal. The receive functions of the tactical ANDVT, at the tandem point, are restricted to principally demodulation of the HF signal, decoding and the generation of the framing signal. If this tandem operation is between two full-duplex terminals, restart is initiated by a drop of carrier by either user. If both users are operating half-duplex terminals, then PTT operation is used by both parties. The DCA terminal must operate in the half-duplex net mode in a manner compatible with net operation on HF. The length of the system preamble transmitted by the DCA user must be extended to permit time for the HF portion of the link to transmit the modem preamble and phase reference frame. At the wireline-to-HF interface, an equivalent PTT function will be derived from the carrier presence signal in the receive wireline modem. This will be used to transfer the HF system to the transmit mode. The HF modem framing will be obtained from the framing data portion of the system preamble. This information must be derived by the terminal prior to the transmission of the HF modem preamble. It will be necessary to provide some buffering of the digital data at the wireline-to-HF interface to insure that the first frame of the system preamble transmitted on the HF portion of the link is compatible with the normal preamble transmitted on HF nets where framing information is derived solely from the modem framing and not from the digital data. In the HF-to-wireline interface, the detection of the modem preamble by the HF modem will be used to generate PTT for the wireline portion of the link. The ANDVT must generate the framing signal to transmit over the wireline, to be followed by the preamble received by the HF modem.

Operation of the tactical terminal with VHF/UHF LOS radio equipment is restricted to half-duplex operation. The sequence of transmissions for the synchronization of the link is shown in Figure 4. A framing signal is transmitted, after an initial modem preamble, because this is not a frame synchronous transmission.



(1) RED ANALOG, VOICE

(2) RED DIGITAL, VOICE OR DATA

(3) BLACK DIGITAL, VOICE OR DATA

(4) BLACK ANALOG, VOICE OR DATA

Fig. 1 - Functional diagram of ANDVT

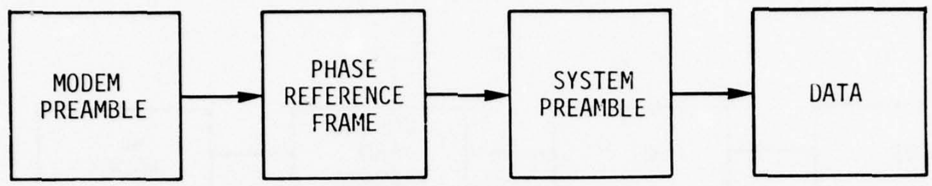


Fig. 2 Sequence of transmissions for HF

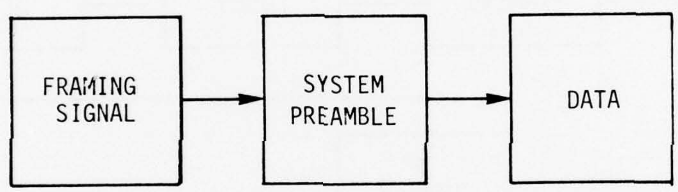


Fig. 3 - Sequence of transmissions for external modem

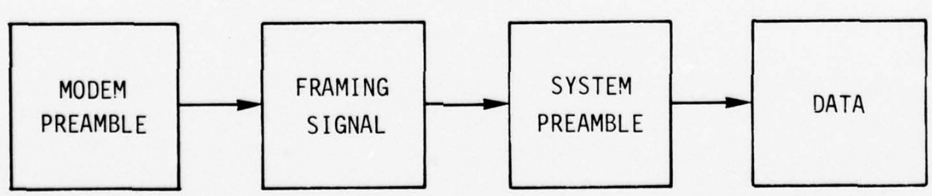


Fig. 4 - Sequence of transmission for VHF/UHF

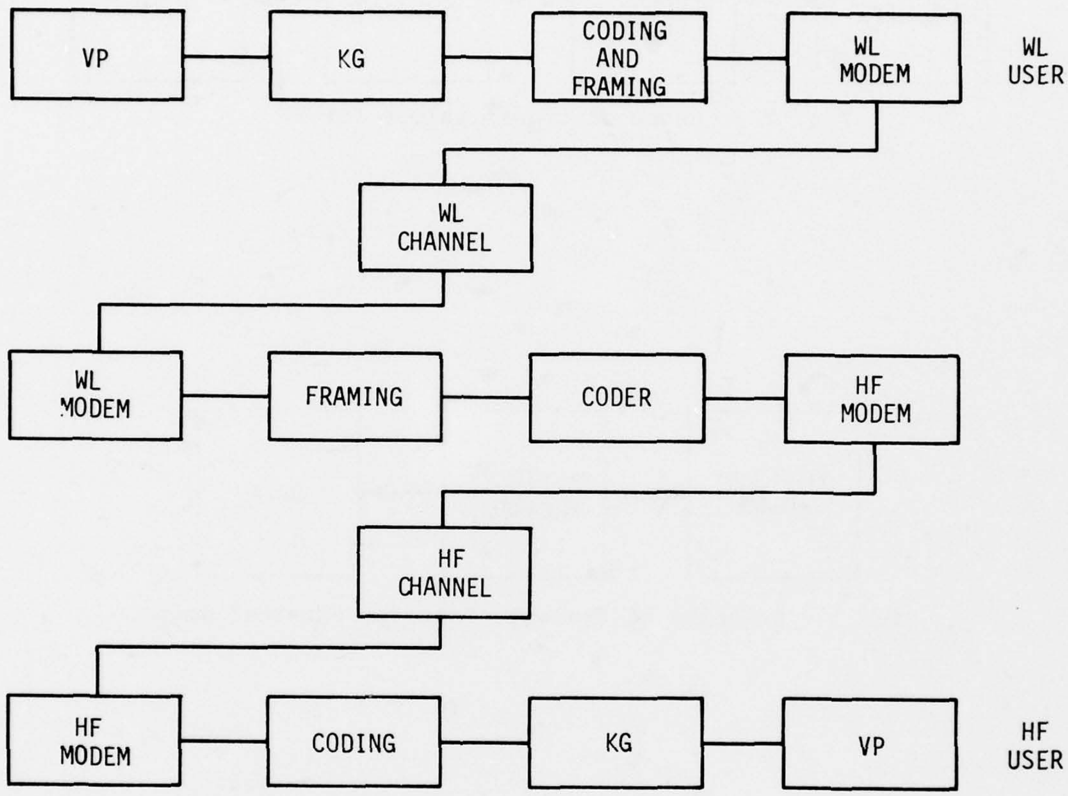


Fig. 5 - Tandem operation of WL and HF links