

RECOMMENDATION ITU-R F.763-4*

**DATA TRANSMISSION OVER HF CIRCUITS USING PHASE SHIFT KEYING
OR QUADRATURE AMPLITUDE MODULATION**

(Question ITU-R 145/9)

(1992-1994-1995-1997-1999)

The ITU Radiocommunication Assembly,

considering

- a) that there is an increasing demand for high-rate data transmission;
- b) that to meet this need, two types of phase-shift keying (PSK) modems may be used, namely parallel transmission modems using multi-channel voice frequency telegraphy and serial transmission modems using a single sub-carrier;
- c) that to compensate for the unfavourable nature of the transmission medium, the following techniques are available for the two types of modems:
 - various forms of dual diversity operation including separate single sideband (SSB) emissions or a single independent sideband (ISB) emission;
 - error detection and error correction coding combined with time interleaving;
 - variable data rate to adapt the system to the channel capacity;

and, for parallel transmission modems only:

- several levels of in-band frequency diversity;
- introduction of guard times between frames to combat multipath propagation and group-delay distortion,

recommends

- 1** that for data transmission at binary data rates up to 2 400 bit/s using frequency-division multiplex (FDM) and PSK systems, the system described in Annex 1 is preferred;
- 2** that for data transmission at binary data rates up to 3 600 bit/s using serial transmission modems, the system described in Annex 2 is preferred;
- 3** that reference should be made to Annex 3 for additional information concerning generalized PSK;
- 4** Annex 4 describes mode/polarization diversity systems to improve the performance of HF PSK systems;
- 5** that for data transmission at binary rates up to 4 800 bit/s using serial transmission modems, the system is described in Annex 5.

* This Recommendation should be brought to the attention of Radiocommunication Study Group 8.

Data transmission at 2 400/1 200/600/300/150/75 bit/s over HF circuits using multi-channel voice-frequency telegraphy and PSK

1 System description

1.1 A receiving/transmitting terminal of the system consists of:

- a sender and receiver of digital information (e.g. computer);
- a modem, the primary function of which is the conversion of information from digital to analogue form compatible with the input to a radio transmitter and conversion of the analogue information at a radio receiver output into digital data compatible with the digital receiver input.

This modem also performs various coding functions and effects diversity combination;

- RF receiving and transmitting equipment connected to antennas.

1.2 At the transmit side, the 2 400 bit/s incoming data stream is fed to a serial-to-parallel converter. At 32-bit intervals (i.e. 13.33 ms intervals) the content of this converter is transferred in parallel to a 32-bit memory device, the output of which is connected to a QPSK modulator.

The modem generates in transmission a composite audio signal consisting of a set of 18 tones in the band 300 to 3 000 Hz.

Of these tones, 16 have a spacing of 110 Hz (935 to 2 585 Hz) and are modulated in differentially encoded quaternary phase shift keying (DE-QPSK) mode, each at 75 Bd, thus permitting a data rate of $16 \times 75 \times 2 = 2\,400$ bit/s.

The tone at 605 Hz is used for the correction of end-to-end frequency errors, including any Doppler effect. The tone at 2 915 Hz (or 825 Hz) is used for system synchronization.

The dual diversity combiner can accept inputs either from two receivers operating in space, frequency or polarization diversity mode or from one receiver operating in ISB mode.

When the data rate is a sub-multiple of the transmission speed, various in-band diversity arrangements can be implemented. As an example, a data rate of 1 200 bit/s provides a dual diversity ($1\,200 \times 2$), a data rate of 600 bit/s, a quadruple diversity (600×4) and so forth, all with a transmission speed of 2 400 bit/s. Utilization of the maximum possible diversity, both in-band and between independent channels, can thus be made according to the data rate selected. Provision is made for 75/150/300/600/1 200 bit/s.

In addition to a choice of coded/uncoded operation, with selectable data rate and diversity mode, this modem also allows setting of the interleaving interval thus providing a flexible communication system as summarized in Table 1.

The transmission signal consists of frames whose duration is 13.33 ms. This includes a time guard (4.2 ms) which is introduced to offset the effects of multipath propagation.

The modem uses two techniques to reduce signal impairments, particularly those caused by impulsive noise and flat fading:

- error correction code;
- time interleaving.

A form of BCH cyclic block code (16,8) is used. The BCH codewords are stored in a memory to be extracted during the interleaving process. Interleaving is obtained by considering:

- the first bit of the last stored word;
- the second bit of the "(m) word stored before";
- the third bit of the "(2 m) word stored before" ...;
- the 16th bit of the "(15 m) word stored before".

TABLE 1

Data rates/modes (independently selectable for transmission and reception)

Data rate (bit/s)	Uncoded modes			Coded modes			
	Diversity modes			Time interleaving Available time spread (transmitter and receiver) (s)	Additional diversity modes		
	In-band	Channel	Total		In-band	Channel	Total
2 400 1 200	– ×2	×2 ×2	×2 ×4	0-12.8	– –	×2 ×2	×2 ×2
600 300	×4 ×8	×2 ×2	×8 ×16	0-25.6 0-51.2	×2 ×4	×2 ×2	×4 ×8
150 75	×16	×2	×32	0-102.5 0-205	×8 ×16	×2 ×2	×16 ×32

The interleaving level (m codewords) can be chosen according to the propagation conditions of the radio path from 0 (no interleaving), 1, 2, 4, 8, 16, 32, or 64, corresponding to a data reception delay ranging from a few milliseconds to tens of seconds. As the wrong bits do not belong to the same coded word, a better protection against burst errors is achieved.

In Fig. 1, the performance of the modem with Gaussian distributed noise is given in terms of bit error probability, P_e , as a function of signal-to-noise ratio, S/N , for both with coding and without coding modes, in a 250-3 000 Hz bandwidth.

The effects of coding become prominent at the higher values of S/N .

The curves were obtained with an experimental test set-up in which the modem was fed with a test pattern to produce the audio frequency tones. The output of the modem was summed with Gaussian noise, filtered and applied to the receiving input of another modem from which the test pattern was retrieved at the output. The test pattern was then fed to a data error analyzer to enable the bit-error ratio (BER) to be determined.

Figure 2 indicates the results of a computer simulation of the modem performance in a fading channel.

A fading channel was simulated in which two equi-amplitude paths carry signals separated by a multipath delay of 1 ms and differing in frequency by 1 Hz, in order to obtain fades which ran through the passband rather than remaining at certain fixed frequencies.

From Fig. 2, it can be seen that the performance is improved by using a combination of the various types of diversity (in-band and out-of-band), error correcting codes and interleaving techniques for 600, 1 200 and 2 400 bit/s rates.

The modem is currently in experimental use as part of an HF link between two radio stations located in central and southern Italy, and separated by approximately 800 km (500 miles).

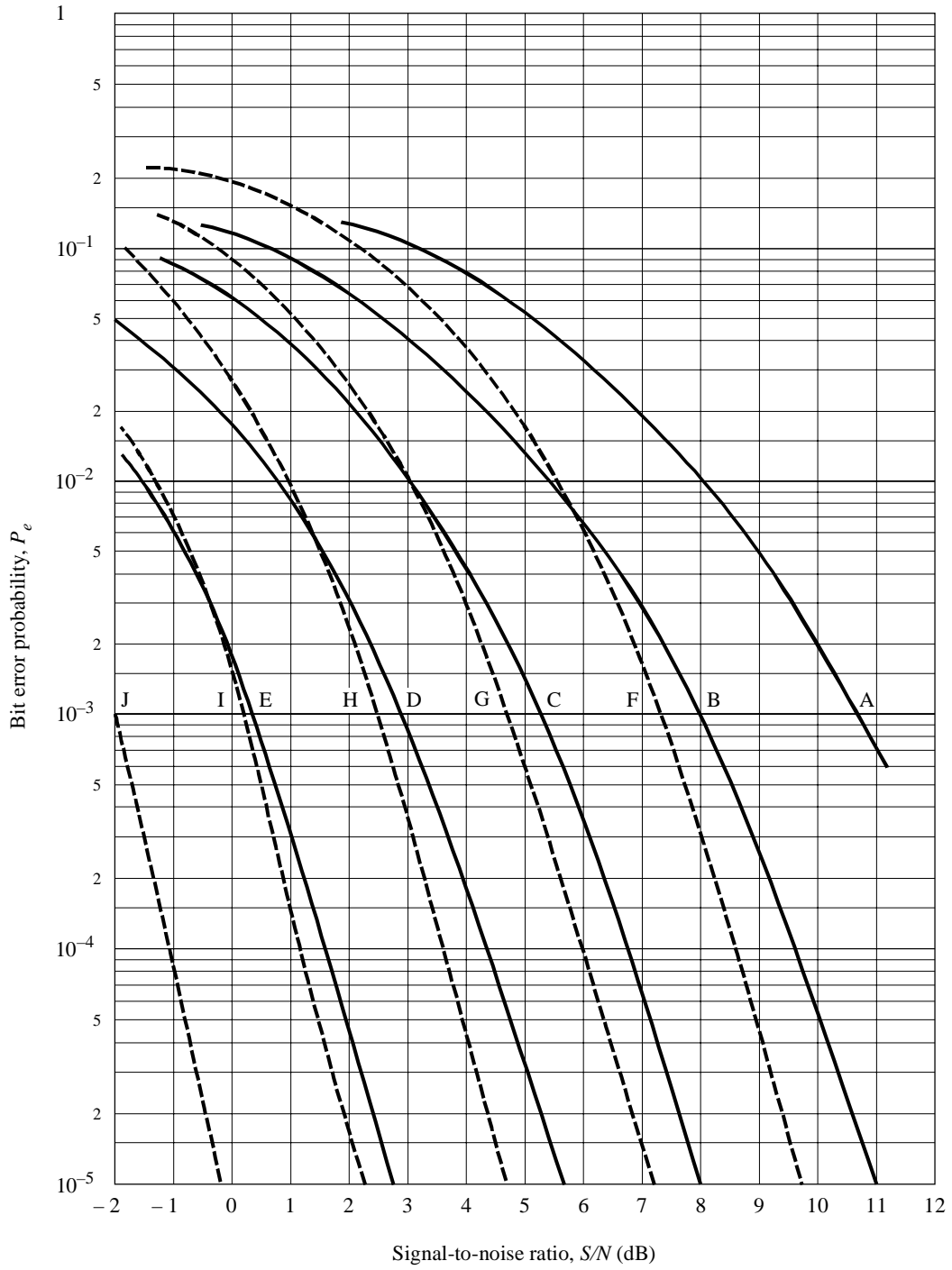
1.3 The RF equipment performs, in transmission, operations relative to channel modulation and produces an emission having suitable radio frequency and power characteristics. Reverse operations, relative to frequency conversion, are carried out in reception so as to obtain the composite audio signal to be conveyed to the modem.

The RF equipment has the following specific characteristics:

- phase jitter: less than 5° for 10 ms time interval (100 samples);
- group delay distortion: 500 μ s in transmission, 500 μ s in reception;
- intermodulation: 36 dB below peak envelope power.

FIGURE 1

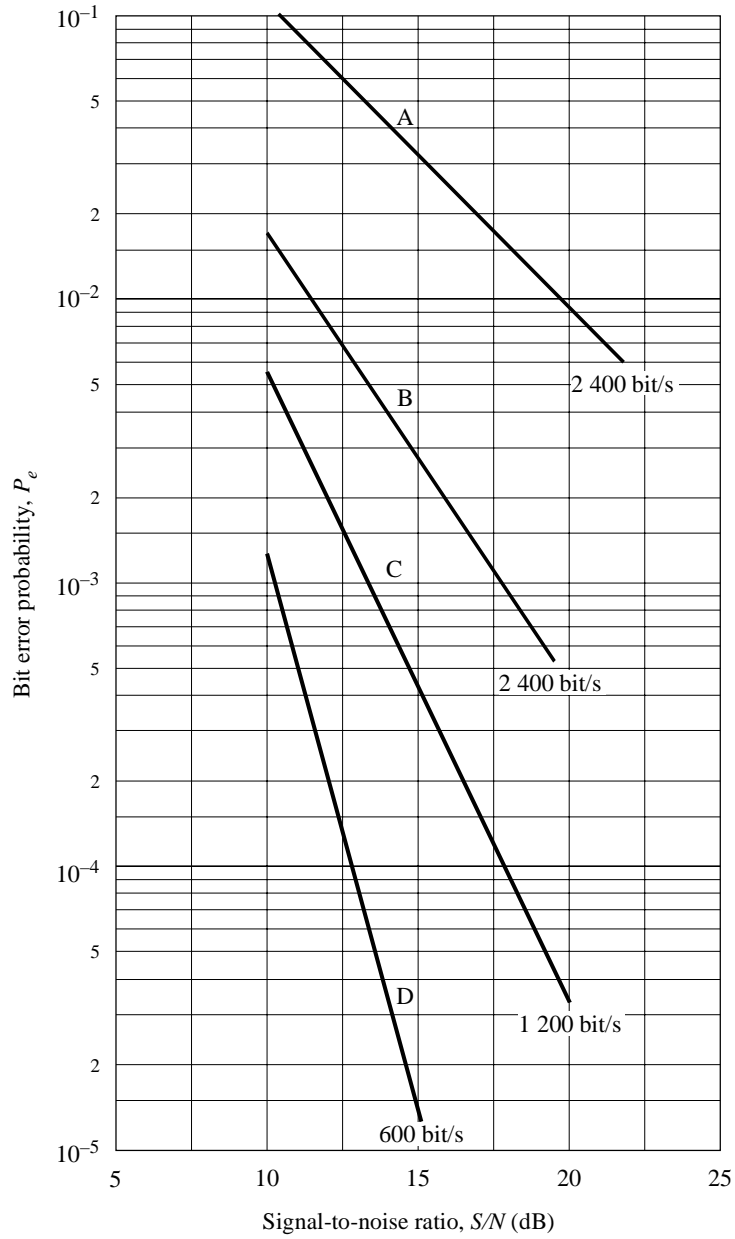
Bit error probability versus S/N for various data rates using with coding or without coding modes with in-band diversity for a non-fading channel with Gaussian noise



Without coding	—	{	A: 2 400 bit/s	{	With coding without interleaving	---	{	F: 1 200 bit/s
			B: 1 200 bit/s					G: 600 bit/s
			C: 600 bit/s					H: 300 bit/s
			D: 300 bit/s					I: 150 bit/s
			E: 150 bit/s					J: 75 bit/s

FIGURE 2

Bit error probability versus S/N for a selective fading channel using data rates of 600, 1 200 and 2 400 bit/s in the following cases



- A: without diversity
- B: out-of-band only
- C: in- and out-of-band diversity
- D: in- and out-of-band diversity with the use of error correcting codes and interleaving

Data transmission at rates up to 3 600 bit/s over HF circuits using a serial transmission modem

1 General

The modem permits data transmission in a 3 kHz HF channel. It receives and reconstitutes digital data at a rate of $\leq 3\,600$ bit/s and generates an analogue AF signal within the 300-3 300 Hz audio band.

It incorporates protection against multipaths, Doppler effect and fading.

2 Modem operating modes

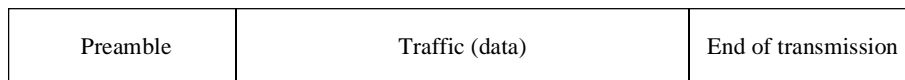
There are three possible operating modes.

2.1 Semi-duplex forward error correction (FEC) mode

2.1.1 This mode uses an MPSK ($M = 2, 4, 8$) modulation at 2 400 Bd, with a user bit rate of 75, 150, 300, 600, 1 200, 2 400 or 3 600 bit/s (not all of the bit rates are available with all of the waveforms), and with frames of 256 modulated symbols (of which 128 are user symbols), i.e. 106.6 ms.

2.1.2 A data exchange comprises three phases, namely preamble, traffic and end of transmission:

FIGURE 3
Description of a communication in FEC mode



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The preamble phase enables the called modem to detect the call and to receive the technical parameters (encoding, interleaving, data rate, modulation) that it needs for the rest of the transmission. The traffic phase contains the data to be transmitted. The end of transmission phase enables the called modem to detect an end of message word in order to terminate the link and return to traffic standby.

The end of transmission is effected when the calling modem transmits on-hook frames. These frames are similar to preamble frames, but include a bit containing the on-hook information.

2.1.3 The functions provided are as follows:

- *Emission:*
 - data encoding and interleaving;
 - framing and modulation;
 - transmission of AF signal.
- *Reception:*
 - reception of AF signal;
 - detection of synchronization;
 - demodulation of received signal;
 - data de-interleaving and decoding.

2.2 Full-duplex FEC mode

This mode amounts to the same thing as two independent FEC-type semi-duplex links. In each direction a preamble followed by data and an end of message word are sent and recognized by the called modem. As in the semi-duplex FEC mode, this preamble specifies the technical parameters that are to follow.

2.3 Automatic repeat request (ARQ) mode

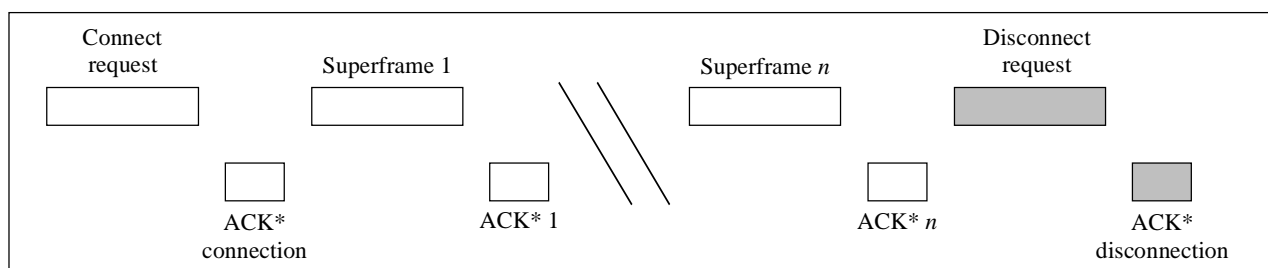
2.3.1 This mode uses an MPSK ($M = 2, 4, 8$) modulation at 2 400 Bd, with a user bit rate of 600, 1 200, 1 800 or 2 400 bit/s (not all of the bit rates are available with all of the waveforms), with frames of 256 modulated symbols (of which 128 are user symbols), i.e. 106.6 ms.

2.3.2 The ARQ mode is a data transmission mode involving selective repetition by block. The data for transmission are divided up into blocks corresponding to a modem frame. The calling modem sends a superframe of N blocks (N is nominally equal to 64, but may be lower than this during transmission of the last data) and waits for the called modem to acknowledge its receipt.

If any blocks have not been correctly received, they are re-transmitted in the following superframe, which is made up with new blocks.

The phases contained in this mode are call set-up (connection), data transmission and end of transmission (disconnection). In addition, the ARQ mode allows for momentary disconnection, caller/called party switching, flow control, and adaptive power, data rate and frequency control.

FIGURE 4
Description of a communication in ARQ mode



* ACK: acknowledgement

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The ARQ mode thus comprises two distinct phases, namely a transmission phase (transmission of a superframe at the calling end, and of an acknowledgement at the called end), and a reception phase (reception of an acknowledgement at the calling end, and of a superframe at the called end).

2.3.3 Adaptive control

2.3.3.1 The ARQ mode allows adaptive power, data rate and frequency control. Of these, only the adaptive data rate control is entirely managed by the modem. In the case of power control, the modem indicates to the system the adaptation to be effected and continues the transmission, while in the case of frequency control, the modem momentarily disconnects itself after indicating to the system the need to find a new frequency.

2.3.3.2 The adaptive power control procedure is based on statistical measurements of the link quality. Adaptive power increase is achieved very rapidly, while power decrease involves a large time constant.

2.3.3.3 Adaptive data rate control is effected on three of the data rates chosen from among the four that are available, namely 2 400, 1 800, 1 200 and 600 bit/s.

Adaptive increases in data rates are based on statistical measurements of the link quality, while decreases are based either on statistical measurement of the link quality, or on the non-reception of data or acknowledgements during the transmission.

2.3.3.4 If the adaptive data rate decrease control is not sufficient to continue the transmission, a request is made to the system to implement adaptive frequency control.

In order that a new frequency may be sought, the modem momentarily disconnects itself and stands by to resume the transmission, storing the data which have not yet been transmitted.

2.3.3.5 It is possible to set up the modem in ARQ mode in such a way that it does not implement adaptive data rate control. In this case, only the frequency and power control are effected.

2.3.4 The functions provided are as follows:

- *Send, at the calling end:*
 - data segmenting,
 - data encoding,
 - framing and modulation,
 - transmission of AF signal.
- *Send, at the called end:*
 - encoding of acknowledgements,
 - framing and modulation,
 - transmission of AF signal.
- *Receive, at the calling end:*
 - reception of AF signal,
 - detection of synchronization,
 - received signal demodulation,
 - decoding of acknowledgements.
- *Receive, at the called end:*
 - reception of AF signal,
 - detection of synchronization,
 - received signal demodulation,
 - decoding of data,
 - data reassembling.

3 Technical characteristics of the modem

3.1 Modulation

3.1.1 The modulation technique involves phase shift of a sub-carrier with a frequency of 1 800 Hz. The modulation rate is 2 400 Bd, with a minimum accuracy of 10^{-5} .

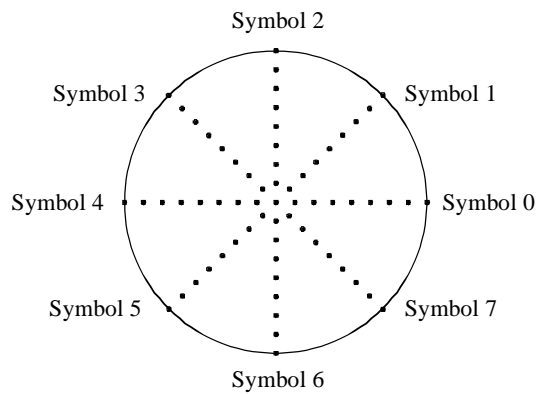
3.1.2 The clock stability associated with the generation of the 1 800 Hz is 10^{-5} .

3.1.3 The phase shift of the modulated signal in relation to the unmodulated reference sub-carrier may take on one of the following values:

Symbol No.	Phase
0	0
1	$\pi/4$
2	$\pi/2$
3	$3\pi/4$
4	π
5	$5\pi/4$
6	$3\pi/2$
7	$7\pi/4$

Symbol number n is associated with the complex number $\exp(jn\pi/4)$.

FIGURE 5
Encoding of phase states



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3.2 Transcoding

Transcoding is an operation in which a symbol to be transmitted is associated with a group of binary digits.

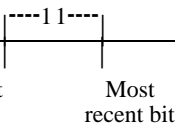
3.2.1 Data rate of 1 200 bit/s: 2-PSK

Transcoding is effected by associating a symbol with a binary digit according to the following rule:

Bit	Symbol
0	0
1	4

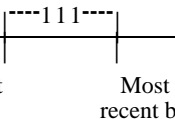
3.2.2 Data rate of 2 400 bit/s: 4-PSK

Transcoding is effected by associating a symbol with a set composed of two consecutive binary digits according to the following rule:

Dibit	Symbol
00	0
01	2
10	6
 -----11----- Oldest bit Most recent bit	4

3.2.3 Data rate of 3 600 bit/s: 8-PSK

Transcoding is effected by associating a symbol with a set composed of three consecutive binary digits according to the following rule:

Tribit	Symbol
000	1
001	0
010	2
011	3
100	6
101	7
110	5
 -----111----- Oldest bit Most recent bit	4

3.3 Frame structure

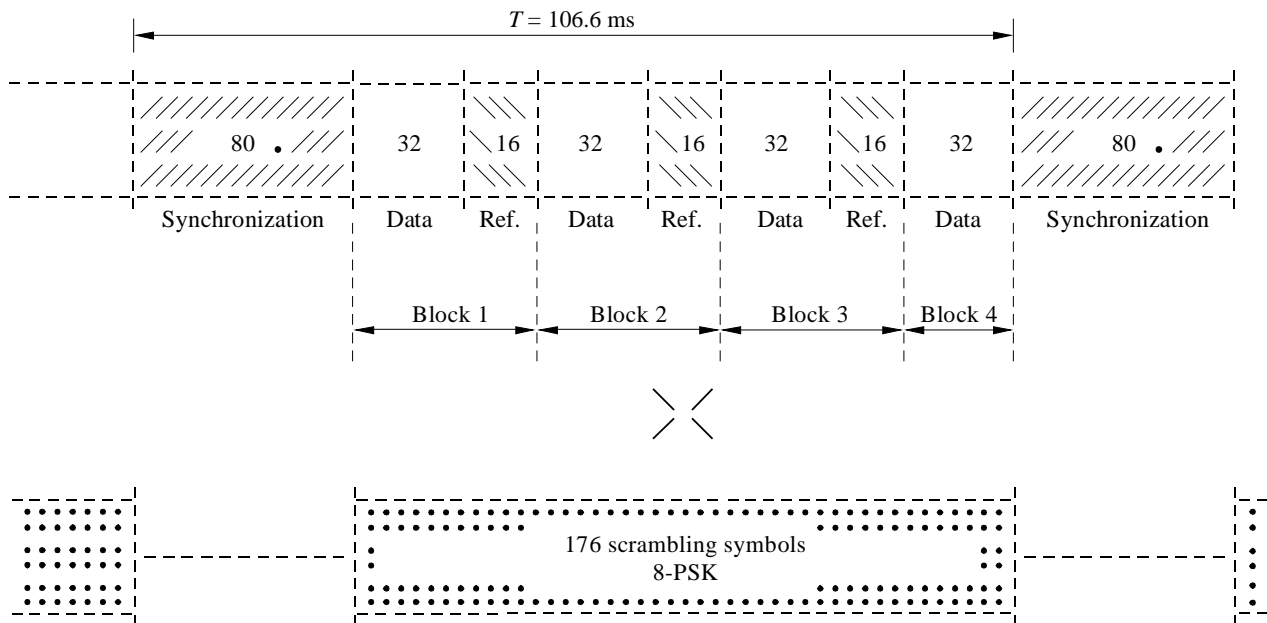
3.3.1 The symbols to be transmitted are structured in recurring frames of 106.6 ms in length. The number of binary digits transmitted per frame is 128 bits at 1 200 bit/s, 256 bits at 2 400 bit/s and 384 bits at 3 600 bit/s.

3.3.2 A frame is made up of 256 symbols, of which the breakdown is as follows: 80 symbols for synchronization, 48 reference symbols and 128 data symbols.

Figure 6 depicts the frame structure.

3.3.3 The synchronization sequence is transmitted in 2-PSK, at a modulation rate of 2 400 Bd. It is used by the modem for detecting the presence of the signal and for correcting the frequency shift resulting either from the Doppler effect or the difference between the transmit and receive carrier, bit synchronization and either the equalization time in the case of equalization by recursive filtering, or the HF channel evaluation in the case of detection by the maximum likelihood method.

FIGURE 6
Frame structure



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3.3.4 The reference and data symbols are structured in four blocks, the first three of which comprise 32 data symbols followed by 16 reference symbols, with the last block comprising 32 data symbols. All of the reference symbols correspond to the symbol number 0.

These 176 reference and data symbols are scrambled by a scrambling sequence comprising 176 symbols which is repeated every 106.6 ms. This sequence is transmitted in 8 state phase modulation at the rate of 2 400 Bd. It is thus possible to create a frame with 8 phase states, whatever the data rate (1 200 bit/s, 2 400 bit/s or 3 600 bit/s).

The scrambling operation consists in adding modulo 8, the symbol number associated with the data to the symbol number associated with the scrambling, which amounts to complex multiplication of the data symbol by the scrambling symbol.

3.4 Error correction coding, interleaving

The use of error correction coding in conjunction with adequate interleaving can considerably improve the BER.

On the basis of the three base modes without redundancy, namely

- 3 600 bit/s 8-PSK,
- 2 400 bit/s 4-PSK,
- 1 200 bit/s 2-PSK,

the coding permits the introduction of various redundancy possibilities.

3.4.1 FEC mode

This involves the use of convolutional coding in combination with interleaving which is also convolutional. The convolutional code used is redundant code 2 and constraint length $K = 7$, associated with the characteristics polynomial 171,133 (octal representation).

Redundancies lower than 2 are obtained by puncturing, while redundancies higher than 2 are obtained by repetition.

Among the various possibilities, we would mention the following:

Data rate with coding (bit/s)	Waveform	Redundancy	Method for obtaining this code rate
2 400	8-PSK	3/2	Conversion of data rate 1/2 to data rate 2/3
1 200	4-PSK	2	Unmodified code at data rate 1/2
600	2-PSK	2	Unmodified code at data rate 1/2
300	2-PSK	4	Code at data rate 1/2 repeated 2 times
150	2-PSK	8	Code at data rate 1/2 repeated 4 times
75	2-PSK	16	Code at data rate 1/2 repeated 8 times

3.4.2 ARQ mode

A Reed-Solomon (RS) coding is used, and there is no interleaving.

Data rate with coding (bit/s)	Waveform	Redundancy	Coding (symbols of 8 bits)
2 400	8-PSK	3/2	RS (48,32)
1 800	4-PSK	4/3	RS (32,24)
1 200	4-PSK	2	RS (32,16)
600	4-PSK	4	RS (32,8)

3.5 Spectrum of the modulated signal

The spectrum of the modulated signal after filtering and 1 800 Hz transposition is shown in Fig. 7. The total bandwidth is equal to 3 000 Hz.

3.6 Frequency error tolerance between transmission and reception HF carriers

The modem must be able to tolerate a shift of ± 75 Hz between the transmission and reception HF carriers (transmitter/receiver frequency error and Doppler shift included) and a frequency variation rate of at most 3.5 Hz/s.

4 Interfaces with other equipment

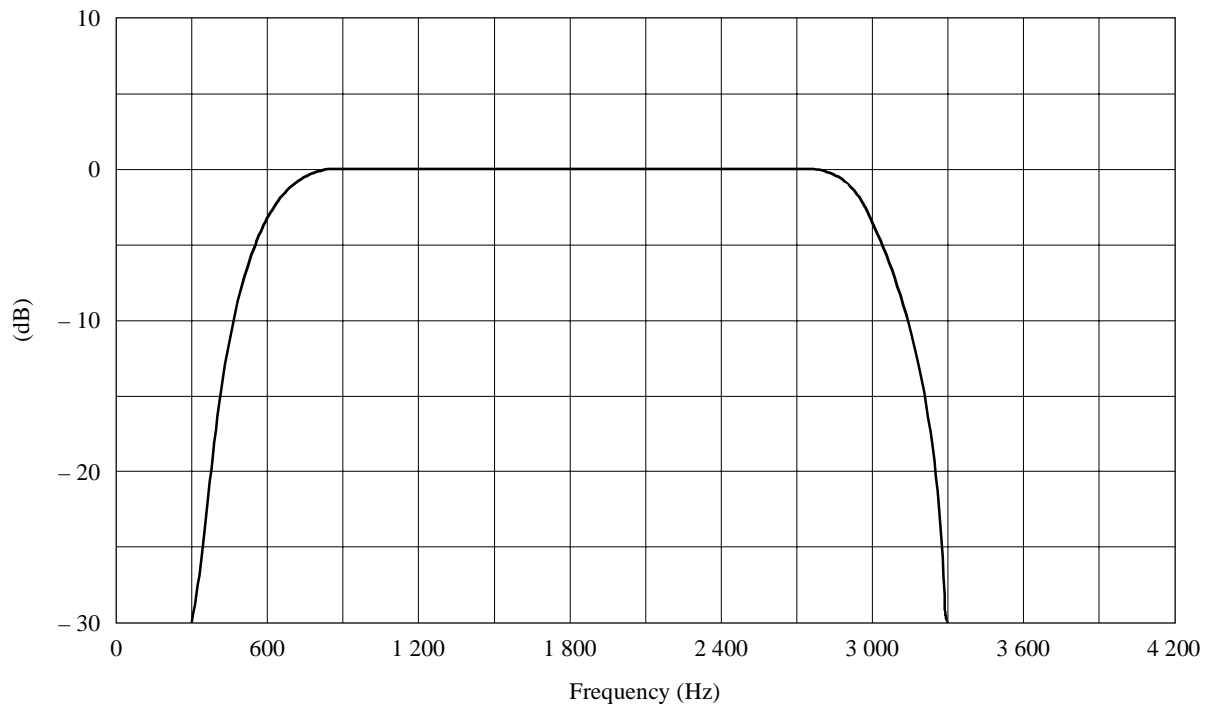
4.1 Modem interface with the data terminal

This satisfies ITU-T Recommendation V.24, the electrical characteristics of the interface being in conformity with ITU-T Recommendation V.11 (RS 422).

4.2 Modem interface with the transmitter and the receiver

The input and output circuits of the modem are of the balanced to earth type, having an impedance of 600 Ω at 0 dBm.

FIGURE 7
Spectrum of the modulated signal



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4.3 Quality of performance of associated transmitters and receivers

To obtain optimal performance, the following characteristics for transmitters and receivers are recommended:

4.3.1 They must have a passband such that between 300 Hz and 3 300 Hz variations of transmission loss are at most ± 2 dB.

NOTE 1 – The operation of a serial modem with a system bandwidth of 300 to 3 000 Hz is possible with reduced performances. Further study would be needed to design a serial modem with a sub-carrier of 1 650 Hz, operating with reduced bandwidth systems.

4.3.2 The group delay must not vary by more than 0.5 ms over 80% of this passband.

4.3.3 The accuracy of the transmitter and receiver pilot frequencies must be at least 10^{-6} .

4.3.4 The time constant of the automatic gain control (AGC) circuit must be less than 10 ms for de-sensitization and less than 25 ms for re-sensitization.

ANNEX 3

Transmission systems using PSK

1 Introduction

In HF channels information at bit rates of over 200 bit/s is normally transmitted using multistate methods and complex signals. This generally involves a combination of frequency-shifted orthogonal subcarriers with 2-PSK. With the latter technique a bit rate twice that obtainable with FSK can be achieved in the same frequency band and the redundancy can be used to increase noise immunity. Apart from multi-frequency PSK, practical interest attaches to a more general type of modulation – generalized PSK, in which the information to be transmitted is contained not in the differences between

the instantaneous phases of the sine-wave signals but in the difference between the phase spectra of complex orthogonal signals. The amplitude spectra of such signals coincide and may be matched with the channel frequency characteristic (or the interference spectrum) without violating the conditions of mutual orthogonality. On this basis it is possible to consider the construction of adaptive modems with a higher noise immunity or traffic capacity.

The practical application of generalized PSK has been held up in the past by the well-known difficulties involved in the synthesis and processing of complex signals. The basic problems have now been solved, thanks to the theory of synthesis which has been developed, and the availability of microelectronic modules with a high degree of integration, which has removed the obstacle of technical circuit complexity. This Annex sets forth the main principles governing the design of modems with generalized PSK, describes a variant which has been developed and gives a number of test results.

2 Theoretical questions

2.1 Selection of signals

As was pointed out by Shannon, in order to achieve a transmission rate equal to the communication capacity in channels with a frequency characteristic of $Y(\omega)$ and a Gaussian noise of $N(\omega)$, use has to be made of signals characterized by a steady state Gaussian process with a power of P and a power spectrum of:

$$F(\omega) = \begin{cases} B - \frac{Y(\omega)}{N(\omega)} & \text{for } \omega \in \Omega \\ 0 & \text{for } \omega \notin \Omega \end{cases} \quad (1)$$

where the integration range Ω is determined from the condition $F(\omega) \geq 0$ and the constant B depends on the power of the signals. Since in practice there are always standards governing the permissible limits for the delay in information being transmitted, the maximum signal duration and the number of signals have to be limited. Under these conditions finite-dimensional combinations of determined orthogonal signals, the squares of whose spectral density moduli coincide with $F(\omega)$, may be considered as being close to optimum. However from (1) it follows that $F(\omega) = 0$ at all frequencies where $B < Y(\omega)/N(\omega)$, i.e. mutual orthogonality has to be preserved when individual portions of the spectrum are rejected. The multi-frequency signals used in existing modems do not possess this property. Moreover, their orthogonal spectrum shape is optimum only for channels with a flat frequency characteristic and white noise type interference. Calculations show that failure to meet these conditions may result in information transmission rate losses of up to 40% of the channel's communication capacity.

Another criterion for assessing the optimum nature of orthogonal signal combinations is the requirement as regards the shape of their autocorrelation function. For example, to ensure stability in the operation of a synchronization system, the main lobe of this function has to be narrow enough and the side lobes must not exceed a given level. In this case mutual orthogonality must be assured at a given signal amplitude spectrum which does not necessarily satisfy condition (1).

In view of the above, in order to achieve generalized PSK, a special class of signals based on the use of complex systems of functions with double orthogonality was developed. Their spectral densities may be represented as follows:

$$S_k(\omega) = |S(\omega)| e^{j[K\psi(\omega) + \alpha(\omega)]} \quad (2)$$

where:

$$|S(\omega)|^2 = A \left| \frac{d\psi(\omega)}{d\omega} \right|$$

where:

A : constant factor

$\alpha(\omega)$: arbitrary bounded function.

For a given amplitude spectrum, it is therefore possible to define the phase spectrum of the signals and therefore their spectral density. Further synthesis involves finding samples of spectral signal densities with different serial numbers and transforming them using a Fast Fourier Transform (FFT) into time samples. Synthesis of the signals may be combined

with coding of the signals in the time domain using the Reed-Solomon code; for this purpose a number of zero samples has to be added beforehand to the spectral density samples and only then can the FFT be performed. It should be noted that this type of mixed coding (orthogonal in the frequency domain and Reed-Solomon code in the time domain) is most effective for HF channels.

2.2 Selection of a processing algorithm

In the case of multi-state methods for the transmission of information, it is best to process the signals to be received using the optimum algorithm for reception "as a whole". The simplest way of implementing such an algorithm is to use component demodulators; for this purpose the following conditions have to be met:

- the multi-state signals have to be component-type, i.e. they have to be made up of the sum of the elementary signals;
- each elementary signal must contain information about the corresponding element of a code word $b_{i,k}$;
- the interference affecting the elementary signals must be mutually independent.

In this case the decision rule is as follows:

$$\max \left[L_i = \sum_{k=1}^N e_{i,k} y_k \right] \quad (3)$$

where:

$e_{i,k}$: sign coefficient which takes the value: +1 when $b_{i,k} = 1$, and
 -1 when $b_{i,k} = 0$

$$y_k = \ell_n \frac{W(Z_{k/1})}{W(Z_{k/0})}$$

where:

Z_k : complex input signal (see Fig. 1)

$W(Z_{k/1})$: probability of Z_k being 1

$W(Z_{k/0})$: probability of Z_k being 0.

Optimality in this case is determined by the extent to which the signals used meet the conditions listed above. The first two involve the possibility of using a component demodulator. For these conditions to be met, it is sufficient for each spectral density sample (or its components) to contain information about the sign of the corresponding binary symbol. The condition stipulating the mutual independence of interference can be reduced to a condition stipulating the independence of the projections of the received signal vector onto the system on the basis of Fourier transform functions. This condition is fulfilled in the case of independent fades in individual frequency bands, invariance of the basic functions to time shifts and interference with a flat power spectrum. In practice it is impossible for the various requirements listed above to be met fully, so that the noise immunity of the component demodulator will be lower than the potential noise immunity although much higher than in the case of separate signal element reception.

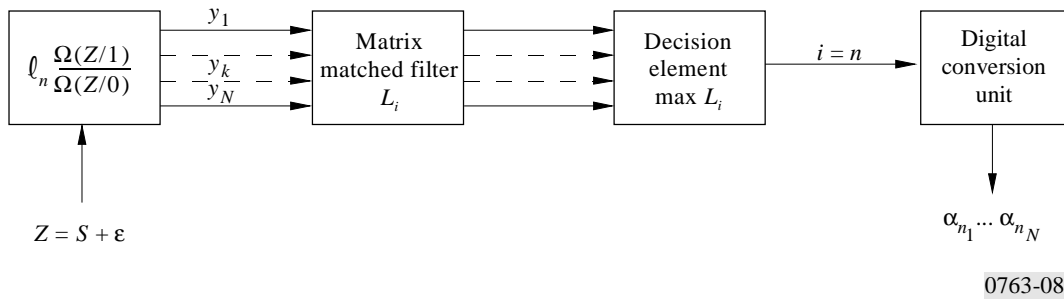
The block diagram of the receiving section of the modem which implements decision rule (3) consists of the following units (see Fig. 8): a unit for calculating the logarithm of the likelihood ratio y_k ; a unit for calculating the linear forms L_i ; a decision element determining the number of the linear form with the maximum value; a digital converter which compares with each number its own combination of binary symbols; and this provides an assessment of the information sequence transmitted.

3 Description of the system

A block diagram of the system is shown in Fig. 9. It consists of the following elements: the user terminals; the signal conversion unit (modem) located either in the immediate vicinity of the terminals or in a separate communication control unit; the receiving and transmitting SSB equipment and the corresponding antennas. When the modem is installed in the control unit, communication with the terminal is established via tone frequency channels.

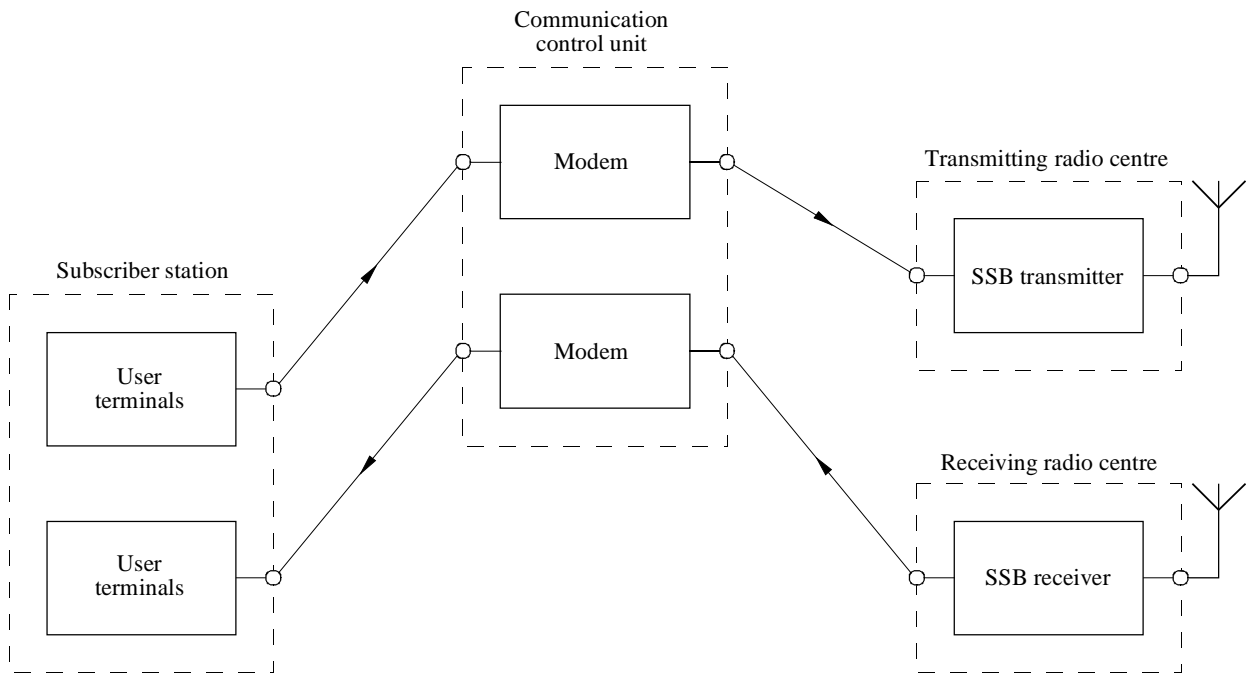
When it is set up in the immediate vicinity of the terminal, it can be connected up by d.c. circuits.

FIGURE 8
Receiving section of modem



0763-08

FIGURE 9
Block diagram of system



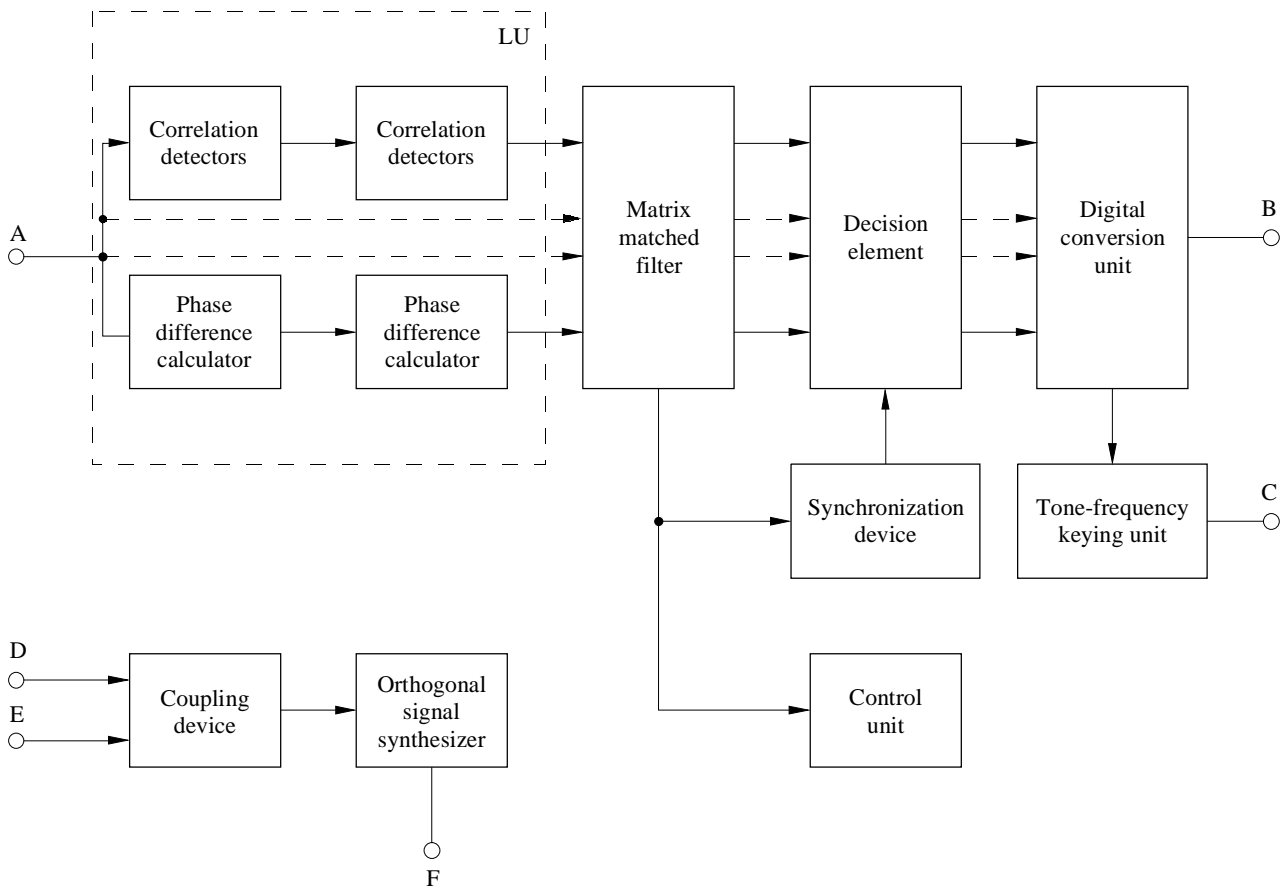
0763-09

A block diagram of the modem embodying the principles considered above is shown in Fig. 10. The modem is designed to transmit digital information at 600-1 200 bit/s. For lower bit rates, an additional codec has to be used. A bit rate of 2400 bit/s is obtained by increasing the number of signals used and switching over to separate signal element reception. The modem transmitter consists of a coupling device (CD) and an orthogonal signal synthesizer (OSS).

The CD is designed to match the modem with the user's terminal via the tone frequency channels or the d.c. circuits and to control the synthesizer. It includes a tone frequency amplifier-rectifier, a regenerator, and a logic control circuit.

The OSS shapes the analogue signals and amplifies them to the requisite level. It consists of a coding unit, a ROM, a digital to analogue converter (DAC), an low frequency filter and a power amplifier. A specific feature of the operation of the OSS is that time samples of all the signals to be used for the transmission of information are already entered in its ROM. These samples were calculated beforehand on a computer in accordance with the rules laid down in the previous section.

FIGURE 10
Block diagram of modem



- A: modem receiver input
- B: modem receiver tone frequency output
- C: d.c. modem receiver output
- D: tone frequency input of modem transmitter
- E: d.c. input of modem transmitter
- F: modem transmitter output

0763-10

In an initial stage, in order to verify the basic principles used, a set of 16 bi-orthogonal signals was synthesized; these had a flat amplitude spectrum in the range 1.1-2.42 kHz and an effective band of 0.66-2.86 kHz. Their spectral densities were represented by means of four complex samples, each of which could provide information on the signs of two binary symbols. To transpose the spectrum to these samples, two zero samples were added and after a Fourier transform an additional multiplication by a complex component was carried out.

The time samples of the signals calculated in this way were entered in an 8 bit grid in the ROM and, after they had been read at a timing frequency of 8.8 kHz, it was possible to obtain at the output of the DAC unit analogue signals with a duration of 3.33 ms and an orthogonality interval of 2.27 ms.

The sequence of operations in the modem transmitter is as follows: the binary information signals from the terminal are regenerated, combined to form 4-bit code words and are then fed to the input of the relative coding circuit which controls the selection from the ROM of one of the 16 forms of the signal. From the output of the ROM the samples are converted by the DAC unit into an analogue signal which after being amplified is fed along the tone frequency channel to the input of the SSB transmitter.

As shown in Fig. 8 the receiving section of the modem consists of the following elements: a unit which calculates the logarithms of the likelihood ratio (correlation detectors (CDT); a phase difference calculator (PDC)); a matrix matched filter (MF) which calculates all the linear forms L_i ; a decision element (DE) which determines the number of the maximum form; and a digital conversion unit (DCU). It also contains a synchronization device (SD) and a control unit (CU). Provision is made for a single or a dual operating mode with space or polarization diversity.

The transformation of the analogue signals into spectral density samples is carried out by the correlation detectors which calculate the in-phase and the quadrature component of each sample. The initial uncertainty relating to the phase of the channel is then eliminated using the PDC and the phase spectrum of the signal received is calculated. The matched filter is a matrix adder and each of its columns is adjusted for an appropriate sample selection using inverting amplifiers. The DE looks for the column with the maximum output voltage and, using the DCU, transmits the corresponding 4 element sequence of binary symbols which is fed to the input terminal either directly or via the tone-frequency keying unit (TKU).

The CU works on the principle that the voltages at the output buses of the matched filter coincide precisely to within a constant factor with the distribution of *a priori* probabilities. It is clear that the performance of the channel will be better the "steeper" this distribution is since in the ideal case the voltage must appear only on one of the output buses of the matched filter. The difference between the maximum voltage and the voltage whose level is closest to it at the other bus may be used to evaluate the quality of the channel in the information transmission process.

4 Experimental investigations

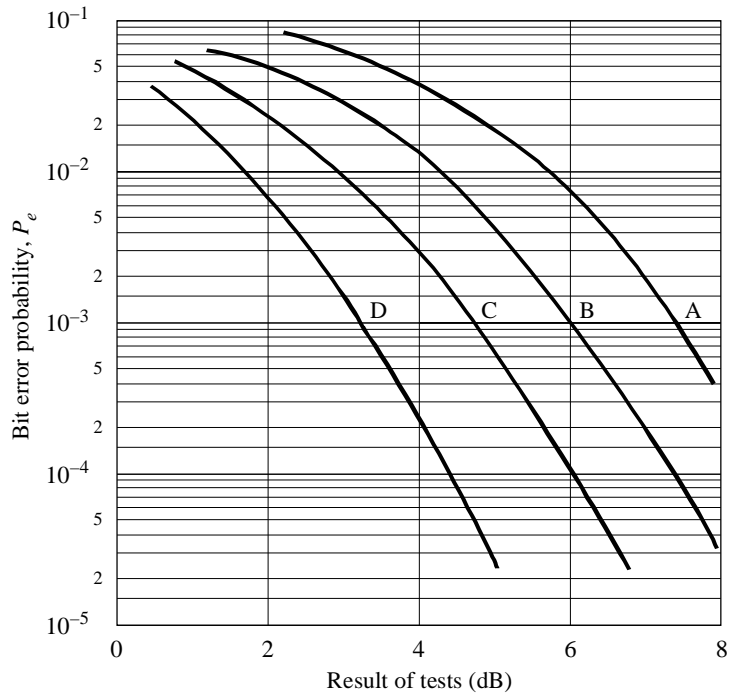
The laboratory tests on the modem were carried out using a modelling test bench which included the following elements: the SSB receiver; a two-ray channel simulator; a noise generator; and a digital counter to calculate the number of errors. A pseudo-random sequence (PRS) from a generator built into the modem was used as a test combination. Three modes of operation were analysed: a channel with constant parameters and white noise; a single-ray channel with Rayleigh fading; and a two-ray channel with a difference in the ray propagation time of 1 ms, identical ray amplitudes and Rayleigh fading. The results of the tests are shown in Figs. 11 and 12. By way of comparison, Fig. 11 shows curves for the noise immunity of a multi-frequency modem as described in Annex 1 for the same transmission speed. As the curves show, the modem studied has a better noise immunity. Comparison of curves A and B in Fig. 12a) show that the modem has a higher noise immunity in the two-ray channel than in the one-ray channel. The reasons for this is that in the case of flat fades, decision rule (3) is no longer optimum. In a two-ray channel a predominant role is played by frequency selective fades which the modem can combat more effectively. The dotted line in the figure shows the theoretical curve for the noise immunity of an optimum non-coherent separate signal element reception using binary PSK in the case of Rayleigh fading.

Link tests on the modem were carried out on 3 600 km and 4 300 km latitudinal paths. Use was made of a 15 kW SSB transmitter, rhombic transmitting antennas and fishbone receiving antennas (dual reception). Tests on the first path were carried out during the day and during the night on one frequency. On the second path two frequencies were used. The information bit rate was 1 200 bit/s. On the basis of 5 min measurements, integral curves were plotted showing the error rate distribution; these are shown in Fig. 12b).

5 Conclusions

The use of generalized PSK in combination with reception "as a whole" opens up additional possibilities for increasing noise immunity in the transmission of digital information. A modem developed to serve as a practical example of how the generalized PSK method may be implemented uses signals with a flat spectrum and from this point of view is similar to the modems described in Annex 1. Tests have shown that for links over 3 000-4 000 km it guarantees a bit rate of 1 200 bit/s with an error rate not exceeding 1×10^{-4} to 1×10^{-3} for 95 to 98% of the time.

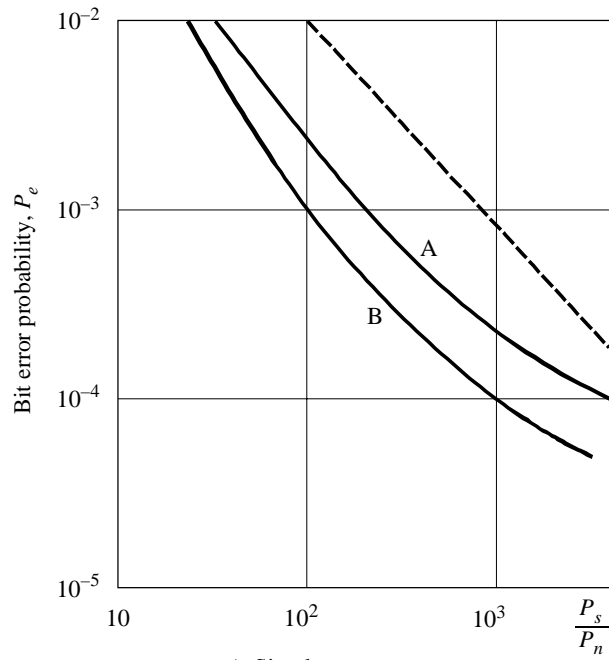
FIGURE 11
Noise immunity of modem



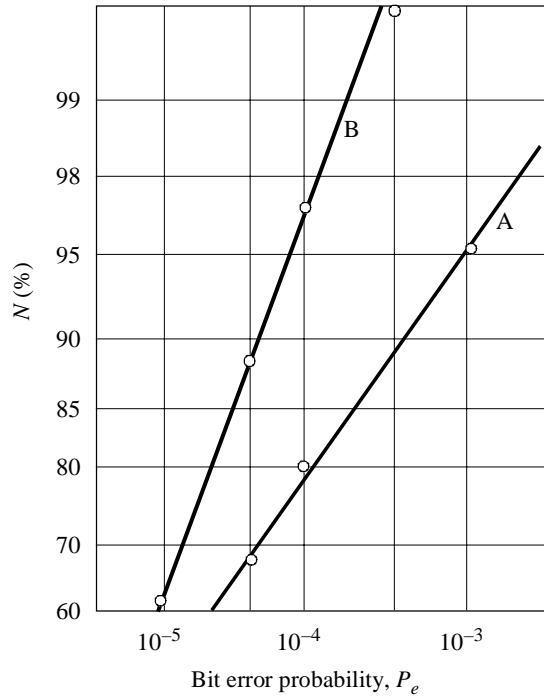
A: multi-frequency } 1 200 bit/s
 B: DPSK }
 C: multi-frequency } 600 bit/s
 D: DPSK }

0763-11

FIGURE 12
Noise immunity of modem in fading channels



A: 1 ray
B: 2 ray
Transmission speed: 1 200 bit/s



A: 3 600 km
B: 4 300 km
Transmission speed: 1 200 bit/s

ANNEX 4

Mode/polarization diversity in high-frequency radio data systems

1 Introduction

The amplitude of a received HF radio signal fluctuates when its direction of polarization changes with respect to the receiving antenna with minima occurring when the polarization is orthogonal and maxima when the polarization is parallel. Fading due to polarization changes has been confirmed by experiments which have found that a minimum received signal level on one antenna element often coincides with a maximum signal level on an orthogonal element. This effect may be exploited by using a system of orthogonal antenna elements to improve system performance.

Many serial-tone HF modems incorporate adaptive equalization techniques such as those described in Annex 2. Some modems use a waveform in which a preamble is inserted periodically in the data stream. The preamble, which consists of known symbols, permits the instantaneous impulse response of the channel to be estimated. An adaptive equalizer can then use the estimated impulse response to combine energy from different paths having different delays. The impulse response is kept current through a least mean-squares update procedure to update the adaptive equalizer.

After equalization, the existence of several distinct propagation modes can be advantageous since they are unlikely to suffer from simultaneous fades thereby increasing the probability that some of the transmitted energy will be received. This phenomenon, known as mode diversity, can be exploited so long as the transmitted energy arriving at the receiver is sufficient to overcome the noise. Mode diversity gain can be best utilized if the path delay difference is large enough to avoid flat fading. By using the orthogonal antenna elements an artificial multipath of some fixed value can be created at the input of the demodulator. In this manner, polarization diversity gain can be achieved by taking advantage of the modem's ability to cope with inter symbol interference and improve performance through mode diversity.

Two different techniques have been considered. The first so called transmit diversity, uses two orthogonal antennas each driven by a separate but phase and frequency locked transmitter, with the baseband input to one of the transmitter delayed, and communicating with a receiver having a single antenna. The second, receive diversity, uses a single transmitter and antenna but two phase and frequency locked receivers connected to orthogonally polarized antennas. The receiver outputs were connected to a diversity combiner, again with one path delayed at baseband, which produced the input to the modem. The receiver outputs connected to a diversity combiner whose function was to combine the two signals to form the input signal to the modem. This simple combiner allows for receive diversity without modification to the modem. HF receivers usually employ an AGC to accommodate the wide dynamic range of a signal, so as to maintain an output close to some set level. When the input signal level to the receiver is reduced during a fade, the receiver gain is increased by the AGC action. The AGC voltage is therefore a convenient measure of instantaneous S/N . The design of the combiner should emphasize the component with better S/N at the expense of the component with the poor S/N . For this reason the AGC voltages of the receivers are utilized by the diversity combiner to determine the proportion of the two signals making up the sum. The resulting signal was then applied to the modem input.

For the system described in Annex 2, where the equalizer capability extends over 5 ms, a baseband delay of 2.7 ms has been found to be optimum. It has been found that the best results are obtained when the delayed path was the weaker path. This was due to the particular synchronization techniques used in the modem. For this reason, the procedure of using signal delay with the vertical antennas ensured the stronger signal preceded the weaker signal in both techniques.

2 Conclusions

This kind of diversity can significantly improve the performance of HF radio data systems. Transmit diversity can reduce the error rate by up to four orders of magnitude while receive diversity can improve the error rate by up to three orders of magnitude. The improvement offered by polarization diversity may be assessed by considering the amount of additional transmit power required to improve the performance of a system without diversity to the level achieved with

diversity. For a modem incorporating adaptive equalization, the use of transmit diversity is equivalent to a transmitter power increase of about 6-8 dB while the simple receive diversity is equivalent to a power increase of 3-4 dB. For a system employing transmitter diversity two 100 W transmitters could replace a 1 kW transmitter if 7 dB gain is achieved. This reduction in transmitter power coupled with the fact that polarization diversity can be implemented at either the transmit or receive end of a link without modification to existing modems, could represent a significant cost reduction. The type of diversity employed in a particular application will be dependent on the type of link involved. That is, the base station would likely employ diversity, while a remote station would not. Transmit and receive diversity in particular are useful when the performance of data communication links to mobile platforms or remote sites can be enhanced with additional antennas, receivers and transmitters at the base station location.

ANNEX 5

Data transmission at rates up to 4 800 bit/s over HF circuits using a serial PSK or quadrature amplitude modulation (QAM) transmission modem

1 General

This modem permits data transmission with information rates of up to 4 800 bit/s using 16-QAM within a 300-2 700 Hz bandwidth. The modulation method is switched, according to link quality, to QPSK at 2 400 bit/s or to BPSK at 1 200 bit/s.

2 Features

- Information rates of up to 4 800 bit/s are available.
- The information rate is switched to 2 400 bit/s (with QPSK) or 1 200 bit/s (with BPSK) according to link quality.
- The transmission bandwidth is within 300-2 700 Hz which permits 3 kHz channel separation.
- The protocol includes a synchronizing sequence of 28 symbols related to each data frame at 112 symbols so that the raw bit rates for transmission are 6 kbit/s, 3 kbit/s and 1.5 kbit/s.
- Switching of the bit rate with modulation mode is smoothly achieved only by mapping switch without changing the signalling rate.
- A bidirectional decision feedback equalizer (DFE) is used.

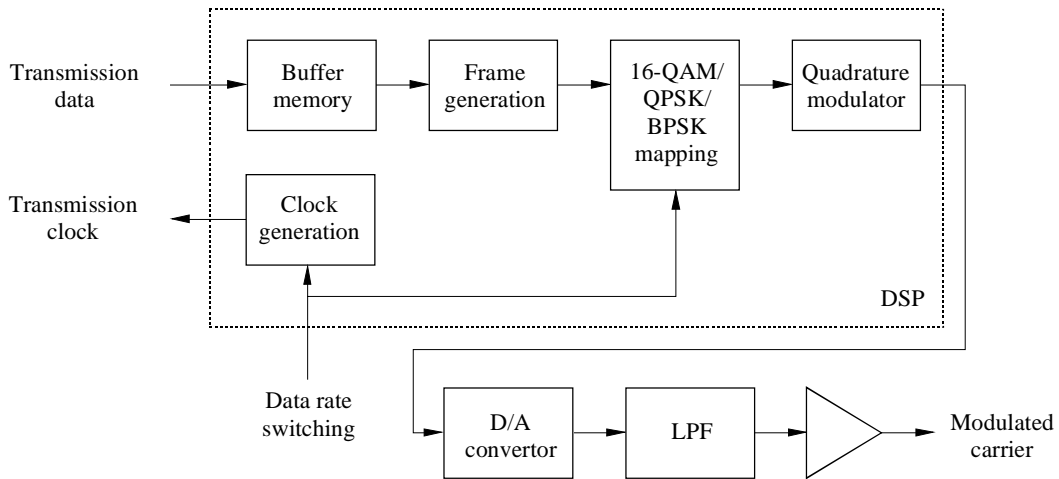
3 Specification

Modulation mode	16-QAM	QPSK	BPSK
Carrier bit rate (kbit/s)	6	3	1.5
User bit rate (kbit/s)	4.8	2.4	1.2
Signalling rate (kBd)	1.5		
Frame length	140 symbols (93.3 ms)		
Synchronization sequence	28 symbols		
Data length	112 symbols		
Equalization	Bidirectional DFE		

4 Block diagram of signal processing

Figures 13a and 13b show block diagrams of modulator and demodulator, respectively.

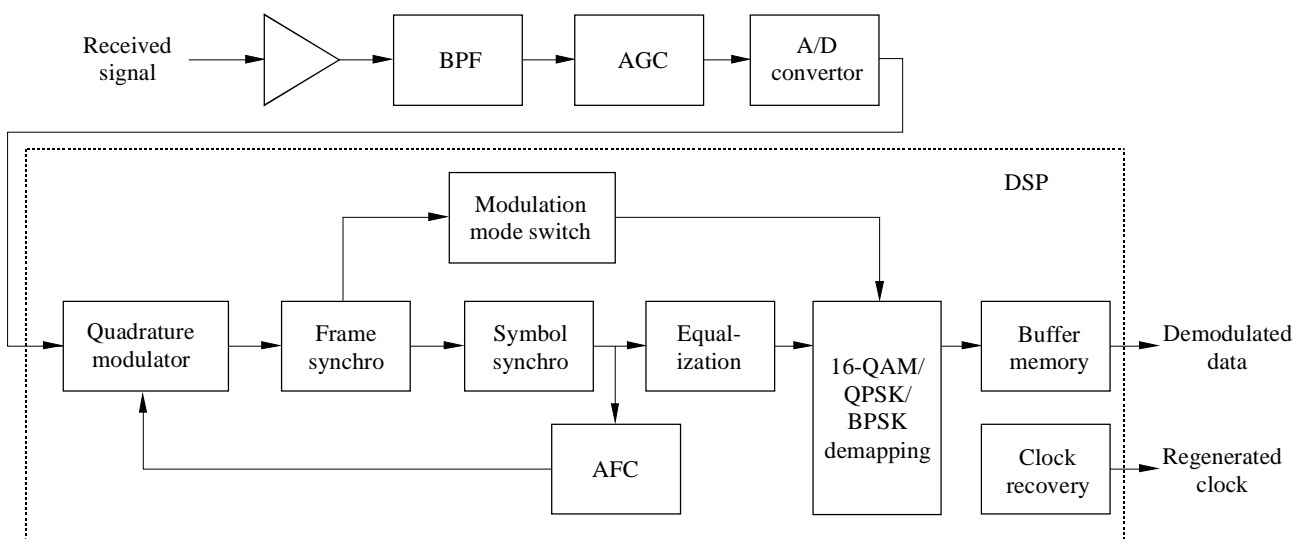
FIGURE 13a
Block diagram of modulator



DSP: digital signal processor
LPF: low pass filter

0763-13a

FIGURE 13b
Block diagram of demodulator



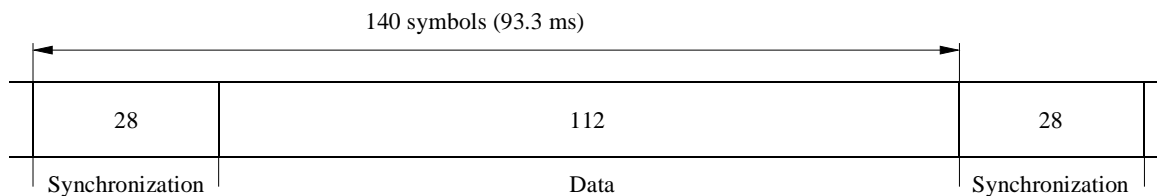
AFC: automatic frequency control
BPF: band-pass filter

0763-13b

5 Frame structure

The symbols to be transmitted are structured in recurring frames of 93.3 ms in length as shown in Fig. 14.

FIGURE 14
Frame structure



0763-14

6 Coding rule and constellation diagram of 16-QAM

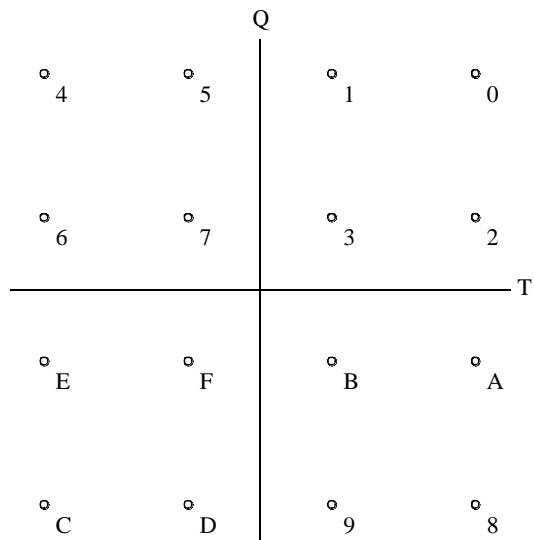
Table 2 and Fig. 15 show the coding rule for 16-QAM and the constellation diagram of 16-QAM, respectively.

TABLE 2
Coding rule of 16-QAM

Tetrabit	Symbol
0000	0
0001	1
0010	2
0011	3
0100	4
0101	5
0110	6
0111	7
1000	8
1001	9
1010	A
1011	B
1100	C
1101	D
1110	E
<div style="display: flex; align-items: center; justify-content: center;"> <div style="border-left: 1px dashed black; padding-left: 2px;">1111</div> </div>	F

Oldest bit
Most recent bit

FIGURE 15
Constellation diagram of 16-QAM



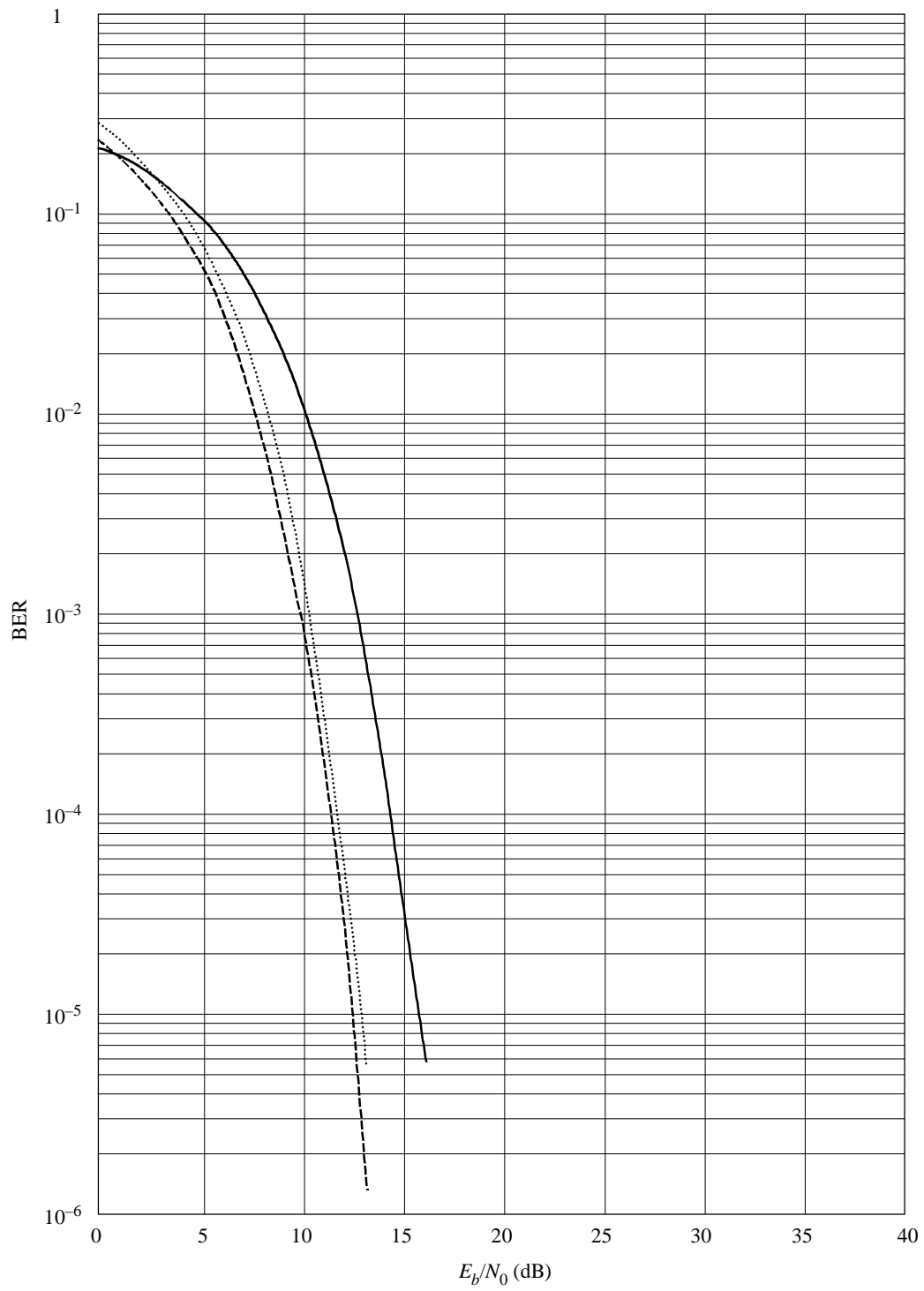
0763-15

7 Test data

In the test described below, the DFE equalizer used 14 feedforward taps and six feedback taps with the capability of equalizing over a maximum delay of five symbols. Figure 16 shows the non-fading tests in Gaussian noise. The fading tests were conducted in accordance with Recommendation ITU-R F.520 with equal path gains and path delay differences of 0.5-3 ms, and fading rate of 0.5 Hz. Figures 17-19 show the bit error test results in the fading environment.

FIGURE 16

BER versus noise spectral density for a no-fading channel with Gaussian noise



— 16-QAM
- - - QPSK
..... BPSK

FIGURE 17
 16-QAM BER versus noise spectral density for a fading channel

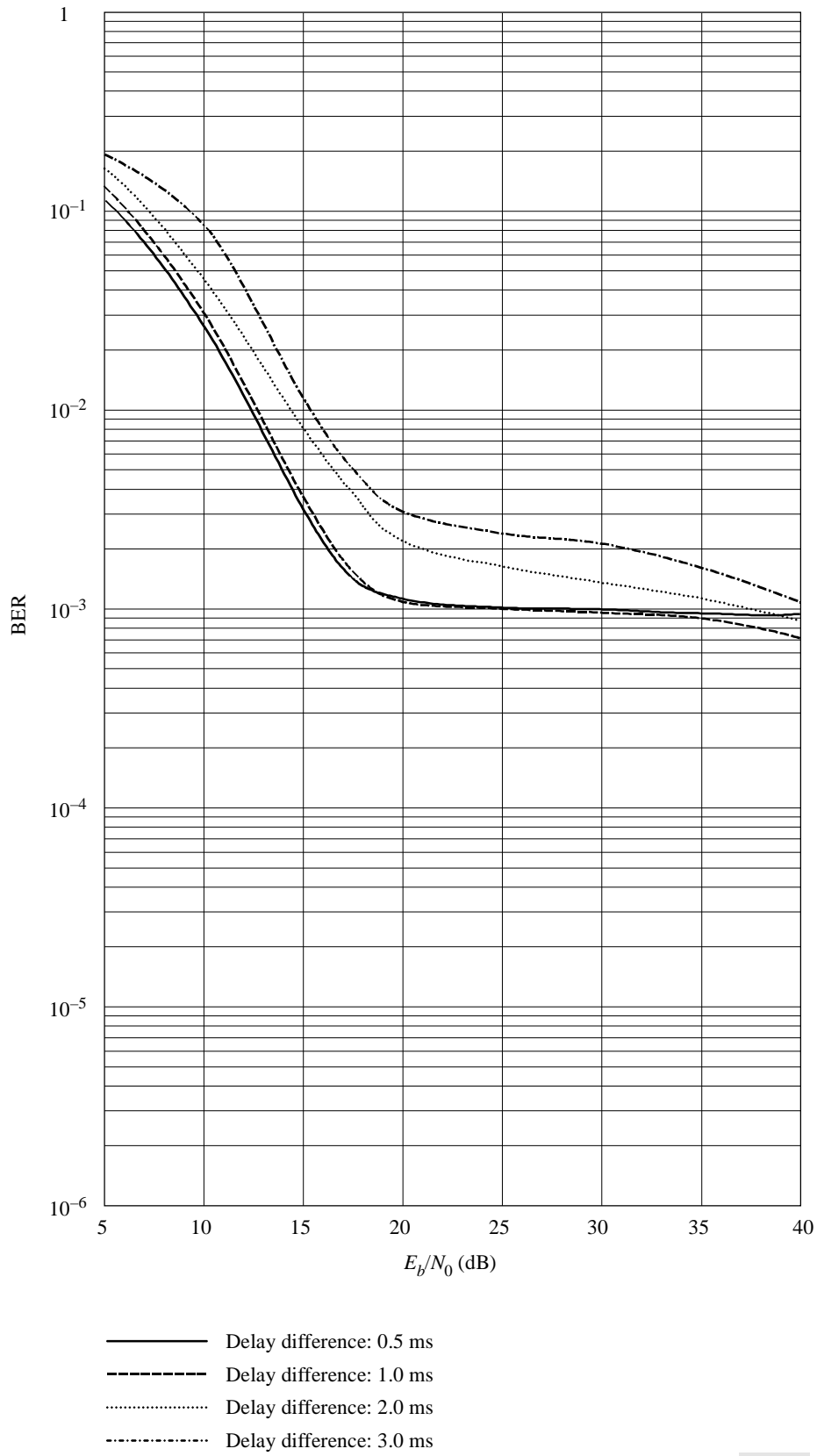


FIGURE 18
QPSK BER versus noise spectral density for a fading channel

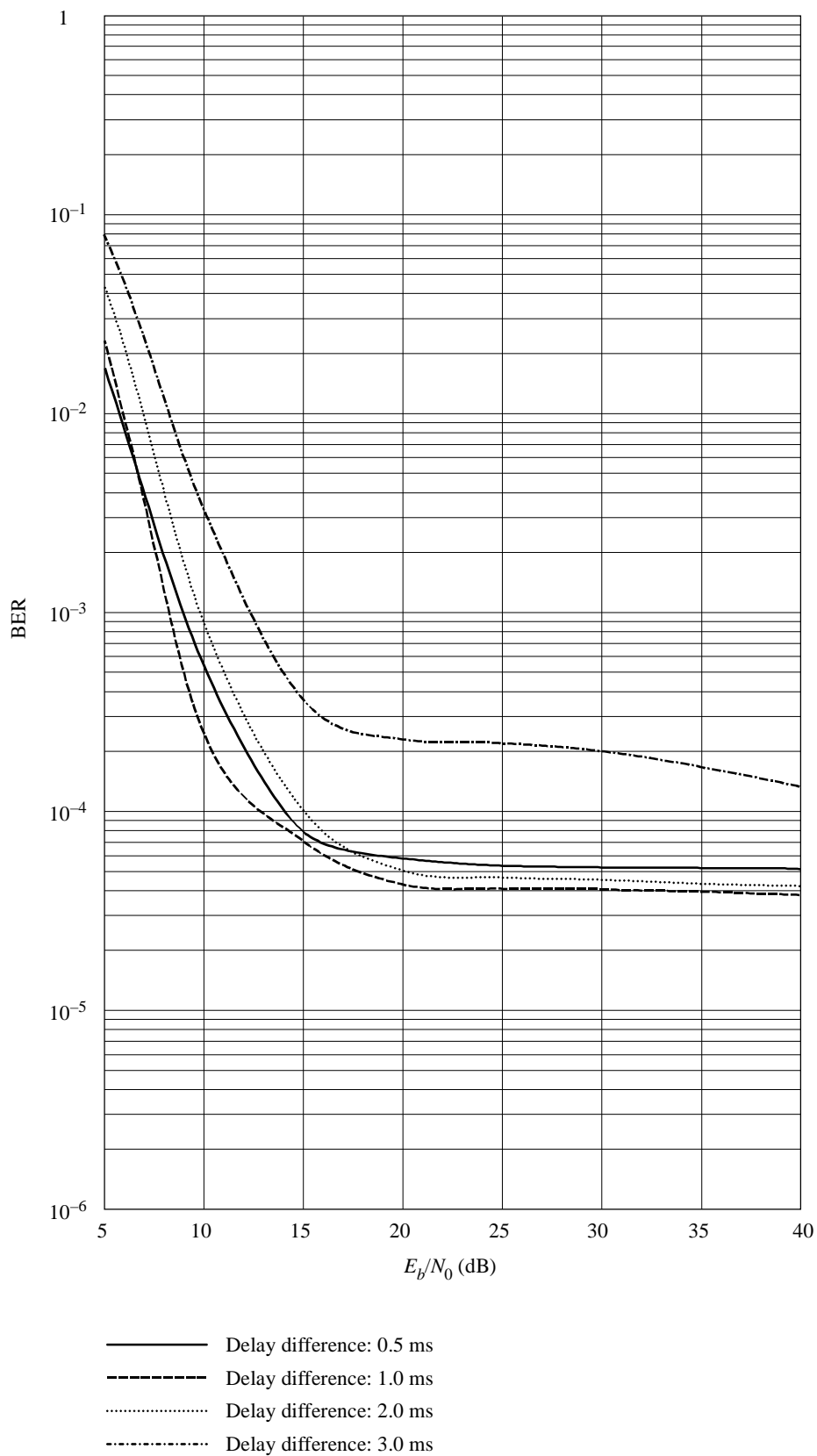


FIGURE 19
 BPSK BER versus noise spectral density for a fading channel

