RECOMMENDATION ITU-R M.1798[*, **]

Characteristics of HF radio equipment for the exchange of digital data and electronic mail in the maritime mobile service

(2007)

Scope

In accordance with Resolution 351 (WRC-03), the maritime community is asked to consider the use of new digital technology in the maritime mobile service (MMS) in the MF and HF bands. Resolution 351 (WRC-03) also cites that the need to use new digital technologies in the MMS is growing rapidly, and that the use of new digital technology on MF and HF frequencies allocated to the MMS will make it possible to better respond to the emerging demand for new maritime services. It is also noted that WRC-03 has modified Appendix 17 of the Radio Regulations (RR) to permit, on a voluntary basis, the use of various channels or bands for initial testing and future introduction of new digital technology. This Recommendation describes a MF/HF radio systems and a HF data transfer protocols currently used in the MMS for the exchange of data and electronic mail on frequencies of RR Appendix 17, and on non-RR Appendix 17 frequencies, providing a similar functional capability to narrow-band direct printing (NBDP) and many other features.

A method of providing completely transparent user interoperability is also described.

The ITU Radiocommunication Assembly,

considering

a) that Resolution 351 (WRC-03) invites ITU-R to finalize studies currently ongoing:

– to identify the technical characteristics necessary to facilitate the use of digital systems in the MF and HF bands allocated to the maritime mobile service (MMS), taking into account any relevant ITU-R Recommendations;

– to identify the digital system(s) to be used in the MF/HF bands by the MMS;

b) that IMO invited ITU to develop a Recommendation describing the technical characteristics of such systems (data exchange on HF), taking into account *resolves* 1 of Resolution 351 (WRC-03);

c) that there are already several HF digital radio systems working worldwide, and that there is a need to specify the technical characteristics of HF radio systems and equipment for the exchange of HF data and electronic mail on mobile frequencies, including RR Appendix 17 frequencies;

d) that the system should be able to operate with standard global maritime distress and safety system (GMDSS) ship radio equipment;

e) that there are existing and developing global and regional HF electronic mail services operating on RR Appendix 17 frequencies and mobile frequencies outside of RR Appendix 17 (the use of mobile frequencies outside of RR Appendix 17 by the MMS is in conformity with ITU rules);

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[*] This Recommendation should be brought to the attention of the International Maritime Organization (IMO).

[**] *Note by the BR Secretariat* – Page 7 (section before control – over (0x86)) page 8 (last paragraph of section IRS response format), § 4.1.9 and last legend of Fig. 27 were amended editorially in February 2008.

f) that the use of software-defined radios will in future bring economical, technical and spectrum efficiency benefits, and that it should be possible to introduce the use of such radios without the need for further regulatory changes;

g) that a high-speed data service over HF radio may be useful for low-level graphics, and the updating of Electronic Chart Display and Information Systems (ECDIS);

h) that HF data services will enhance operational efficiency and maritime safety;

j) that the introduction of new digital technology in the MMS shall not disrupt the distress and safety communications in the MF and HF bands including those established by the International Convention for the Safety of Life at Sea, 1974, as amended;

k) that the limited use of NBDP remains for distress communications in the polar regions (A4), since no coverage from geostationary-satellite networks provides service to maritime;

l) that HF data services may require bandwidths greater than 3 kHz;

m) that a maritime HF data system providing automatic connection with internet service providers would improve traffic-handling efficiency;

n) that HF has the potential to provide greater coverage in Arctic NAVAREAS than either Inmarsat EGC or 518 kHz NAVTEX;

o) that there is a continuing need for ship-to-ship digital interoperability;

p) the continued expansion of HF maritime digital data services will generate increasing demands for maritime mobile RR Appendix 17 spectrum;

q) that multiple standards for electronic mail could be used to encourage technological development, thus encouraging continued competition, so that users may benefit from continued advances in technology whilst noting the need for interoperability across networks, particularly for future distress and safety purposes, and the distribution of Maritime Safety Information (MSI),

noting

1 that the characteristics for HF data services described in Annex 1 can be considered as meeting the requirements for exchange of digital data and electronic mail in the MMS[1],

recommends

1 that the examples of HF maritime data services, characteristics and modem protocols given in Annex 1 should be used in systems for the transmission and reception of data to and from ships using HF;

2 that system interoperability should be achieved for the transmission of data messages in both the ship-to-shore and shore-to-ship direction should be achieved at the internet protocol (IP) level;

3 that in order to maintain ship-to-ship interoperability and compatibility with existing GMDSS equipment, the system should be capable of automatically accommodating radiocommunications in accordance with Recommendations ITU-R M.476 and ITU-R M.625 in both the forward error correction (FEC) and automatic repeat request (ARQ) modes;

4 that, if used in the GMDSS, this system should meet the applicable requirements of the IMO.

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[1] Recognizing the need to comply with Chapter VII of the Radio Regulations.

Annex 1

Examples of HF maritime data services, characteristics and modem protocols

1 Introduction

This Annex details the two following HF electronic mail systems currently in use:

System 1: HF data services modem protocol using orthogonal frequency division multiplexing (OFDM);

System 2: Electronic mail system using the Pactor-III protocol, including system used by the Global Link Network (GLN).

2 System interoperability

Ship-to-shore

In the ship-to-shore direction interoperability is maintained by the internet service provider (ISP) at the IP level. Typically, a ship will enter an e-mail, with or without attachments in the e-mail system and then click on the “send” button in the way that is familiar to all of us. This applies to any location, pole-to-pole, at any time.

Shore-to-ship

In the system as described in this Recommendation, there are no interoperability concerns on the part of the shore-side user. The shore-based sender of an e-mail to a ship can merely:

– click on the “reply” button; or

– address the message to shipname@xxx.com or callsign@xxx.com.

The e-mail will be delivered via whatever system the ship is using. If there is a system failure, there will be an automatic re-route via an alternate system. These automated decisions are based on the contents of an extensive database. Consequently, the e-mail may be delivered via HF or an alternate satellite-based system. If there is an overall system failure, addressing problem or non-delivery for any reason, the system support operators will be alerted and take corrective action. This ensures that shore-based users need not be concerned about what system or network the ship is using. They need only address the e-mail and click on “send”.

3 System 1 – HF data services modem protocol using orthogonal frequency division multiplexing (OFDM)

Overview

This Recommendation describes the architecture for an OFDM modem for a HF channel using digital signal processing (DSP). The algorithm definition and description of the implementation is provided. This includes the protocol, modulator and demodulator definition. The final section outlines how frequencies are selected and used in a spectrum-efficient manner.

There are two basic approaches for implementation of a wideband modem, single carrier and multicarrier. The OFDM modem described and in use is a multicarrier approach. The main advantage of using a multicarrier approach is that an equalizer is not required for estimating the fading channel, because the individual subcarrier bandwidth is small and can tolerate moderate fading. Thus, the multicarrier approach is a less complex implementation. Also, the multicarrier approach was selected to make the individual subcarriers similar to narrow-band DATAPLEX. The disadvantage of a multicarrier approach is that it is more sensitive to frequency offset and oscillator phase noise.

HF modem protocol

Introduction

The OFDM waveform uses 32 carriers to transmit 32 blocks every 1 520 ms. Like Recommendation ITU-R M.625 TOR transmissions, OFDM is a half-duplex communication protocol where, at any given time, one station is the information-sending station (ISS) and the other is the information-receiving station (IRS). The basic timing cycle is fixed, and the original calling or MASTER station establishes the cycle timing.

In the following sections, this paper describes the OFDM basic timing cycle, the block formats, and the basic link operations such as OVER, END and link establishment.

OFDM modulation

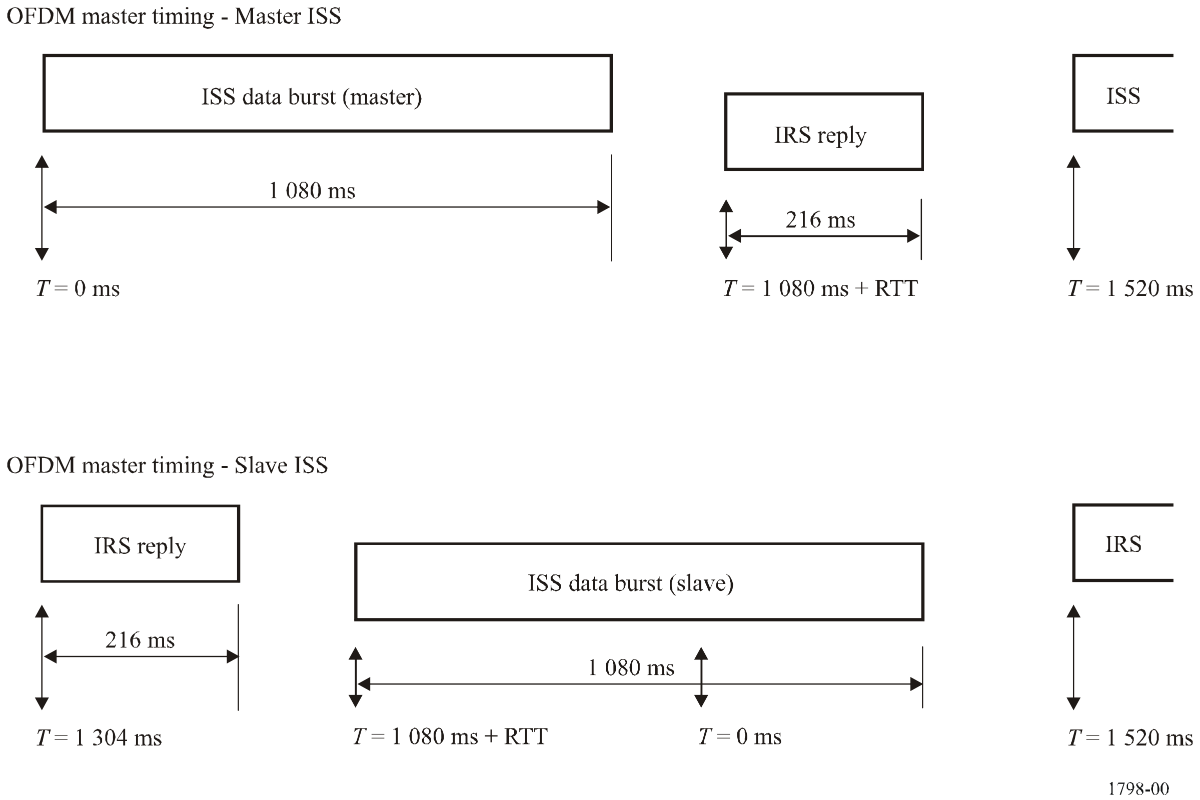
The OFDM waveform uses 32 carrier frequencies centred at 1 700 Hz. A full description of the waveform is in the following sections describing the modulator and demodulator.

All OFDM transmissions use the 32 carrier ($N = 32$), 4 phase ($M = 4$) waveform where the ISS station sends one long data blocks per carrier for a total of 32 data blocks per burst. The IRS station responds with a 32 carrier ($N = 32$), 4 phase ($M = 4$) short burst containing 2 bytes per carrier for a total of 64 bytes.

Frame timing

Like Recommendation ITU-R M.625 TOR, OFDM is a half-duplex protocol where one station is the ISS and the other is the IRS. When linked, the OFDM cycle time is fixed at 1 520 ms; the ISS transmits a 1 080 ms long data burst, and the IRS replies with a 216 ms short response burst. The basic timing cycle at the MASTER station is summarized below for MASTER-ISS and SLAVE-ISS. NOTE − RTT is the round-trip propagation and SLAVE station processing time.

The OFDM $T = 0$ cycle time reference is established by the MASTER station when the link starts. When ISS, the MASTER station always starts transmitting at $T = 0$, and the SLAVE station response must be completely received within the 440 ms receive interval immediately following the MASTER’s 1 080 ms data burst. The SLAVE station always transmits the IRS reply as soon as it can after it receives the end of the MASTER ISS burst. When the MASTER is IRS, the 216 ms IRS reply starts 1 304 ms into the 1 520 ms cycle time so that the end of the reply occurs at the same time the MASTER ISS data burst would have ended. The SLAVE data burst starts at the same time in the cycle as the SLAVE IRS reply. The OFDM cycle timing philosophy follows the example set by Recommendation ITU-R M.625, except that the OFDM cycle time allows a greater path distance (224 ms versus 170 ms) between the two linked stations.



ISS block format

The OFDM protocol uses the ISS block illustrated below to transmit both data bytes and control messages to the IRS station. Every ISS transmission sends one data block on each of 32 carriers for a total of 32 blocks per long burst. Since the ISS sends a maximum of 32 blocks with 10 bytes per block every 1 520 ms, the resulting maximum data throughput for OFDM $N = 32$ $M = 4$ is about 210 bytes or 1 684 bit/s.

ISS data block

SEQ_NR | LEN (11 bits) (5 bits)	**DATA** (10 bytes)	**CRC** (2 bytes)

SEQ_NR – 11-bit block sequence number 1 to 0x7FF
0x000 means discard this block

LEN – 0 to 10 is the number of valid data bytes in the block
31 signals a CONTROL block

DATA – 0 to 10 data bytes when LEN is 0 to 10
CONTROL block when LEN is 31

CRC – 16-bit CRC sequence

Each data block starts with an 11-bit sequence number (SEQ_NR) that is used to correctly order the blocks at the IRS end of the link. The sequence number is incremented from 1 to 2 047 (0x7FF) with every new data or control block transmission so that the IRS station can reconstruct the entire data transmission by presenting the blocks in the correct order at the receive end. The sequence number rolls over from 2 047 to 1 after the 2 047th block has been encoded. The sequence number of a control block indicates when the control block should be decoded. The sequence number is set to 1 when the link starts, and it is not changed during OVERs.

During the link, the ISS station must ensure that no more than MAX_SEQ_NR_DIFF sequence number blocks are outstanding at any time, where MAX_SEQ_NR_DIFF is a programmable value less than (2 047 − 64) or 1 983. In other words, the difference between the oldest and newest block sequence number in any given ISS long burst must be less than or equal MAX_SEQ_NR_DIFF. This restriction is meant to limit the number of buffered blocks at the IRS end, and to allow the link to "catch up" if, for some reason, one or more blocks continue to fail to decode error-free at the IRS end.

The protocol allows the ISS station to repeat blocks in the same long burst. If the ISS station approaches the MAX_SEQ_NR_DIFF difference between the oldest and newest block sequence numbers in any given long burst, the oldest blocks should be repeated in the remaining open long burst slots to improve the probability that the block is received correctly. At any time, the ISS station may repeat current blocks if there are no new data blocks pending.

The 0000 sequence number is a special case. When a block is transmitted with a 0000 sequence number, this block can be discarded by the IRS station without any further decoding. At the end of an ISS transmission, for example, the 0000 blocks may be used as filler for all blocks after the last block containing valid data. The significance of the 0000 block will become apparent later when discussing the ARQ operation when the IRS station requests the retransmission of corrupted data blocks. If the ISS station transmits a 0000 block, it does not need to retransmit that block if the IRS station signals an error for that block. Note that the ISS station may also repeat current blocks rather than transmit 0000 blocks.

The 5-bit length (LEN) field serves a dual purpose. If LEN is a number between 0 and 10 it indicates the number of valid data bytes in the DATA portion of the block. The bytes after the first LEN bytes in the DATA portion of the block should be ignored. Note that 00 is a valid data block length that can be used to signal an idle or no data block. Unlike the 0000 sequence block, an idle block must be retransmitted if the IRS station signals an error for that block.

When LEN is set to 31, the block is identified as a CONTROL block, and the control message is contained in the data portion of the block. Like data blocks, if the IRS station signals an error receiving this block, it must be retransmitted. In addition, the ISS station may repeat CONTROL blocks in the same long burst just as it may repeat DATA blocks. Of course, the repeated block must have the same block sequence number.

The 16-bit CRC at the end of all blocks is a standard ITU-T polynomial remainder calculated over the entire block from the start of the sequence number field to the end of the data field. After the CRC is XOR'd with 0xFFFF, the two CRC bytes are transmitted, low byte first, at the end of the block. At the IRS location the CRC checker is initialized to 0xFFFF, and the calculated CRC remainder from the sequence number byte to the end of the block will equal 0xF0B8 if no errors have occurred.

Data blocks

In the OFDM ISS data blocks the LEN parameter is set to the number of valid data bytes in the block: 0 to 10 bytes.

OFDM data block

SEQ_NR | LEN (11 bits) (5 bits)	**DATA** (10 bytes)	**CRC** (2 bytes)

LEN – 00 to 10 valid data bytes

In any given ISS burst the data blocks may be assigned to carriers in any order. It is incumbent on the IRS station to reassemble the original data message in the correct order based on the sequence numbers in data blocks.

If the ISS station does not have enough blocks to fill all 64 slots, the ISS station may repeat current blocks in the remaining slots starting with the oldest block. The repeated blocks give the IRS station a second chance to decode all blocks error-free. Alternatively, the ISS station can fill the unneeded blocks is 0000 sequence number blocks, and those blocks will be discarded at the IRS end.

The ISS station must never have a range of more than MAX_SEQ_NR_DIFF block sequence numbers outstanding where MAX_SEQ_NR_DIFF is a programmable value. This means that in any given ISS long burst, the difference between the oldest sequence number and the newest number, taking the count wrap at 2 047 into account, must be less than or equal MAX_SEQ_NR_DIFF.

Control blocks

The OFDM protocol transmits control messages by setting the LEN field to 31 and loading the command in the first byte of the DATA field. The sequence number field is set to the next available number. All control frames are retransmitted if the IRS station fails to decode the block error-free.

ODFM has three control messages: MY_CALL, OVER and END.

OFDM control block

SEQ_NR | 11111 (11 bits) (5 bits)	**CONTROL | IDLE FILL PATTERN** (1 bytes) (9 bytes)	**CRC** (2 bytes)

SEQ_NR – 11-bit sequence number; it cannot be 0000

LEN – 31 for control block

CONTROL – OVER or END control code

IDLE FILL PATTERN – 10101010 (repeated 9 times)

Control blocks may be sent by the ISS at any time, and the IRS station must recognize the control command at the point it appears in the reconstructed serial data. For example, when the ISS OVER command is transmitted, no data blocks with a higher sequence number than the OVER command should be transmitted since the ISS station will shortly become IRS. The ISS station should generate the command block only once, but it can repeat this control block in unassigned carrier slots.

The CONTROL byte codes are shown below.

CONTROL – OVER (0x86)

1 0 0 0 0 1 1 0

CONTROL – END (0x98)

1 0 0 1 1 0 0 0

CONTROL – MYCALL (0xE0)

1 1 1 0 0 0 0 0

Typical OVER and END control blocks are shown below:

OVER CONTROL BLOCK

SEQ_NR | 11111	**10000110**	**IDLE FILL PATTERN**	**CRC**

END CONTROL BLOCK

SEQ_NR | 11111	**10011000**	**IDLE FILL PATTERN**	**CRC**

OFDM acquisition burst

The ISS station and the IRS station send a 1 700 Hz tone before the start of each burst. This tone is used to determine the frequency offset.

IRS response format

When a station is the IRS, it receives 32 data blocks from the ISS station every 1 520 ms, and it responds with an ACK or NAK signal for each of the blocks. In addition, the IRS response sends link control commands to OVER the link and to END the link. The IRS response message is transmitted as a 216 ms short OFDM block sent in 32-carrier ($N = 32$) 4-phase ($M = 4$) format. There are 2 bytes sent per carrier; two bytes per carrier are assigned to each of the data blocks on that same carrier in the ISS long burst transmission.

On each carrier, only one IRS response code is transmitted for the data block received from the ISS station on the same carrier.

BLOCK 1 RESPONSE (16 bits)

The IRS station sends the following response codes:

ACK/NAK

FORCED_OVER

END_ACK

Any response other than one of these is treated as if a NAK was received. In this section, the coding for each of these response codes is listed along with a brief description.

ACK/NAK

The IRS station decodes and calculates the CRC for each of the 32 incoming data blocks in the ISS long burst. If the CRC indicates that the block has been received without error, the IRS station responds with an ACK on the same carrier. In an error is detected, a NAK is transmitted. At the ISS end, an ACK signals the successful transmission of a block, and that block is dropped from the transmit queue. A NAK, on the other hand, forces the ISS station to retransmit the block on a different carrier. If the IRS station receives a block containing a sequence number that it has already acknowledged, it sends another ACK and discards the block. Any unknown response is treated by the ISS as if it was a NAK.

ACK Code (0x56A9)

0 1 0 1 0 1 1 0 1 0 1 0 1 0 0 1

NAK Code (0xA956)

1 0 1 0 1 0 0 1 0 1 0 1 0 1 1 0

The ACK/NAK responses are used by the ISS and IRS stations to gauge the quality of the link and determine when to abort the link. With OFDM we have 32 individual ACK/NAK responses every cycle and deciding when to drop the link is somewhat more complicated. For the OFDM initial implementation, we use the number of consecutive blocks where NO blocks decode correctly to increment the error counter. If the IRS and ISS stations see MAX_BLK_ERR transmit cycles without one single block ACK, the link will be aborted, where MAX_BLK_ERR is a programmable value. MAX_BLK_ERR equal to 20 is about 30 seconds. Any block ACK will reset the error count to 0.

FORCED_OVER

Typically, the OFDM ISS station controls the switch from ISS to IRS by transmitting the OVER control block to the IRS on one or more carriers. However, the IRS station can force an OVER by transmitting the FORCED_OVER code word. To avoid an outstanding data block problem, the FORCED_OVER code word will only be transmitted when the last block from the ISS on that carrier was received without error.

FORCED_OVER Code (0x6A95)

0 1 1 0 1 0 1 0 1 0 0 1 0 1 0 1

END_ACK

The IRS transmits the END_ACK code word in response to the ISS END control block to signal the end of the link. The END_ACK will be transmitted in response to each ISS END control block to ensure that the ISS station received the acknowledgment code word. When the ISS station receives one or more END_ACK response messages, it immediately goes to STANDBY even if there are outstanding unacknowledged data blocks. The IRS station uses the END_ACK response to force termination of the link immediately.

END_ACK Code (0x956A)

1 0 0 1 0 1 0 1 0 1 1 0 1 0 1 0

OFDM OPERATION

In this section, the important protocol exchanges between ISS and IRS are discussed. Here, the data and control blocks and response code words that have been defined in the earlier section are combined to create the OFDM protocol. This section describes the ISS-IRS exchange during data block transfer, link OVER, link speed change, link END, and link CALLING operations.

ISS-IRS exchange

During an OFDM link, one station is the ISS and the other is the IRS. The ISS station transmits data blocks, and the IRS acknowledges those blocks when they are received error-free. The ACK and NAK code word responses from the IRS signal the ISS which blocks to send in the next burst.

Since OFDM transmits 32 blocks per burst, a procedure must be defined to assign data blocks to specific waveform carrier frequencies. The transmit data bytes fill 10-byte data blocks and the sequence number for each block indicates the order of these blocks. When an actual transmit frame is built, the individual data blocks are assigned, in order, starting with the first block on the first carrier, the second block on the second carrier and so on until the first 32 transmit blocks are assigned a carrier. The transmit block assignments are shown below for a typical first transmission.

The block sequence numbers start with block 0001 in the first data block after a link is established, and the numbers increment with each transmit block built until the end of the link. After the 2 047th block, the sequence number wraps to block 0001 again.

ISS OFDM TRANSMIT BURST

Carrier 1	Block 0001	CRC
Carrier 2	Block 0002	CRC
Carrier 3	Block 0003	CRC
Carrier 4	Block 0004	CRC
…	…	…
Carrier 30	Block 0030	CRC
Carrier 31	Block 0031	CRC
Carrier 32	Block 0032	CRC

If all of the blocks are decoded error-free, the IRS transmits a short response burst containing an ACK for each data block on each carrier. The ACKs are not sequence numbered.

IRS OFDM RESPONSE BURST

Carrier 1	ACK (for block 1)
Carrier 2	ACK (for block 2)
Carrier 3	ACK (for block 3)
Carrier 4	ACK (for block 4)
…	…
Carrier 30	ACK (for block 30)
Carrier 31	ACK (for block 31)
Carrier 32	ACK (for block 32)

When a corrupted data block is detected, the IRS station sends a NAK response for that block on the same carrier. The ISS station retransmits every data block that the IRS has not ACK'd, including those blocks where no valid IRS response was decoded. To maximize the chance that the block will get through the next time, the ISS station will retransmit blocks on a carrier where the last block was properly ACK'd. For example, re-sent blocks are assigned first to carriers where both blocks were ACK'd the previous cycle, then to carriers where only one block was ACK'd the previous cycle. Moving the data blocks should keep data moving even if one or more of the carriers is blocked by interference. New blocks are added in the remaining open block slots, starting first with the carriers where both blocks were previously ACK'd then continuing with the carriers where one block was previously ACK'd. If there are no new blocks, then current blocks, starting with the oldest sequence number, may fill the open carrier slots.

For example, if we consider the case where we have only four carriers and two blocks are corrupted, the ISS station will retransmit the blocks as shown below:

ISS

Carrier 1	DBlock 0001	CRC
Carrier 2	DBlock 0002	CRC
Carrier 3	DBlock 0003	CRC
Carrier 4	DBlock 0004	CRC

IRS

Carrier 1	ACK (for block 1)
Carrier 2	**NAK** (for block 2)
Carrier 3	ACK (for block 3)
Carrier 4	**NAK** (for block 4)

ISS

Carrier 1	DBlock 0002	CRC
Carrier 2	DBlock 0005	CRC
Carrier 3	DBlock 0004	CRC
Carrier 4	DBlock 0006	CRC

IRS

Carrier 1	ACK (for block 2)
Carrier 2	ACK (for block 5)
Carrier 3	ACK (for block 4)
Carrier 4	ACK (for block 6)

Note that the retransmitted blocks have been moved to block positions where the block was ACK'd in the last cycle. In the case above, DBlock 0007 is sent as the first block in Carrier 4 rather than Carrier 2 because there was an error in the Carrier 2 position in the last burst. It makes sense to fill the "good" positions first and leave the previously NAK'd positions to last to increase the probability that a block is transferred successfully. If a carrier is completely masked due to some channel interference or bandwidth limitation on one of the radios, new data blocks should first be assigned to those carriers that are getting through. The example below shows how this might apply in our simple case:

ISS

Carrier 1	DBlock 0001	CRC
Carrier 2	DBlock 0002	CRC
Carrier 3	DBlock 0003	CRC
Carrier 4	DBlock 0004	CRC

IRS

Carrier 1	**NAK** (for block 1)
Carrier 2	ACK (for block 2)
Carrier 3	ACK (for block 3)
Carrier 4	**NAK** (for block 4)

ISS

Carrier 1	DBlock 0005	CRC
Carrier 2	DBlock 0001	CRC
Carrier 3	DBlock 0004	CRC
Carrier 4	DBlock 0006	CRC

IRS

Carrier 1	ACK (for block 5)
Carrier 2	ACK (for block 1)
Carrier 3	ACK (for block 4)
Carrier 4	**NAK** (for block 6)

In this example, new blocks are assigned to Carriers 1 and 4 last since those carriers showed errors on the previous transmit cycle. If Carrier 4 is failing to pass blocks due to a bandwidth limitation, then we will resend Blocks 12 and 13 since all earlier blocks were transferred without error.

If there is no data to transmit, the ISS station can send blocks with the sequence number set to 0000. The IRS station ignores these blocks, and they do not need to be retransmitted if the IRS station returns a NAK for that block. As shown below, the ISS station may also repeat current blocks, starting with the oldest, in the remaining slots to increase the probability that the block will be received error-free.

If the ISS station has fewer than 32 blocks to transmit, the ISS station may repeat current blocks in the remaining open carrier blocks. Since the IRS station must use the sequence number to reconstruct the serial byte steam, a second block with the same block sequence number will be ignored. Repeating the blocks in the ISS long burst provides a second chance for the block to be received error-free.

ISS

Carrier 1	DBlock 0001	CRC
Carrier 2	DBlock 0002	CRC
Carrier 3	DBlock 0003	CRC
Carrier 4	DBlock 0004	CRC

IRS

Carrier 1	**NAK** (for block 1)
Carrier 2	ACK (for block 2)
Carrier 3	ACK (for block 3)
Carrier 4	**NAK** (for block 4)

In this example, the ISS station has 5 blocks to send and it repeats Blocks 1 to 3 in the remaining blocks. At the IRS end, the first DBlock 0001 is NAK'd, but the second copy is received error-free. The ISS station does not need to resend DBlock 0001. The second copy of DBlock 0003 is NAK'd, but the first copy was received OK; the ISS station does not need to resend this block. Note that DBlock 0004 is NAK'd, and the ISS station will need to resend this block since it was sent only once in the long burst.

No attempt is made by the IRS to compare multiple copies of blocks with the same sequence number. It is assumed that the first block received with a correct CRC is a valid block, and that block is queued for output to the serial port. The IRS should also ACK all blocks received error-free even if it is a repeated block.

Flow control

The ODFM protocol does not include any specific link level flow control codes to allow the IRS station to halt ISS block transmission. Flow control is required; however, if the IRS station is unable to empty receive block buffers due to external serial port or USB port flow control activation. If the external flow control stops RX data output for an extended period of time, the IRS receive buffers may fill leaving no place to store new ISS data blocks.

When the IRS needs to slow the ISS block transfer rate, it can NAK some of all of the ISS long burst blocks even if the block CRCs are correct. If all of the blocks are NAK'd the ISS station will repeat all blocks in the next long burst. Note that halting the link data transfer with NAKs for a long period of time may cause the ISS station to abort the link.

OVER

The link OVER can be initiated from the ISS or the IRS end. The ISS requests the OVER by transmitting the OVER control command as one of the long burst data blocks. The ISS station can request the OVER at any time, but it should stop building new transmit data blocks after the OVER is issued. When the IRS station receives the OVER control command, it checks to confirm that all data block sequence numbers up to the OVER control block sequence number have been received. If there are no missing blocks, the IRS station sends the FORCED_OVER response message instead of ACK for all correctly decoded blocks and NAKs for the bad blocks. If there are missing blocks, the IRS station continues to send ACK/NAK response messages until all missing blocks have been received correctly, then it sends the FORCED_OVER response message instead of an ACK for all correctly decoded blocks. Note that there is no guarantee that those blocks with sequence numbers after the OVER block will be acknowledged before the link OVER occurs. The ISS end must keep track of the outstanding blocks.

The ISS station should fill all data blocks after the OVER with blocks containing the sequence number 0000 so that those blocks will not need to be re-sent while waiting the IRS station to start the OVER sequence. The ISS station can also repeat current data blocks in the remaining open slots.

The IRS station can force an OVER at any time by sending at least one FORCED_OVER response message instead of an ACK when responding to the ISS long burst. When the ISS station detects the FORCED_OVER, it immediately turns the link around and it keeps track of those blocks that have not been acknowledged. All outstanding blocks will be transmitted after the next OVER.

ISS

Carrier 1	DBlock 0005	CRC
Carrier 2	DBlock 0006	CRC
Carrier 3	CBlock 0007 OVER	CRC
Carrier 4	DBlock 0000	CRC

IRS

Carrier 1	ACK (for block 5)
Carrier 2	ACK (for block 6)
Carrier 3	ACK (for block 7)
Carrier 4	**NAK** (for block 8)

ISS

Carrier 1	DBlock 0000	CRC
Carrier 2	DBlock 0001	CRC
Carrier 3	DBlock 0004	CRC
Carrier 4	DBlock 0000	CRC

IRS

Carrier 1	**NAK**
Carrier 2	FORCED_OVER
Carrier 3	FORCED_OVER
Carrier 4	**NAK**

IRS

Carrier 1	NAK
Carrier 2	NAK
Carrier 3	NAK
Carrier 4	NAK

ISS

Carrier 1	DBlock 0010	CRC
Carrier 2	DBlock 0011	CRC
Carrier 3	DBlock 0012	CRC
Carrier 4	DBlock 0013	CRC

IRS

Carrier 1	ACK (for block 10)
Carrier 2	ACK (for block 11)
Carrier 3	ACK (for block 12)
Carrier 4	ACK (for block 13)

END

Either the ISS or IRS station can terminate the OFDM. Typically, the ISS ends the link by transmitting one END control block as the next block after the last data block. When the IRS station receives the END control block, it confirms that all data blocks with sequence numbers up to the END block have been received. If there are no outstanding blocks, the IRS station transmits a short burst with all slots set to END_ACK. If there are outstanding blocks, the IRS continues to send ACK/NAK response messages until all of the outstanding blocks are received correctly. Note that any data blocks that the ISS station transmits with sequence numbers after the number in the END block are discarded.

The ISS station should encode all blocks after the END control message using a sequence number of 0000 so that they will not be retransmitted.

When the ISS station receives four or more END_ACK response messages in the short block, it stops transmitting immediately and returns to STANDBY. The IRS station repeats an END_ACK frame two times after the last END control block is received to ensure that the ISS station receives the END_ACK message.

The IRS station issues the END_ACK response message when it wants to force link termination. When the ISS station receives the END_ACK response message, it immediately stops transmitting and returns to STANDBY even if there are outstanding data blocks.

ISS

Carrier 1	DBlock 0005	CRC
Carrier 2	DBlock 0006	CRC
Carrier 3	CBlock 0007 END	CRC
Carrier 4	DBlock 0000	CRC

IRS

Carrier 1	ACK (for block 5)
Carrier 2	ACK (for block 6)
Carrier 3	ACK (for block 7)
Carrier 4	**NAK** (for block 8)

ISS

Carrier 1	DBlock 0000	CRC
Carrier 2	DBlock 0000	CRC
Carrier 3	DBlock 0000	CRC
Carrier 4	DBlock 0000	CRC

IRS

Carrier 1	END_ACK
Carrier 2	END_ACK
Carrier 3	END_ACK
Carrier 4	END_ACK

IRS

Carrier 1	END_ACK
Carrier 2	END_ACK
Carrier 3	END_ACK
Carrier 4	END_ACK

OFDM link terminated

CALLING

The DATAPLEX link is established when the master station calls a remote station using a 9-byte CALLING block transmitted with a FSK100 format. A unique 2-byte sync code at the beginning of the block identifies the CALLING block and establishes the link timing. This CALLING block is repeated every 1 020 ms, the DATAPLEX cycle time.

The remote station SELCAL is transmitted in 4.5 bytes by packing two SELCAL digits per byte; all SELCALs must have 9 digits with values of 0x0 to 0x9. The lower four bits of the last SELCAL byte selects the link format and a single byte calling frame TYPE byte completes the data portion of the CALLING block. A single byte checksum is included to confirm that the calling frame has been received error-free.

When an idle station receives a CALLING block with the local SELCAL and a correct checksum, a DATAPLEX link can start using the format specified by the calling station. After the link acknowledgment control code is received, the first data block transmitted by the master station contains the SELCALl of the calling station in a MYCALL control block. This block follows the previously described control block convention except that the MYCALL byte is followed by the master station SELCAL transmitted with two SELCAL digits per byte. After this first block is acknowledged in an FSK or DPSK DATAPLEX link, the link begins normal ISS-IRS data transfer exchanges.

Note that the sequence number is set to 0001 for the first block sent by the MASTER and the SLAVE after the link switches to OFDM.

CALLING control block

10101100	00110101	SC1 | SC2	SC3 | SC4	SC5 | SC6	SC7 | SC8	SC9 | RATE	TYPE	CKSUM

NOTE 1− SC1-SC9 are the 9 SELCAL digits, 4 bits each, [0x0 − 0x9]
RATE = link format (2 = FSK200; 3 = FSK100;
4 = DPSK600; 5 = DPSK400; 6 = DPSK200;
8 = OFDM[N = 32, M = 4])
TYPE = 8-bit value passed to the application in the link request status message
CKSUM = 00 – (sum of bytes from SC1|SC2 to TYPE)

In the following example, the master station requests a link using OFDM format RATE 8 ($N = 32$, $M = 4$), and the remote station acknowledges the link request.

ISS IRS

CALLING block (FSK100) --->

CALLING	**SELCAL**	**8**	**TYPE**	**CKSUM**

(My SELCAL received OK; link in FSK200)
<--- Start ODFM link

LINK_ACK

CALLING block (FSK100) --->

CALLING	**SELCAL**	**8**	**TYPE**	**CKSUM**

<--- Start ODFM link

LINK_ACK

ISS – OFDM (cycle change to 1 520 ms)

Carrier 1	MYCALL 0001	CRC
Carrier 2	MYCALL 0001	CRC
Carrier 3	MYCALL 0001	CRC
Carrier 4	MYCALL 0001	CRC

IRS − OFDM

Carrier 1	ACK (for block 1)
Carrier 2	ACK (for block 2)
Carrier 3	ACK (for block 3)
Carrier 4	ACK (for block 4)

The linking process starts in DATAPLEX FSK100 format and switches to OFDM after the ISS and IRS stations have correctly received the DPSK acquisition burst. The protocol cycle time switches from 1 020 ms to 1 520 ms after the ISS station receives the LINK_ACK response code from the IRS station.

The change in cycle time is a critical point in the linking protocol. Two possible error conditions can occur: first, the ISS station may not hear the IRS CS1 response code, and second, the IRS station may not hear the first ISS OFDM long burst.

There will be times that a channel supports FSK100 but not OFDM. When either the ISS or IRS station has repeated the OFDM long burst (ISS) or CS1 response (IRS) MAX_OFDM_LINK times without successfully establishing the OFDM link, both ISS and IRS must abort the link and return to STANDBY. MAX_OFDM_LINK is a programmable retry counter value.

Illustrated below is an example where the ISS station fails to decode the first CS1 response code from the IRS station. The ISS station repeats the DPSK_ACQ burst on a 1 020 ms cycle waiting for the CS1 while the IRS station is waiting for the first OFDM long burst.

ISS **IRS**

<--- OVER OK

CS0

DPSK Acquisition Burst (T = 0 ms) --->

DPSK_ACQ

<--- DPSK ACQ OK (T = 720 ms + RTT)

CS1

ISS fails to decode CS1! Repeat DPSK_ACQ

DPSK Acquisition Burst (T = 1 020 ms) --->

DPSK_ACQ

DPSK Acquisition Burst (T = 2 040 ms) --->

DPSK_ACQ

DPSK Acquisition Burst (T = 4 080 ms) --->

DPSK_ACQ

<--- DPSK ACQ OK (T = 720 ms + RTT + 4 080 ms)

CS1

ISS – OFDM (cycle change to 2 672 ms)

Carrier 1	DBlock 0001	CRC
Carrier 2	DBlock 0002	CRC
Carrier 3	DBlock 0003	CRC
Carrier 4	DBlock 0004	CRC

IRS − OFDM

Carrier 1	ACK (for block 1)
Carrier 2	ACK (for block 2)
Carrier 3	ACK (for block 3)
Carrier 4	ACK (for block 4)

In the following example, the IRS station fails to decode the first OFDM long burst from the ISS station. The ISS station starts sending OFDM long bursts, but the IRS station does not receive a good burst until after it has repeated the CS1 response code. Note that the second IRS response code is transmitted during the time that the ISS is sending the second OFDM long burst.

ISS IRS

<--- OVER OK

CS0

DPSK Acquisition Burst (*T* = 0 ms) --->

DPSK_ACQ

<--- DPSK ACQ OK (*T* = 720 ms + RTT)

CS1

ISS – OFDM (cycle change to 2 672 ms)

Send OFDM long burst (*T* = 0 ms) --->

Carrier 1	DBlock 0001	CRC
Carrier 2	DBlock 0002	CRC
Carrier 3	DBlock 0003	CRC
Carrier 4	DBlock 0004	CRC

IRS fails to decode OFDM block! Repeat CS1

Send OFDM long burst (*T* = 2 672 ms) --->

Carrier 1	DBlock 0001	CRC
Carrier 2	DBlock 0002	CRC
Carrier 3	DBlock 0003	CRC
Carrier 4	DBlock 0004	CRC

<--- DPSK ACQ OK (*T* = 720 ms + RTT + 4 080 ms)

CS1

Send OFDM long burst (*T* = 5 344 ms) --->

Carrier 1	DBlock 0001	CRC
Carrier 2	DBlock 0002	CRC
Carrier 3	DBlock 0003	CRC
Carrier 4	DBlock 0004	CRC

IRS − OFDM

<--- Send OFDM short burst (1 080 ms + RTT + 5 344 ms)

Carrier 1	ACK (for block 1)
Carrier 2	ACK (for block 2)
Carrier 3	ACK (for block 3)
Carrier 4	ACK (for block 4)

Functional description

Modulator

Shown in Fig. 1 is the modulator architecture. A number of parameters listed in Table 1 are used to define the modulator. The information bits, $x_1(n)$, of length log2(M)*L*N are first formatted into N frames, $x_2(m, n)$, as shown in Fig. 3 for $M = 4$. Each of the N parallel channels of length log2(M)*L are scrambled into $x_2(m, n)$. These scrambled frames are then mapped to L by N symbols, $x_4(m, n)$, and differentially encoded into symbols, $x_5(m, n)$. To aid in synchronization, a sequence of size S symbols is added resulting in $(L + S)$ by N symbols, $x_6(m, n)$. The $(L + S)$ by N symbols, $x_6(m, n)$, are applied to the input of the complex inverse fast Fourier transform (IFFT) resulting in the output, $x_7(m, n)$, of sample rate $fs1$. A cyclic extension of length P symbols is added resulting in $(L + S)$ by $(N + P)$ samples, $x_8(m, n)$. The samples are then converted from parallel to serial to get a complex signal, $x_9(n)$, of sample rate $fs2$ and length $(L + S)*(N + P)$. The modulated signal is interpolated by R resulting in $(L + S)*(N + \mathrm{P})*R$ samples, $x_{10}(n)$, at a sample rate of $fs3$. The upconverter converts the complex baseband modulated signal to a real passband signal, $x_{11}(n)$ for input to a digital-to-analogue (D/A) converter. Details of the individual blocks are provided below.

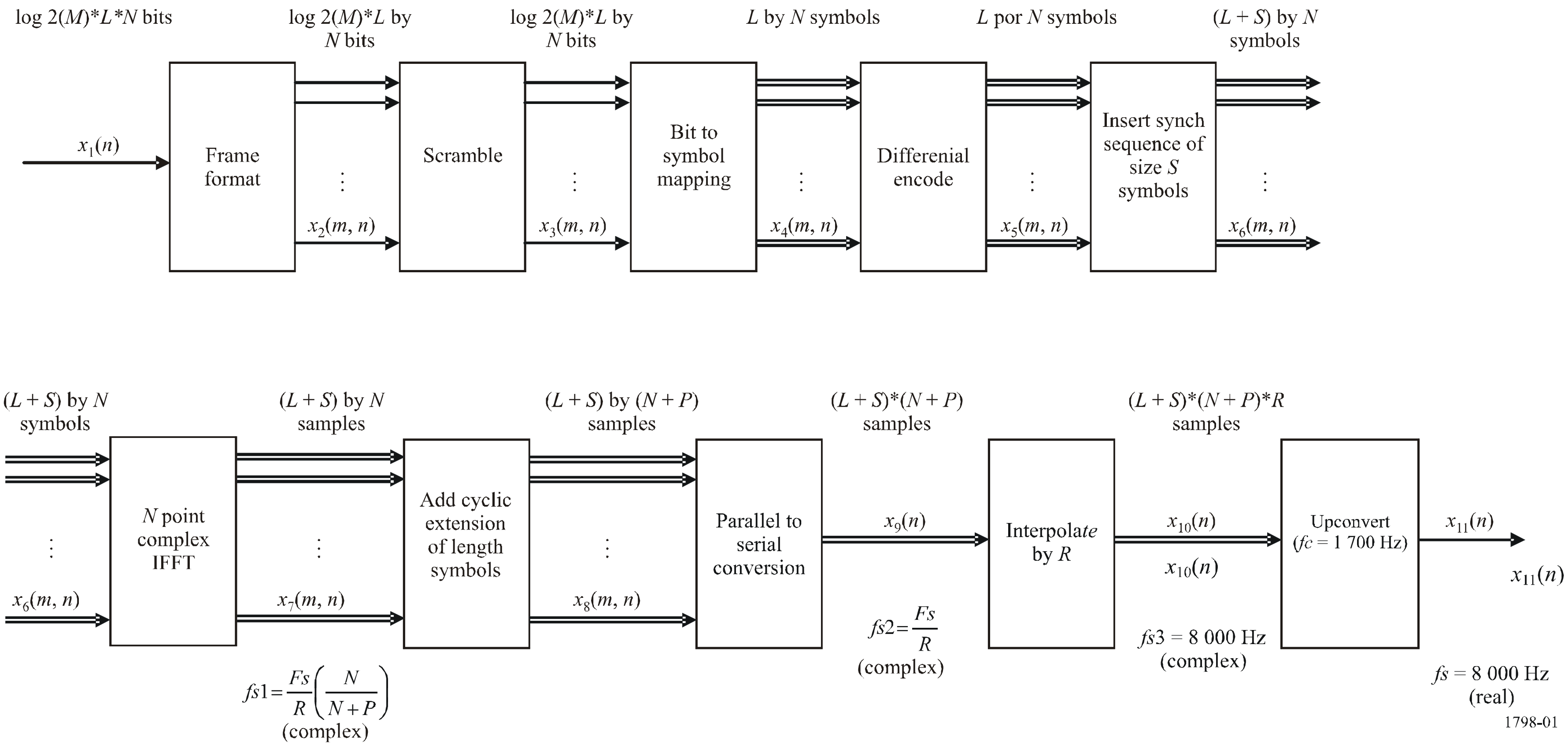
TABLE 1

Modulator parameter descriptions

Parameter	Description
N	IFFT length
P	Extension length in samples
M	Order of PSK
L	Number of parallel symbols in burst
R	Interpolate rate
S	Number of synch symbols
Fs	Sample rate (Hz)

FIGURE 1

OFDM modulator



Design parameter selection

The modulator output has an audio spectrum with a 3 dB bandwidth of 300-3 000 Hz, and a centre frequency of 1 700 Hz. The modulator parameter values are shown in Table 2 for six possible parameter combinations. The number of PSK phases, M, is either 4 or 8. The number of subcarriers (N) are configurable as $N = 16$, 32, or 64 and were selected so that the resulting subchannel bandwidth, or symbol rate, is less than 200 Hz. The audio CODEC sample rate was selected to satisfy the Nyquist criterion, and is fixed at $Fs = 8$ kHz. The interpolator rate is fixed at $R = 3$, resulting in an overall symbol rate of 8 000/3 = 2 666.66 Hz, and a signal bandwidth of about the same. The values selected for the HF modem are $N = 32$ and $M = 4$.

TABLE 2

Modulator parameter values

N	***P***	***M***	***L* long**	***L* short**	***R***	***S***	***Fs***
16	2	4	288	32	3	8	8 000
32	4	4	144	16	3	4	8 000
64	8	4	72	8	3	2	8 000
16	2	8	288	32	3	8	8 000
32	4	8	144	16	3	4	8 000
64	8	8	72	8	3	2	8 000

A frame format is defined such that 64 frames are transmitted per long burst, independent of N. For the case of $N = 32$, two frames are sent on each of $N = 32$ sub-channels. A summary of the parameters and effective throughputs is shown in Table 3.

TABLE 3

OFDM modem parameters

	$M = 4$ $N = 32$
Sample rate out (Fs) (samples/s)	8 000
IFFT size (N)	32
Extension length (P) (s)	4
Interpolate rate (R)	3
Data symbols in burst (L)	144
Sync symbols in burst (S)	4
Phases to modulate (M)	4
Sample rate out of IFFT (samples/s)	2 370.3704
Bits input	9 216
Symbols input	4 608
Symbols into IFFT	4 736

TABLE 3 (*end*)

	M = 4 **N = 32**
Sample rate with extension (samples/s)	2 666.6667
Burst length (s)	1.998
Raw throughput (bit/s)	4 612.6126
Channel symbol rate (samples/s)	83.333333
Sync symbols in short burst (S)	4
Data symbols in short burst (L)	16
Short burst length (s)	0.27
Propagation delay (s)	0.224
Spacing of bursts (s)	2.492
Bytes per frame	36
Header bytes	4
CRC bytes	4
Effective throughput (bit/s)	2 876.4045
Utilization factor	0.6235955

The value of P was chosen so that the burst length (s) is greater than the maximum HF channel delay spread. Assuming a maximum spread of 2 ms (see Recommendation ITU-R F.520), the number of required samples at Fs = 8 000 Hz is at least 16. For the case of N = 32, the extension is 1.5 ms (P = 4).

Using the modem parameter values selected, throughput analysis results are shown in Table 4. The signal generated by the OFDM modulator is passed through an HF channel using the model defined in Recommendation ITU-R F.520. All of the simulations were run using 6 400 frames, or 100 bursts.

TABLE 4

Throughput simulation results for various extension lengths

FFT size (N)	**Extension (P)**	**Phases (M)**	**Throughput *good* channel (bit/s)**	**Throughput *moderate* channel (bit/s)**	**Throughput *poor* channel (bit/s)**
32	4	4	2 088.3	1 632.2	467.7
32	8	4	1 906.6	1 547.8	1 076.5
32	16	4	1 561.9	1 481.4	519.6

The remaining modem parameters to select have to do with burst lengths, or how much information and overhead bits to use in each burst. The protocol selected for the OFDM modem is ARQ, like that used in DATAPLEX, except the number of acknowledgements per burst is multiplied 64 times. Selection of the burst length parameters, L and S, in Table 3 is determined from analysis of the ARQ performance.

The performance of an ARQ protocol can be represented by a utilization factor (η), which is the proportion of time the transmission is not idle, assuming there is always a frame to be transmitted. For the case of error-free transmission and reception the factor is:

$$\eta = \frac{T_f}{T_f + 2\tau + T_p + T_a} \tag{1}$$

where:

- T_f: frame length
- τ: one-way propagation delay
- T_p: frame processing time
- T_a: acknowledge burst length.

The maximum value of η is 1, which indicates maximum utilization. Selecting parameters that maximize η optimizes performance of an ARQ scheme.

For a channel with the probability of an unsuccessful transmission of a data or acknowledge frame given by P_f, the utilization factor is:

$$\eta = \frac{T_f}{\left(T + T_f\right)\frac{P_f}{1 - P_f} + \left(T_f + 2\tau + T_p + T_a\right)} \tag{2}$$

where T is the retransmit time. Note that for $P_f = 0$, equation (2) becomes equation (1). One method of determining the ARQ parameters is to fix T, τ, T_p, and T_a; and select the optimum T_f for a given P_f.

Assume that for $N = 64$ the short burst requires $L = 8$ symbols to transmit the acknowledgement and $S = 2$ symbols for synchronization. For $N = 32$ and $N = 16$, the parameters are selected to give the same length (ms) as for $N = 64$. This results in a short burst of length $T_a = 270$ ms. Assume a maximum one-way propagation delay of $\tau = 110$ ms, as in DATAPLEX, which allows for a one-way distance of over 20 625 miles. The frame process time, T_p, is significantly less than the other parameters and is set at a value of 100 ms for this analysis.

The overall symbol rate of $fs = 2\,666.6$ Hz with $M = 4$ and $N = 64$ results in an effective subchannel bit rate of $R_b = \log 2(M)*fs/N = 83.33$ Hz. The number of bits in a frame is:

$$N_b = R_b T_f \tag{3}$$

and the probability of frame error is:

$$P_f = P_e N_b \tag{4}$$

where P_e is the probability of a bit error. The retransmit time is:

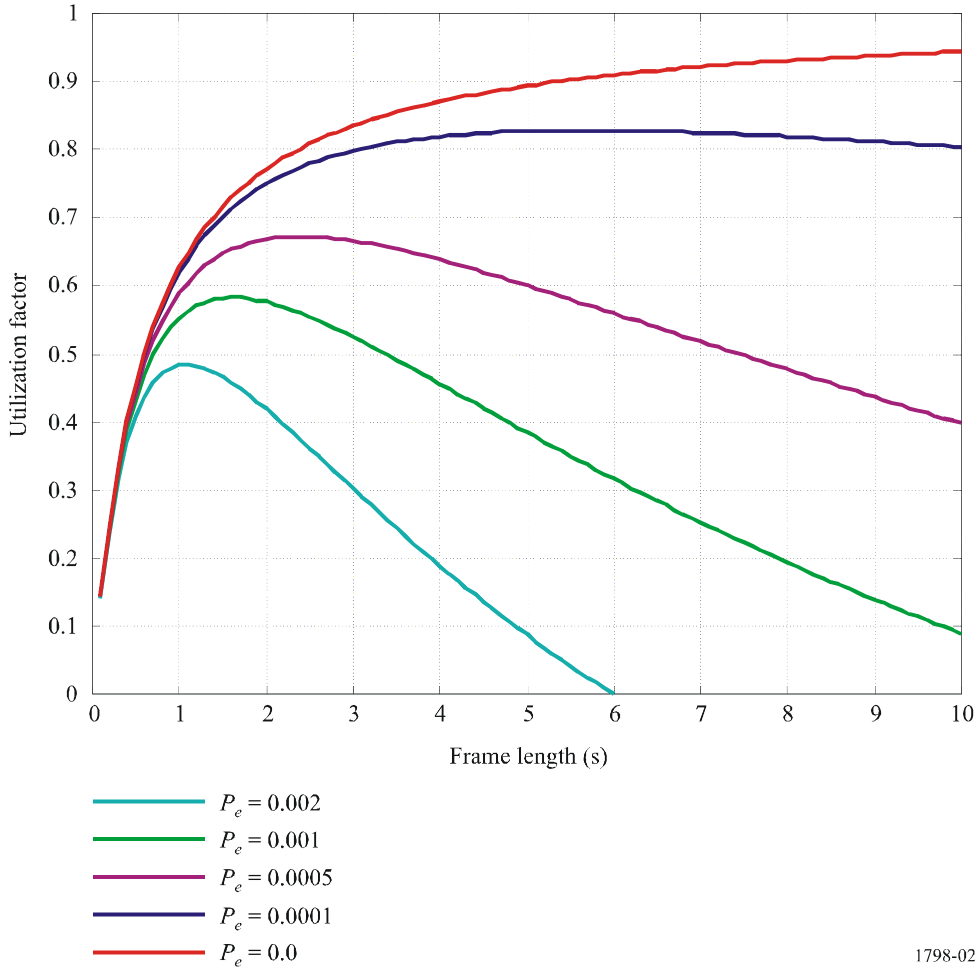
$$T = T_f + T_a + \tau \tag{5}$$

The optimization procedure involves using equation (2) and finding the maximum value of η as a function of T_f for a given P_e.

Figure 2 shows the optimization curves for bit-error probabilities of $P_e = 0.002$, 0.001, 0.0005, 0.0001, and 0.0. A first try at selection of burst size was to make the frame length nearly the same as for DATAPLEX. For the long burst, selecting L value of 144 for $N =$ 32 results in a burst length of 1.998 as shown in Table 3. For this burst size of 1.998, the resulting utilization factor is nearly optimized for a P_e of about 0.001.

FIGURE 2

OFDM ARQ utilization



Long-frame format

Each burst consists of 64 frames with each frame having a 16-bit sequence number (SEQ_NUM), information bits (INFORMATION), and a 16-bit cyclic redundancy check code (CRC). For $M = 4$ there are 14 bytes of INFORMATION for a total frame size of 18 bytes. Figure 3 shows the frame structure for $M = 4$. Input to the frame formatter is log $2(M)*L*N$ bits and output are N parallel frames of log $2(M)*L$ bits.

Cyclic redundancy check (CRC)

To verify whether the frame received has any errors, a cyclic redundancy check (CRC) is used. The CRC is the same as that used in DATAPLEX and is transmitted in each the 64 frames in the long burst. The CRC is a 16-bit standard ITU-T with generator polynomial.

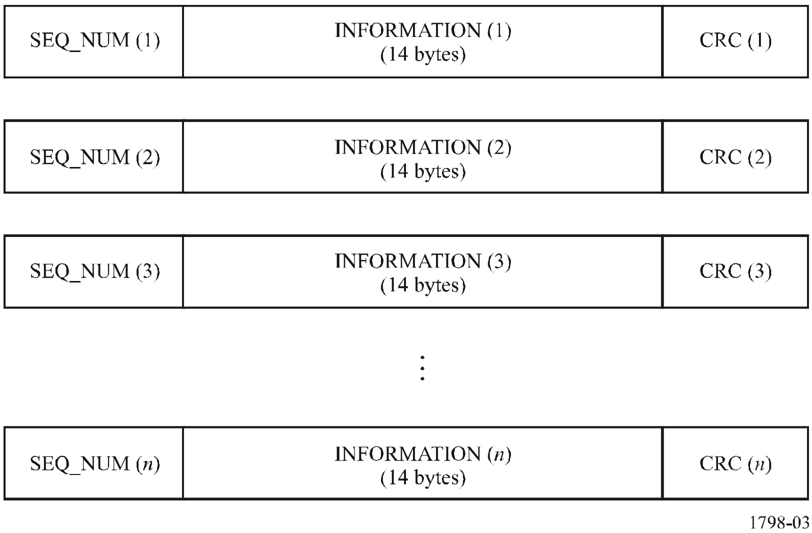
$$x^{16} + x^{12} + x^{5} + 1 \tag{6}$$

Sequence numbers

A sequence number of length 16 bits is included at the start of each of the 64 frames in a burst. They are used for signifying to the receiver the frame order for parallel-to-serial conversion. The sequence numbers also allow for the possibility of not using all 64 of the frames in a burst for transmission. The generation of the sequence is the function of the protocol layer and is outside the scope of this Recommendation.

FIGURE 3

Frame structure for $M = 4$

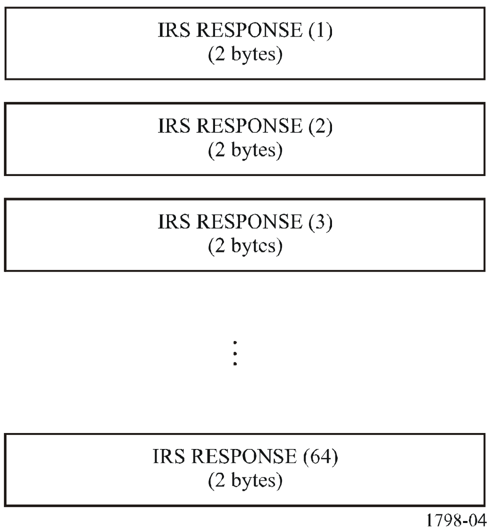


Short-frame format

The short frames are used as acknowledgments to the long frame and have the same function as information receiving station (IRS) response characters in DATAPLEX. A sequence number or CRC is not required. Shown in Fig. 4 are the frame formats for $M = 4$. In DATAPLEX the IRS RESPONSE is of length 8 bits. For the OFDM modem the IRS RESPONSE is longer and of length 16 or 24 bits, thus allowing for better cross-correlation properties of the IRS RESPONSE than DATAPLEX.

FIGURE 4

Frame structure for $M = 4$



Scrambler

Each of the 64 frames in each burst is scrambled for two beneficial effects. Scrambling produces bit patterns that have statistical properties, which make synchronization algorithms perform better. Another effect of scrambling in OFDM is the introduction of randomization of the subchannel phases. Since OFDM modulation is a sum of N individual band-limited signals, randomizing the phases reduces the peak-to-average power ratio of the modulated signal. Without scrambling, there is a greater potential for the generation of large amplitude spikes, although there is still the possibility of amplitude spikes with scrambling.

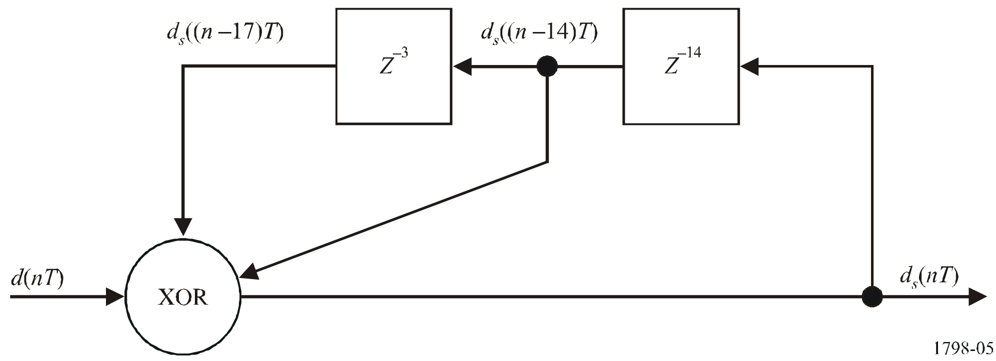
The scrambler is defined by the polynomial $1 + x^{14} + x^{17}$ or by the recursive equation:

$$d_s(nT) = d(nT) \text{ XOR } d_s((n-14)T) \text{ XOR } d_s((n-17)T \tag{7}$$

To implement the scrambler, a 17-state register is required along with an exclusive-or function as shown in Fig. 5.

FIGURE 5

Bit scrambler



To prevent the possibility of the same scramble pattern on different frames, the initial starting phase for each of the 64 frames differs by a single iteration. For the first frame, initializing the state register to 0, inputting an alternating 0/1 pattern, and iterating 18 times sets the starting phase. Scrambling for subsequent frames is done the same except the number of iterations is increased one each time. To save on processing time, the initial state registers could be saved in a table and read when initializing the scrambler for each frame.

Bit to symbol mapping

For $M = 4$ there are four possible phase values with each phase corresponding to two bits or one symbol. The bits are first mapped to symbols represented by phase values as in Table 5. Another way of representing the symbols is shown as the I and Q amplitudes of a complex signal. Note that the phases are spread at an interval of $\pi/2$ for $M = 4$. Shown in Fig. 6 is a two-dimensional representation of the mapping.

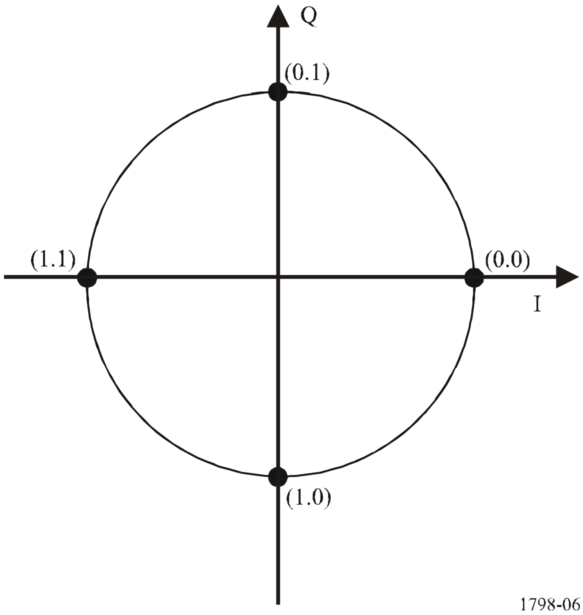
TABLE 5

Bit to symbol mapping for *M* = 4

Input bit pairs x_b		**I value**	**Q value**	**Output phase**
0	0	0	0	0
0	1	0	1	$\pi/2$
1	0	0	−1	$-\pi/2$
1	1	−1	0	π

FIGURE 6

Mapping for $M = 4$



Differential encode

The symbols out of the bit to symbol mapping are differentially encoded as the cumulative summation:

$$\psi(n) = [\psi(n-1) + \varphi(n)]_{mod\ 2\pi} \qquad (8)$$

where $\psi(n)$ is the encoded phase output and $\varphi(n)$ is the phase of the mappings in Table 5. The possible encoded phase values are $[0, \pi/2, \pi, 3\pi/2]$ for $M = 4$.

Synchronization sequence

To aid synchronization in the demodulator, S symbols are added to the start of each of the N parallel symbols prior to the IFFT. There exist methods that can synchronize to as few as two symbols, or no symbols. For larger number of synch symbols, the timing estimate is better at the expense of reduced throughput.

The methodology for synchronization is different for OFDM than of a single carrier modem. Timing information in OFDM is used to determine when to take the FFT, as opposed to when to sample the individual symbol. More about synchronization is found in the demodulator description.

The method of synchronization described in this Recommendation made use of the redundancy produced by the cyclic extension, thus removing the need for a synchronization sequence. The synchronization sequence is included for possible future use.

Inverse fast Fourier transform (IFFT)

The IFFT is the main processing function in the OFDM modulator. It combines all the individual parallel signals and makes them orthogonal. The complex IFFT is given by the equation:

$$x(n) = \frac{1}{N}\sum_{k=0}^{N-1} X(k) e^{j2\pi nk/N};\ n = 0, 1, 2, ...N-1 \qquad (9)$$

where N is the size of the IFFT, $X(k)$ are the input symbols, and $x(n)$ are the output samples. Note that the IFFT is computed in blocks of N, therefore requiring a multiple of N input length. Also, note that the output length is the same as the input and is ($L + S$) by N samples. The sample rate out the IFFT is given by:

$$fs1 = \frac{Fs}{R}\left(\frac{N}{N+P}\right) \tag{10}$$

Cyclic extension

To combat HF channel multipath effects, the IFFT output is preceded with a cyclic extension of length P consisting of the last P outputs of each IFFT implementation. This has the effect of maintaining the subcarrier orthogonal condition in the presence of multipath, thus reducing the effect of inter-subcarrier interference. The size of P is selected based on the maximum amount of delay spread in the channel. The values, selected above, are $P = 4$, and 8 for $N = 32$.

Parallel to serial conversion

After adding the cyclic prefix, the ($L + S$) by ($N + P$) samples are converted from parallel to serial resulting in $(L + S)*(N + P)$ samples at a rate of $Fs/R = 8\,000/3 = 2\,666.67$ Hz. The structure is shown in Fig. 7.

FIGURE 7

Parallel to serial conversion sample output structure

$P(1)$	$N(1)$	$P(2)$	$N(2)$	. . .	$P(L + S)$	$N(L + S)$

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Each $N + P$ block of samples can be considered a single, wideband symbol with each burst having $L + S$ samples.

Interpolator

An interpolator filter in the form of a linear-phase FIR is used to convert the sample rate from 2 666.67 Hz to 8 000 Hz. The output sample is at the desired D/A converter rate. The filter is designed using the least-squares error minimization technique with a Hamming window. The interpolation rate is $R = 3$ and the filter length is 33. The spectrum and impulse responses are shown in Fig. 8. Figure 9 shows the baseband modulator signal spectrum.

FIGURE 8

Response of the interpolator filter

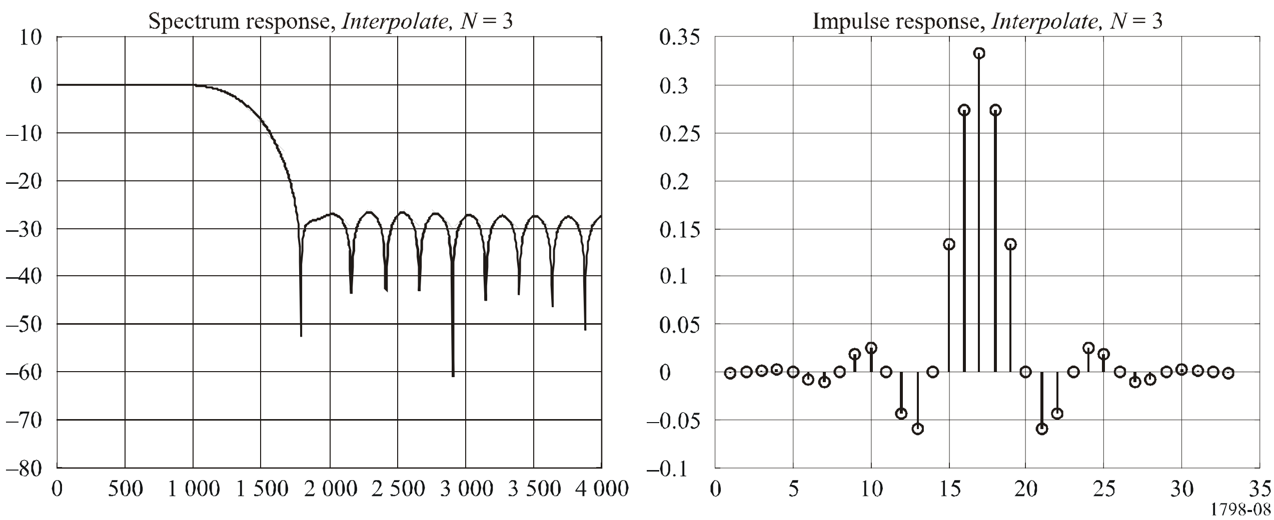
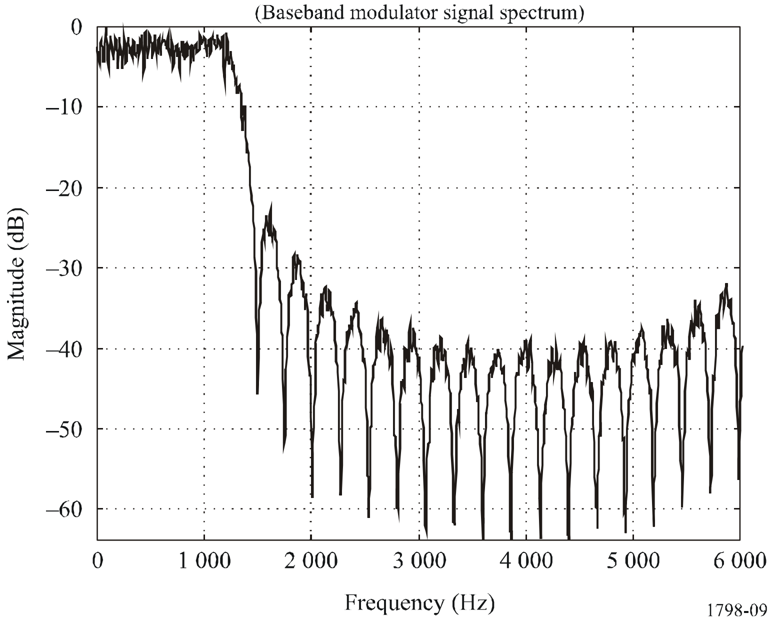


FIGURE 9

Spectrum of the baseband signal



Upconverter

The upconverter converts the baseband signal to a passband signal by mixing with sine and cosine signals at the carrier frequency f_c = 1 700 Hz and summing as shown in Fig. 10. This process also converts the signal from a complex to a real signal as is required for input to an HF radio. The final output sample rate is applied to a D/A converter prior to providing an analogue signal. Shown in Fig. 11 is the spectrum of the OFDM signal.

FIGURE 10

Upconverter

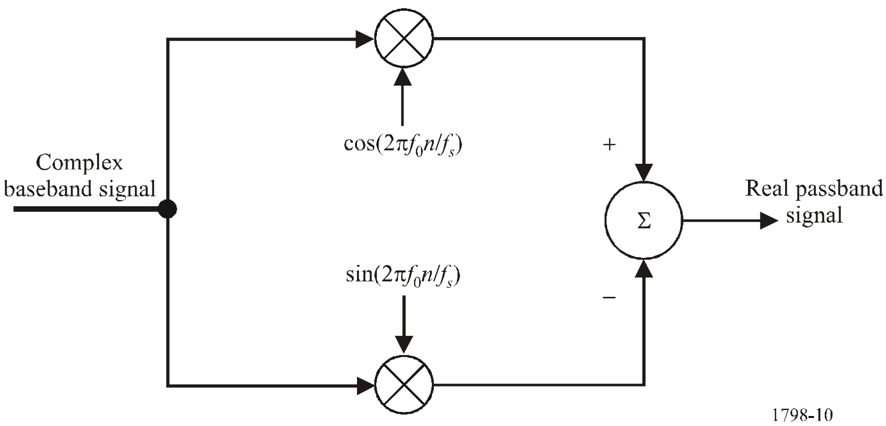
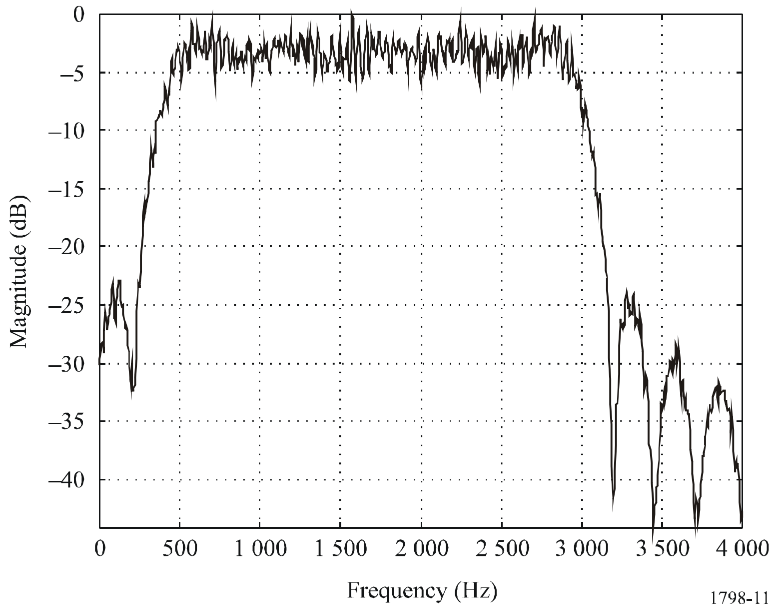


FIGURE 11

Passband modulator signal spectrum



Demodulator

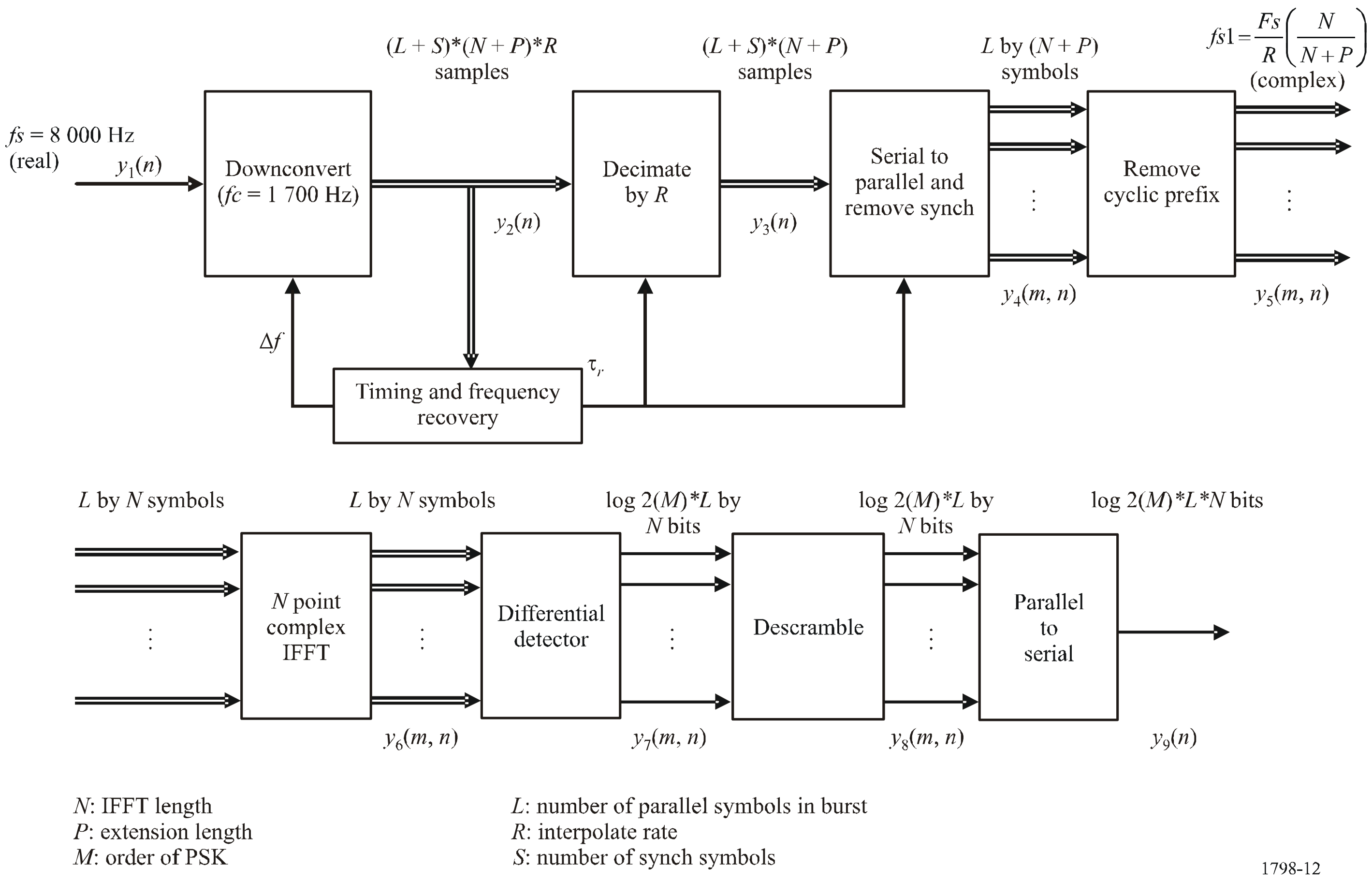
Shown in Fig. 12 is the demodulator architecture. The signal from the A/D converter, $y_1(n)$, at a sample rate of 8 000 Hz and length $(L + S)*(N + P)*R$, is down-converted from a real passband signal into a complex baseband signal, $y_2(n)$. The complex signal, $y_2(n)$, is also used for timing and frequency recovery. The frequency offset, Δf, is used in the downconverter, and the timing recovery, τ_r, is used in selecting the first symbol in the cyclic prefix. The down converter output, $y_2(n)$, is decimated by R into $(L + S)*(N + P)$ samples, $y_3(n)$. The synchronization symbols are then removed and converted from serial to parallel into L by $(N + P)$ symbols, $y_4(m, n)$. Note that at this stage in the demodulator there is one sample per symbol, so the terms "symbol" and "sample" can be interchanged. The cyclic prefix is removed resulting in L by N symbols, $y_5(m, n)$, at a sample rate of:

$$fs1 = \frac{Fs}{R}\left(\frac{N}{N+P}\right) \tag{11}$$

A complex FFT is then applied to $y_5(m, n)$ resulting in L by N symbols, $y_6(m, n)$. Then a detector recovers the symbols using a differential method, which eliminates the need for recovering the carrier phase, but still requiring recovery of the carrier frequency. Frequency is recovered for all of the subcarriers at the same time and is not required for the individual carriers. The detection is done individually on each of the N subcarriers. The symbols out of the detector are mapped into log 2$(M)*L$ by N bits, $y_7(m, n)$, using the same mapping as the modulator. The bits are descrambler using the inverse process as that used in the modulator resulting in log 2$(M)*L$ by N bits, $y_8(m, n)$. The bits are then finally converted from parallel to serial resulting in log 2$(M)*L*N$ bits, $y_9(n)$. Details of the individual blocks are provided below.

FIGURE 12

DPSK demodulator



Downconverter

The downconverter, shown in Fig. 13, performs the reverse operation of the modulator upconverter except the carrier frequency is adaptively updated based on output of the carrier frequency recovery estimators. The input is mixed with quadrature sinusoids at the recovered carrier frequency of $f_0 + \Delta f$. The carrier frequency is $f_0 = 1\,700$ Hz, the sample frequency is $f_s = 8\,000$ Hz, and the frequency offset is Δf. Shown in Fig. 14 is the resulting spectrum output. Note that there is an undesired duplication of the spectrum centred at $2{*}f_0 = 3\,400$ Hz which is removed in the next stage of processing.

FIGURE 13

Downconverter

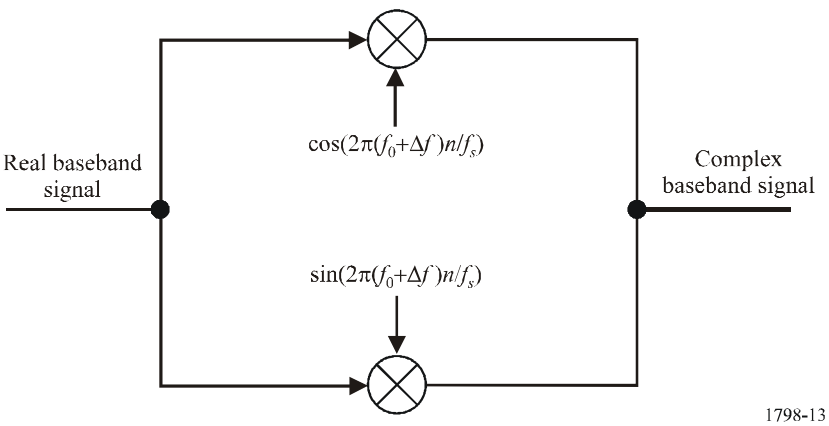
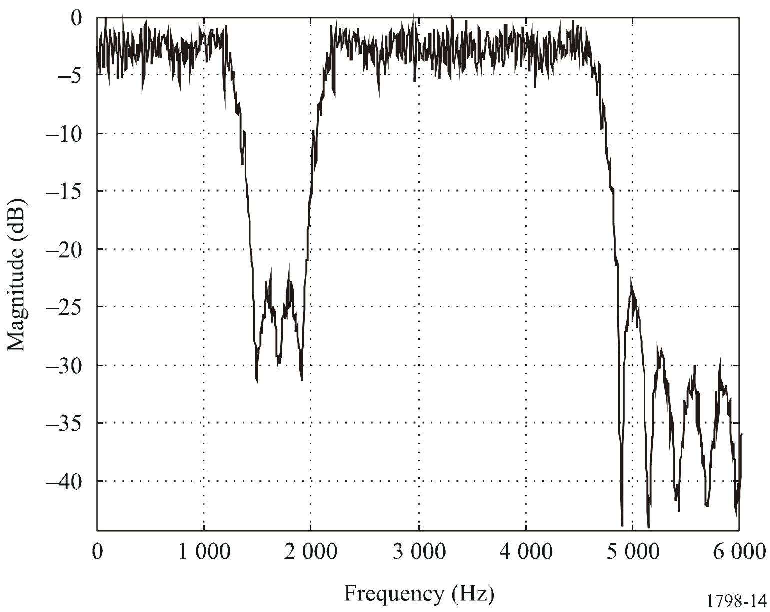


FIGURE 14

Downconverter signal spectrum

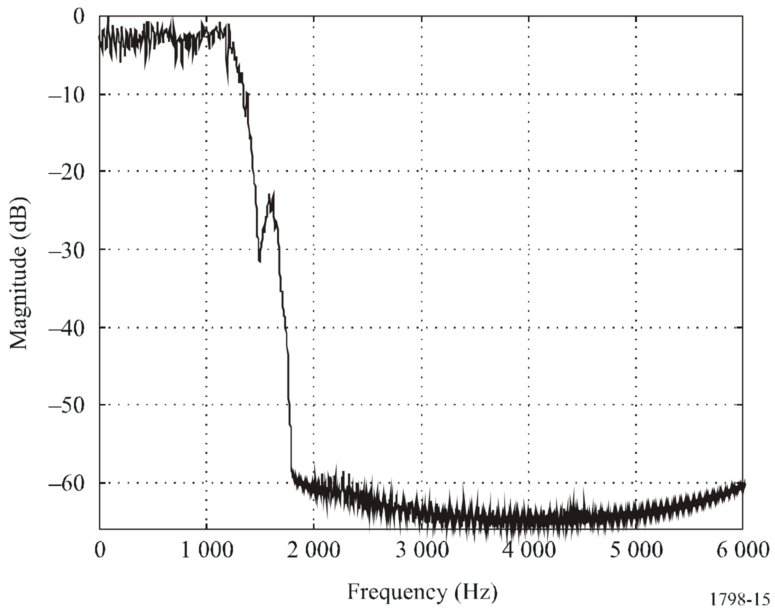


Decimation

The complex downconverter output is decimated by a factor of $R = 3$ from a sample rate of 8 000 Hz to a sample rate of 8 000/3 = 2 666.67 using the same filter as for interpolation in the modulator. Besides decimating, the band repetition centred at 3 400 Hz is filtered out, leaving the complex baseband signal. Shown in Fig. 15 is the resulting output spectrum.

FIGURE 15

Filtered downconverter signal spectrum



Timing and frequency recovery

Two uncertainties in the demodulator are the arrival time of the OFDM symbol and the carrier frequency. From Table 3 the baseband sample rate is 2 666.7 samples/s and the symbol rate is 83.33 symbols/s. This results in 16 samples/symbol. The timing recovery method uses the initial burst tone to capture the initial sample timing and samples in the middle of each symbol. The resolution is one sixteenth of a symbol and the ideal sample time in eight samples into the symbol.

OFDM is sensitive to frequency offset and the frequency recovery must be accurate to within 1 Hz. The frequency recovery algorithm is able to accurately recover frequencies with an offset up to ±50 Hz.

To accommodate off-frequency ship transmissions, the shore receivers in the network automatically track off-frequency ship transmissions, within legal limits, in order to optimize throughput. Such off-frequency operations are recorded and Customer Support is alerted in order to arrange shipboard equipment service.

Degradation due to frequency offset

The importance of frequency recovery in OFDM is illustrated by comparing the degradation due to carrier frequency offset and Weiner phase noise for multicarrier OFDM and single carrier (SC) signals. Analysis results follow for degradation in bit error rate (BER) due to carrier frequency offset and phase noise over an additive white Gaussian noise (AWGN) channel. Results for both single carrier and multicarrier signals are provided, and it is shown that the multicarrier signals are more sensitive to the each of the two degradation parameters.

$$D \approx \begin{cases} \dfrac{10}{\ln 10} \cdot \dfrac{1}{3}\left(\pi N \dfrac{\Delta F}{R}\right)^2 \dfrac{E_s}{N_0} & \text{OFDM} \\ \dfrac{10}{\ln 10} \cdot \dfrac{1}{3}\left(\pi \dfrac{\Delta F}{R}\right)^2 & \text{SC} \end{cases} \tag{12}$$

where N is the number of OFDM channels, ΔF is the frequency offset in Hz, and R is the symbol rate. Also, the S/N is given by E_s/N_0.

$$D \approx \begin{cases} \dfrac{10}{\ln 10} \cdot \dfrac{11}{60}\left(4\pi N \dfrac{\beta}{R}\right) \dfrac{E_s}{N_0} & \text{OFDM} \\ \dfrac{10}{\ln 10} \cdot \dfrac{1}{60}\left(4\pi \dfrac{\beta}{R}\right) \dfrac{E_s}{N_0} & \text{SC} \end{cases} \tag{13}$$

where β is related to the variance of the carrier phase θ by:

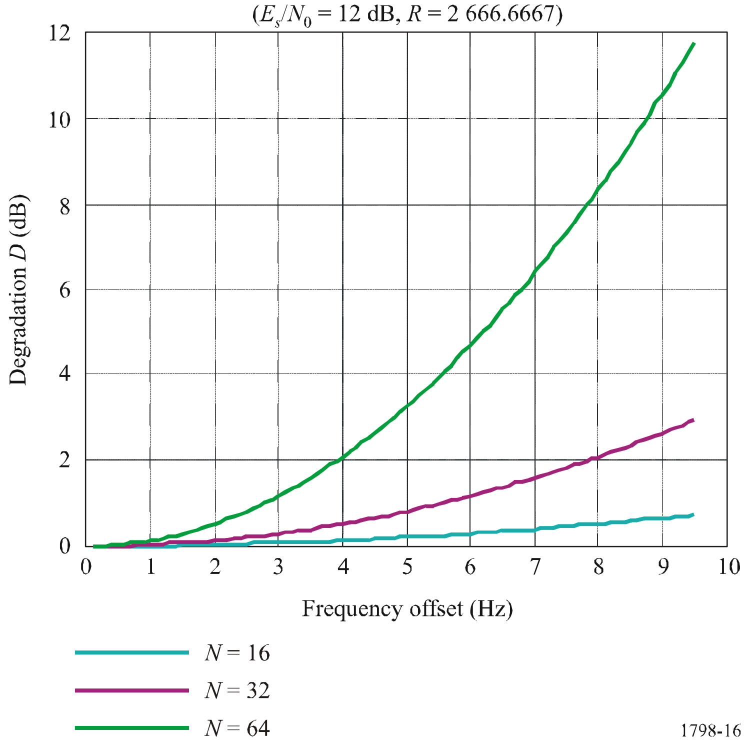
$$\sigma_\theta{}^2 = 4\pi\beta \tag{14}$$

The equations apply to M-PSK and M-QAM modulated signals. For this analysis the target BER is 10^{-3}, which for 4-DPSK modulation corresponds to an E_s/N_0 of about 12 dB. The degradation for OFDM due to frequency offset is shown in Fig. 16. Note that there is more degradation for greater values of N.

FIGURE 16

Degradation for OFDM due to frequency offset

(E_s/N_0 = 12 dB, R = 2 666.6667)



Serial to parallel conversion

Out of the decimator are $(L + S)*(N + P)$ complex samples. The synchronization symbols are removed and converted from serial to parallel resulting in L by $(N + P)$ symbols.

Cyclic prefix removal

The cyclic prefix is removed from the L by $(N + P)$ symbols resulting in L by N symbols.

Fast Fourier transform (FFT)

The FFT is the main processing function in the OFDM demodulator. The complex FFT is given by the equation:

$$X(k) = \sum_{n=0}^{N-1} x(n) \mathrm{e}^{-j2\pi nk/N} \qquad k = 0, 1, 2, \ldots N-1 \tag{15}$$

where N is the size of the FFT, $x(n)$ are the input symbols, and $X(k)$ are the output samples. Note that the FFT is computed in blocks of N, therefore requiring a multiple of N input length. Also, note that the output length is the same as the input and is L by N samples. The sample rate out of the FFT is given by:

$$fs1 = \frac{Fs}{R}\left(\frac{N}{N+P}\right) \tag{16}$$

Differential detection

Output symbols are detected from the phase differences, instead of the absolute phase of the PSK signal, thus giving it the name DPSK. Single-symbol and multiple-symbol detection are shown below.

Single symbol differential detection

The differential encoding of the symbol phase is given as:

$$\varphi_k = \varphi_{k-1} + \Delta\varphi_k \tag{17}$$

The received symbols, given by r_k, are detected using the decision rule:

Choose $\Delta\hat{\varphi}_k$ if $\mathrm{Re}\left\{r_k r_{k-1^*} \mathrm{e}^{-j\Delta\hat{\varphi}_k}\right\}$ is maximum.

For M = 4-PSK modulation the decision process consists of choosing the largest of four values.

Two symbol differential detection

Improvement in differential detection can be obtained my making a decision based on multiple symbols instead of a single one. For AWGN channels the BER approaches that of coherent detection as the number of symbols used in differential detection increases.

The decision rule for a two-symbol detector is:

choose $\Delta\hat{\varphi}_k$ and $\Delta\hat{\varphi}_{k-1}$, if $\mathrm{Re}\left\{r_k r_{k-1^*} \mathrm{e}^{-j\Delta\hat{\varphi}_k} + r_{k-1} r_{k-2^*} \mathrm{e}^{-j\Delta\hat{\varphi}_{k-1}} + r_k r_{k-2^*} \mathrm{e}^{-j(\Delta\hat{\varphi}_k + \Delta\hat{\varphi}_k - 1)}\right\}$ is maximum.

For the case of M = 4-PSK the decision is picking the largest of M^2 = 16 values.

Descrambler

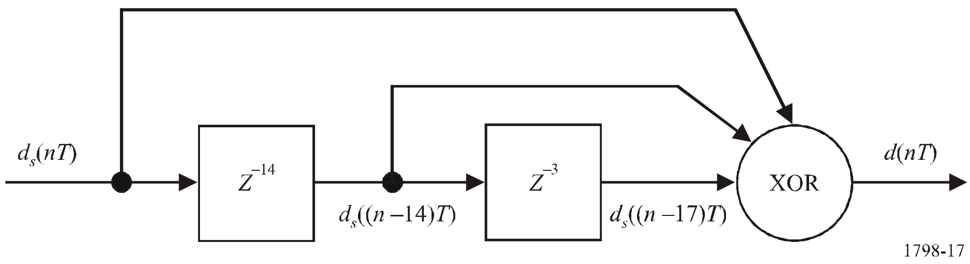
The descrambler is the inverse of the scrambler and is defined by the recursive equation:

$$d(nT) = d_s(nT) \text{ XOR } d_s((n-14)T) \text{ XOR } d_s((n-17)T) \tag{18}$$

To implement the descrambler, a 17-state register is required along with an exclusive or function as shown in Fig. 17.

FIGURE 17

Bit descrambler



The initial phases of the descrambler are set the same as in the scrambler, still using the scrambler implementation.

Parallel to serial conversion

The log2(M)*L by N parallel bits out of the descrambler are converted to log2(M)*L*N serial bits. It is possible to implement the CRC decoder before it is converted from parallel to serial, since the CRC decode is performed on each of the 64 parallel frames in the burst, but it is best performed as part of the protocol layer.

Cyclic redundancy check (CRC) decoder

The CRC decoder is the inverse of the CRC encoder with the generator polynomial:

$$x^{16} + x^{12} + x^5 + 1 \tag{19}$$

If the CRC check fails, the frame is rejected and a request for retransmission is generated.

Frequency selection

In a global communications network with several hundreds of channels, two dozen-plus stations and many thousands of ships moving a high volume of data, it is extremely important to have an efficient frequency selection system. The ALE Mil standard in common use would be totally inadequate and inappropriate in this situation and extremely spectrum-inefficient.

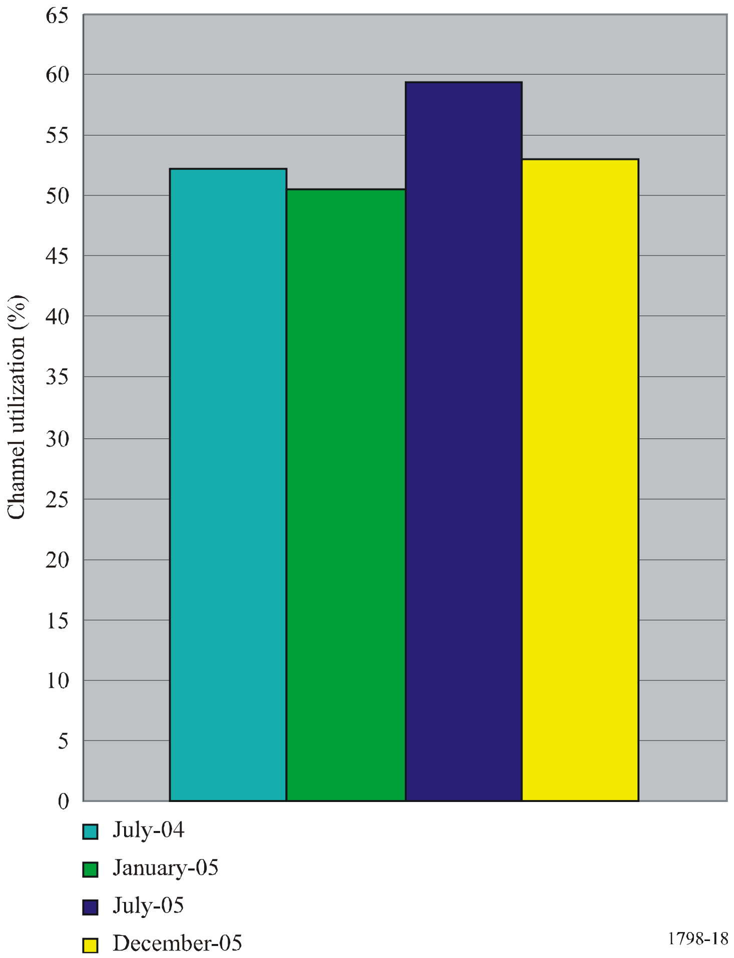
Consequently, one method uses a propagation analysis tool on board the ship that selects frequencies for scanning. Selection decisions are based on dynamically updated current conditions for that date, time and geographic position. This means that there is no waste of spectrum by sounding or attempting links on poor quality channels. The ship looks directly to propagating channels and scans for one that is available (not busy). Current propagation parameters are delivered to ships via the channel "free signals".

Frequency usage

Ships will typically use a channel for anything from less than a minute up to 30 min. Communications vary from brief data bursts for tracking purposes to large files. The combination of large files and the large number of mobiles means that frequencies are occupied almost continuously. This results in the need for exclusive allocations with no possibility for sharing with other users or services. A recent usage record from one shore node is attached (see Fig. 18). If the available time in this chart was reduced by the daily time period when each frequency was not propagating, it can be seen that the occupancy would be close to 100%.

FIGURE 18

Per cent of channel utilization



4 System 2 – Electronic mail system using Pactor-III protocol, including the system used by the Global Link Network (GLN)

Emission type

The system uses ITU emission type 2K20J2D.

Bandwidth

The bandwidth needed is two times 3 kHz (one duplex voice channel).

Communication system components

The system has the following components:

Transmission protocol

The system uses the efficient and well-proven PACTOR-III HF transmission protocol. Maximum net throughput with online data compression is approximately 5 200 bit/s. A description of the protocol is provided in § 4.1.

T-BUS communication protocol

The system uses the T-BUS communication protocol in order to control standard GMDSS HF/MF radio equipment. T-BUS is used by maritime radio manufacturers Skanti and Sailor (and others) in their GMDSS radio equipment. There are several versions of the T-BUS protocol, a description of the Skanti communication protocol is provided in § 4.2.

Modem

It is possible to use different types of modems as long as they can handle RS-232 communications with T-BUS protocol. The Norwegian system uses PTC-II modems.

Replacement for narrow-band direct printing (NBDP)

The HF Mail system is presently able to replace NBDP for general communications, probably also for safety and distress communications in the future.

4.1 The PACTOR-III protocol (Technical Description by Hans-Peter Helfert and Thomas Rink, SCS GmbH & Co. KG, Hanau, Germany)

4.1.1 Introduction

Similar to PACTOR-I and PACTOR-II, PACTOR-III is a half-duplex synchronous ARQ system. In the standard mode, the initial link setup is performed using the FSK (PACTOR-I) protocol, in order to achieve compatibility to the previous systems. If both stations are capable of PACTOR-III, automatic switching to this highest protocol level is performed.

While PACTOR-I and PACTOR-II were developed for operation within a bandwidth of 500 Hz, PACTOR-III is designed specifically for the commercial market to provide higher throughput and improved robustness utilizing a complete SSB channel. A maximum of 18 tones spaced at 120 Hz is used in optimum propagation conditions. The highest raw bit rate transferred on the physical protocol layer is 3 600 bit/s, corresponding to a net user data rate of 2 722.1 bit/s without data compression. As different kinds of online data compression are provided, the effective maximum throughput depends on the transferred information, but typically exceeds 5 000 bit/s, which is more than 4 times faster than PACTOR-II. At low SNR, PACTOR-III achieves a higher robustness compared to PACTOR-II.

The ITU emission designator for PACTOR-III is 2K20J2D.

4.1.2 Speed levels (SLs) and bandwidth

Depending on the propagation conditions, PACTOR-III utilizes 6 different SLs, which can be considered as independent sub-protocols with distinct modulation and channel coding. The symbol rate is 100 bauds on all SLs. Up to 18 tones are used, spaced at 120 Hz. The maximum occupied bandwidth is 2.2 kHz (from 400 to 2 600 Hz). The centre frequency of the entire signal is 1 500 Hz. The tone representing the "lowest" channel is sent at a frequency of 480 Hz, the highest tone is 2 520 Hz. As tones are skipped on the two lowest SLs, the gaps between them increase to N times 120 Hz in these cases. Figure 19 illustrates the number and position of the used channels at the different SLs.

Similar to the PACTOR-II protocol, the digital data stream that constitutes a specific virtual carrier is swapped to a different tone with every ARQ cycle in order to increase the diversity gain by adding additional frequency diversity. Considering that in the normal state the numbers of the virtual data carriers correspond with the numbers of the respective tones, the swapped mode assigns carrier 0 with tone 17, 1 with 16, 2 with 9, 3 with 10, 4 with 11, 5 with 12, 6 with 13, 7 with 14 and 8 with 15. Tones 5 and 12 can be considered as equivalent to the two carriers of PACTOR-II, as they transfer the variable packet headers and the control signals (see below).

FIGURE 19

Number and position of the used channels at the different SLs

CN	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17
SL																		
1						x							x					
2				x		x		x			x		x		x			
3			x	x	x	x	x	x	x	x	x	x	x	x	x	x		
4			x	x	x	x	x	x	x	x	x	x	x	x	x	x		
5		x	x	x	x	x	x	x	x	x	x	x	x	x	x	x	x	
6	x	x	x	x	x	x	x	x	x	x	x	x	x	x	x	x	x	x
TF	480	600	720	840	960	1 080	1 200	1 320	1 440	1 560	1 680	1 800	1 920	2 040	2 160	2 280	2 400	2 520

CN: channel number
TF: tone frequency (Hz)
x: indicates that the tone is used in the perspective SL

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4.1.3 Modulation, coding, and data rates

As modulation, either differential binary PSK (DBPSK) or differential quadrature PSK (DQPSK) is applied. After full-frame bit-interleaving of the entire data packet, an optimum rate 1/2 convolutional code with a constraint length (CL) of 7 or 9 is used. Similar to the PACTOR-II protocol, the codes with higher rates, i.e. rate 3/4 and rate 8/9, are derived from that code by so-called puncturing: Prior to the transmission, certain bits of the rate 1/2 encoded bit stream are "punctured", i.e. deleted and thus not transmitted. At the receiving side, the punctured bits are replaced with "null" bits prior to decoding with the rate 1/2 decoder. The decoder treats these null bits neither as a "1" nor as "0", but as an exactly intermediate value. Thus, these bits have no influence on the decoding process. The coding gain of a "punctured" code nearly matches the coding gain of the best known specific rate 3/4 or 8/9 codes with a comparable constraint length, provided that the puncture pattern is chosen carefully. The major advantage of this approach is that a single code rate decoder (in our case a rate 1/2 decoder) can implement a wide range of codes. Therefore, punctured codes are used in many modern communication systems. In the SCS modems, a Viterbi decoder with soft decision is used for all speed levels, yielding a maximum of coding gain.

Figure 20 shows the modulation, the CL and the code rate (CR) of the applied convolutional code, the physical data rate (PDR), i.e. the raw bit rate transferred on the physical protocol layer, the net data rate (NDR), i.e. the uncompressed user data rate, as well as the crest factor (CF) of the signal for the different SLs.

The following two figures show the BERs for the different speed levels. In Fig. 21, the rates are referenced to the normalized energy per bit (E_b/N_0). Due to the different number of tones (2-18) and the different modulations (DBPSK/DQPSK), this figure does not reveal the performance with respect to the channel S/N. Thus, in Fig. 22, the rates are referenced to the channel S/N at a channel bandwidth (BW) of 3 kHz. The different speed levels cover a wide S/N range. For maximum throughput with SL6, a channel S/N of 14 dB is required.

FIGURE 20

Parameters of the different SLs

SL	Modulation	CL	CR	PDR	NDR	CF (dB)
1	DBPSK	9	1/2	200	76.8	1.9
2	DBPSK	7	1/2	600	247.5	2.6
3	DBPSK	7	1/2	1 400	588.8	3.1
4	DQPSK	7	1/2	2 800	1 186.1	3.8
5	DQPSK	7	3/4	3 200	2 039.5	5.2
6	DQPSK	7	8/9	3 600	2 722.1	5.7

1798-20

It should be noted, that the performance in terms of throughput (bit/s) depends on the implementation of the ARQ protocol and cannot be deduced from the physical data rates and the BERs. Performance measurements will be presented below.

FIGURE 21

BER for the different SLs with respect to the energy per bit

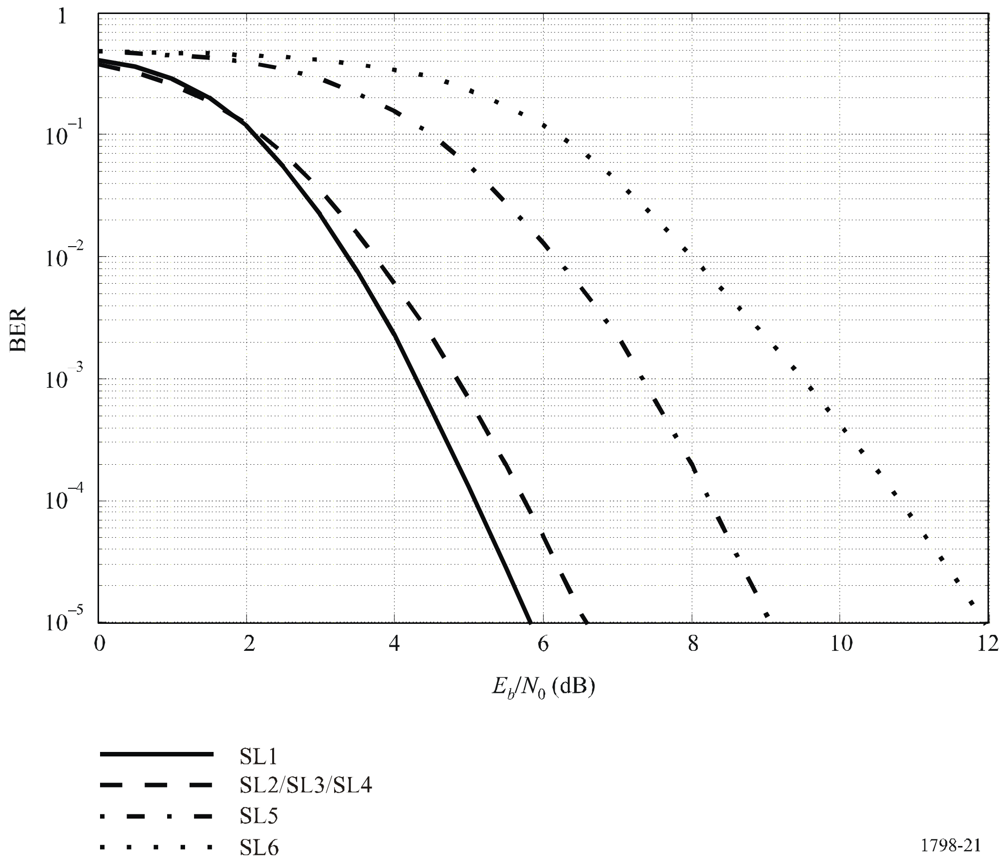
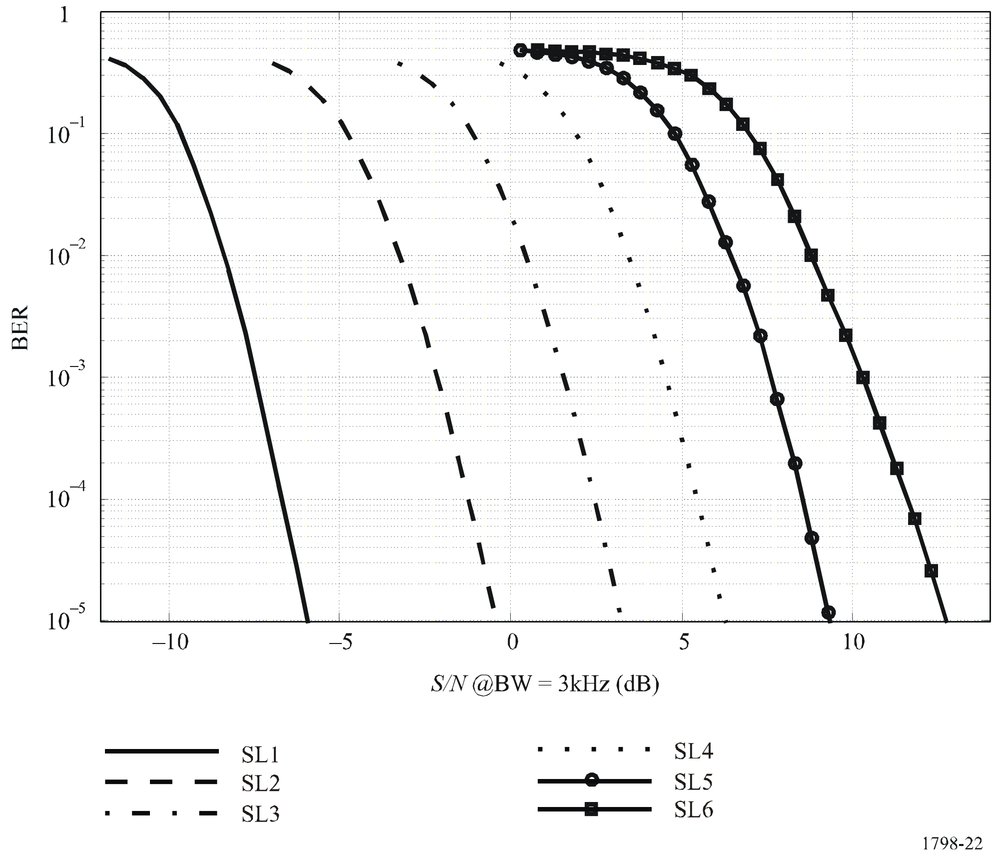


FIGURE 22

BER rate for the different SLs with respect to the channel *S*/*N*



4.1.4 Crest factor (CF) and transmitter output power

One of the most important characteristics of the PACTOR-III signal is the low CF, especially with the lower SLs. As most HF power amplifiers are peak-power limited and use a peak-power automatic level control (ALC), PACTOR-III provides considerably more transmitter output power than comparable multicarrier modes like, for example, OFDM modes when using the same power amplifier, thereby increasing the *S*/*N* at the receiver. Up to SL4, the CF fairly compares to the CF of single-carrier modes. Even with SL5 and SL6, the CF is about 3 dB lower than the CF of typical OFDM modes, thereby doubling the transmitted RMS power. In the context of Digital Radio Mondiale (DRM), it has been found that single-carrier modes perform much better than OFDM modes if the coding is weak (rate > 2/3); OFDM modes without coding are well known to be a disaster when used over highly frequency selective channels. With strong coding (rate <= 1/2), OFDM modes perform slightly better than single-carrier modes. These results are based on two assumptions:

a) the transmitted RMS power is the same for both modes, meaning that the peak power of the OFDM mode is several dBs higher than that of the single-carrier mode;

b) an optimum DFE equalizer is used with the single-carrier mode (an optimum MLSE equalizer cannot be used because the channel impulse response is too long).

If the peak power is held constant, the single-carrier mode performs better for all reasonable coding rates, but the required optimum DFE equalizer presents an inevitable obstacle. PACTOR-III is designed to provide the benefits of both modes by minimizing the CF and avoiding the use of an equalizer.

SCS modems operate with constant peak power at all speed levels to optimally exploit the available output power of peak-power limited HF power amplifiers. Thus, the RMS output power changes when switching through the speed levels, due to the different CFs. The channel *S*/*N* at the receiver changes accordingly. This has to be kept in mind when interpreting the BERs in Fig. 22.

4.1.5 Cycle duration

In the standard mode, the ARQ cycle durations are 1.25 s (short cycles) and 3.75 s (data mode), which is one of the requirements to obtain easy compatibility to the previous PACTOR standards. In this mode, due to signal propagation and equipment switching delays, PACTOR-III is capable to establish ARQ links over a maximum distance of around 20 000 km. To further extend the maximum distance, a Long Path Mode is available, enabling ARQ links up to a maximum distance of 40 000 km, with cycle times of 1.4 s (short cycles) and 4.2 s (data mode), respectively. The calling station initiates a link in Long Path Mode by inverting the first byte of the call sign in the FSK connection frame (for details, see the PACTOR-I protocol description).

4.1.6 Structure of packets and control signals

Except from different data field lengths, the basic PACTOR-III packet structure is similar to the previous PACTOR modes. It consists of a packet header, a variable data field, a status byte and a CRC. Two types of headers are used: Sixteen variable packet headers consisting of 8 symbols each are sent alternately on tones 5 and 12 to code 4 bits of information: bit 0 defines the request-status indicating a repeated packet. Bits 2 and 3 specify the speed levels 1 to 4 according to a modulo-4 logic, whereas the detection of levels 5 and 6 is performed by additionally analyzing the constant packet headers. Bit 4 gives the current cycle duration: “0” specifies short and “1” data cycles. Figure 23 shows the hexadecimal codes of the variable packet headers.

FIGURE 23

Definitions of the variable packet headers (initiating tones 5 and 12)

VH0	0x1873174f	VH1	0xfc0f6047	VH2	0x0a4c7ea7	VH3	0x09bce11f
VH4	0x8e67c43c	VH5	0x7268a47b	VH6	0x842bba9b	VH7	0x87db2523
VH8	0x4d55aa6a	VH9	0xb15aca2d	VH10	0x4719d4cd	VH11	0x44e94b75
VH12	0x3ccd91a9	VH13	0xc0c2f1ee	VH14	0x3681ef0e	VH15	0x357170b6

1798-23

The remaining tones 1-4, 6-11 and 13-18 are preceded by constant headers that characterize the respective tones without transferring any additional information. They support frequency tracking, memory-ARQ, the listen-mode and the detection of the speed levels 5 and 6. Figure 24 presents the hexadecimal codes of the constant packet headers.

FIGURE 24

Definitions of the constant packet headers (initiating tones 1-4, 6-11, 13-18)

CH0	0xc324	CH1	0xf987	CH2	0xb1c8	CH3	0xf370
CH4	0x801d	CH5	0x7c3d	CH6	0xd8f1	CH7	0x5a3c
CH8	0x792d	CH9	0x8397	CH10	0x33aa	CH11	0x5a3c
CH12	0x823c	CH13	0x073f	CH14	0xf798	CH15	0xd801

1798-24

The headers are followed by the data fields that transfer the user information. On the 6 different speed levels, 5, 23, 59, 122, 212, and 284 payload bytes are transferred in the short cycle and 36, 116, 276, 556, 956, and 1 276 payload bytes in the long cycle, respectively. After de-interleaving and decoding of the entire data transferred on all tones within a certain cycle, the actual information packet is obtained, which consists of the user data, a status byte and 2 CRC bytes. The status byte characterizes the packet by a two-bit packet counter to detect repetitions (bit 0 and 1), provides information on the applied data compression (bits 2, 3 and 4), suggests to switch to the data mode when the amount of characters in the transmit buffer exceeds a certain number (bit 5), indicates a changeover request (bit 6) and initiates the link termination protocol (bit 7). For details, see the graphic below. The final part of the packet is a 16-bit CRC calculated according to the CCITT-CRC16 standard.

PACTOR-III uses the same set of six 20-bit control signals (CS) as PACTOR-II. They are transmitted simultaneously on the tones 5 and 12 and all have the maximum possible mutual hamming distance to each other. Hence they reach exactly the Plotkin boundary and represent a perfect code. This allows the use of the cross correlation method for CS detection, a kind of soft decision that leads to the correct detection of even inaudible CS, due to the high correlation gain. CS1 and CS2 are used to acknowledge/request packets and CS3 forces a break-in. CS4 and CS5 handle the speed changes: CS4 demands an increase of the speed to the next higher level. CS5 acts as a NAK asking for a repetition of the previously sent packet and at the same time for a reduction of the speed to the next lower level. CS6 is a toggle for the packet length and inquires a change to long cycles in case that the actual state is short cycles and vice versa. All CS are always sent in DBPSK in order to obtain maximum robustness.

Figure 25 illustrates the PACTOR-III ARQ operation.

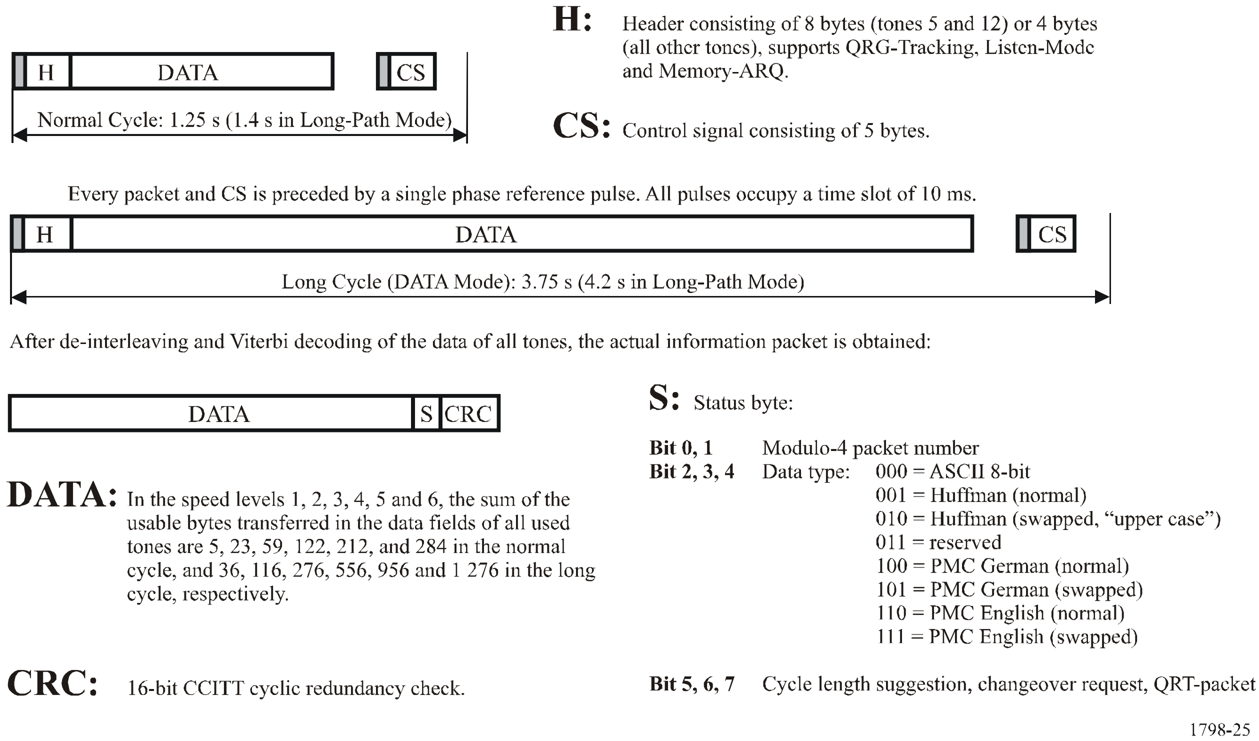
4.1.7 Online data compression

Like in the previous PACTOR modes, automatic online data compression is also applied in the PACTOR-III protocol, comprising Huffman and run-length encoding as well as Pseudo-Markov Compression (PMC, see below). The information sending system (ISS) automatically checks, whether one of these compression modes or the original ASCII code leads to the shortest data package, which depends on the probability of occurrence of the characters. Hence, there is no risk of losing throughput capacity. Of course, PACTOR-III is still capable to transfer any given binary information, e.g. programs or picture and voice files. In case of a binary data transfer, the online data compression normally switches off automatically due to the character distribution. An external data compression in the terminal program is usually performed instead.

Huffman compression exploits the “one-dimensional” probability distribution of the characters in plain texts. The more frequently a character occurs, the shorter its Huffman symbol has to be. More details including the code table used in the PACTOR protocols can be found in the description of the PACTOR-I standard.

FIGURE 25

PACTOR-III ARQ operation



Markov compression can be considered as a double Huffman compression, since it not only makes use of the simple probability distribution, but of the two-dimensional probability. For each preceding character, a probability distribution of the very next character can be calculated. For example, if the actual character is "e", it is very likely that "i" or "s" occurs next, but extremely unlikely that an "X" follows. The resulting probability distributions are much more concentrated than the simple one-dimensional distribution and thus lead to a considerably better compression. Unfortunately, there are two drawbacks: Since for each ASCII character a separate coding table is required, the entire Markov coding table becomes impractically large. Additionally, the two-dimensional distribution and thus the achievable compression factor depends much more on the kind of text than the simple character distribution. We have therefore chosen a slightly modified approach which we called Pseudo-Markov Compression (PMC), because it can be considered as a hybrid between Markov and Huffman encoding. In PMC, the Markov encoding is limited to the 16 most frequent "preceding" characters. All other characters trigger normal Huffman compression of the very next character. This reduces the Markov coding table to a reasonable size and also makes the character probabilities less critical, since especially the less frequent characters tend to have unstable probability distributions. Nevertheless, for optimum compression, two different tables for English and German texts are defined in the PACTOR-II and PACTOR-III protocols and automatically chosen. When transferring plain text, PMC yields a compression factor of around 1.9 compared to 8-bit ASCII.

Run-length encoding allows the effective compression of longer sequences of identical bytes. The special prefix byte "0x1D" is defined, which initiates a 3-byte run length code. The second byte is called the "code byte" and contains the original code of the transferred byte within the range of the entire ASCII character set. The third byte provides the number of code bytes to be displayed on the receiving side within the range between "0x01" to "0x60". Values between "0x00" and "0x1f" are transferred as "0x60" to "0x7f", values between "0x20" and "0x60" are transferred without any change. For example, the sequence "AAAAAAAA" is transferred using the 3-byte run-length code "0x1D 0x41 0x68".

4.1.8 Signal characteristics and practical considerations

As the FSK PACTOR standard is used for the initial link establishment, frequency deviations of the connecting stations of up to ±80 Hz are still tolerated. Similar to the PACTOR-II mode, a powerful tracking algorithm is provided in the SCS modems to compensate any divergence and exactly match the signals when switching to the DPSK mode, which requires a high frequency accuracy and stability.

The PACTOR-III signal provides a very high spectral steepness in order to avoid any spillover in adjacent channels. Therefore, low quality audio filters may cause distortion of the side tones of the higher speed levels, both on the transmitting and on the receiving side. To partly compensate for that, SCS modems allow the amplitude of the signal edges to be enhanced individually in two steps using the "Equalize" command, which defines the function of the PACTOR-III transmit equalizer. A value of "0" switches this function off, "1" means a moderate, and "2" a strong enhancement of the side tones of the signal.

Further, it has to be taken into consideration, that, due to the different possible "tones" settings related to the FSK mode used for the initial link setup, a shift of the center frequency of the signal may occur with the automatic switching to PACTOR-III. Therefore, the "tones" settings should be checked carefully and adapted to the other stations in the network in order to make sure that no offset occurs between the linked stations and the PACTOR-III signal is placed symmetrically within the filter bandwidth. Usually, identical "tones" settings on both sides of a PACTOR-III link are required for proper operation. SCS recommends to set "tones" to "4", defining the FSK connection tones as 1 400 and 1 600 Hz, which are balanced around the PACTOR-III center frequency of 1 500 Hz, to avoid incompatibilities between PACTOR-III users.

Figure 26 shows the spectrum of a PACTOR-III signal at speed level 6 with all 18 tones active.

4.1.9 Performance measurements

The performance of ARQ modes with different speed levels critically depends on the implementation of the ARQ protocol and the automatic selection of an appropriate speed level for the given channel conditions. PACTOR-III comprises memory-ARQ to smooth the transitions between speed levels and to improve the throughput at low S/N ratios. In memory-ARQ, the combination of re-transmitted data packets allows for safe data transmission over extremely bad channels even if each received packet is corrupted. Figure 27 presents the results of throughput measurements over an additive white Gaussian noise channel (AWGN) and poor channel. The S/N is evaluated with respect to the RMS output power at SL1 to correct for the different CFs. Due to the bit error rates presented in Fig. 22, the maximum throughput of 2 720 bit/s should be achieved with SL6 at a channel S/N of more than 14 dB with respect to the RMS output power at SL6. According to Fig. 20, the CFs of SL1 and SL6 differ by 3.8 dB. Therefore, the maximum throughput should be achieved at a channel S/N of more than 18 dB with respect to the output power at SL1 which fairly agrees with the measured AWGN throughput in Fig. 27.

FIGURE 26

Spectrum of a PACTOR-III signal at SL6 with all 18 tones active

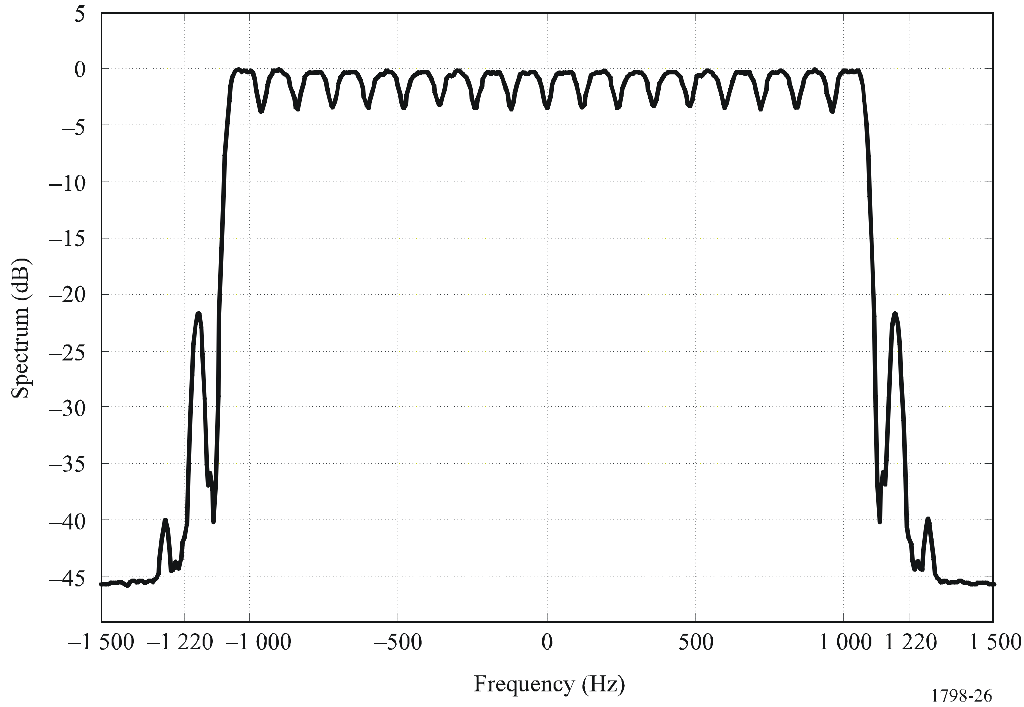


FIGURE 27

Throughput of PACTOR-III

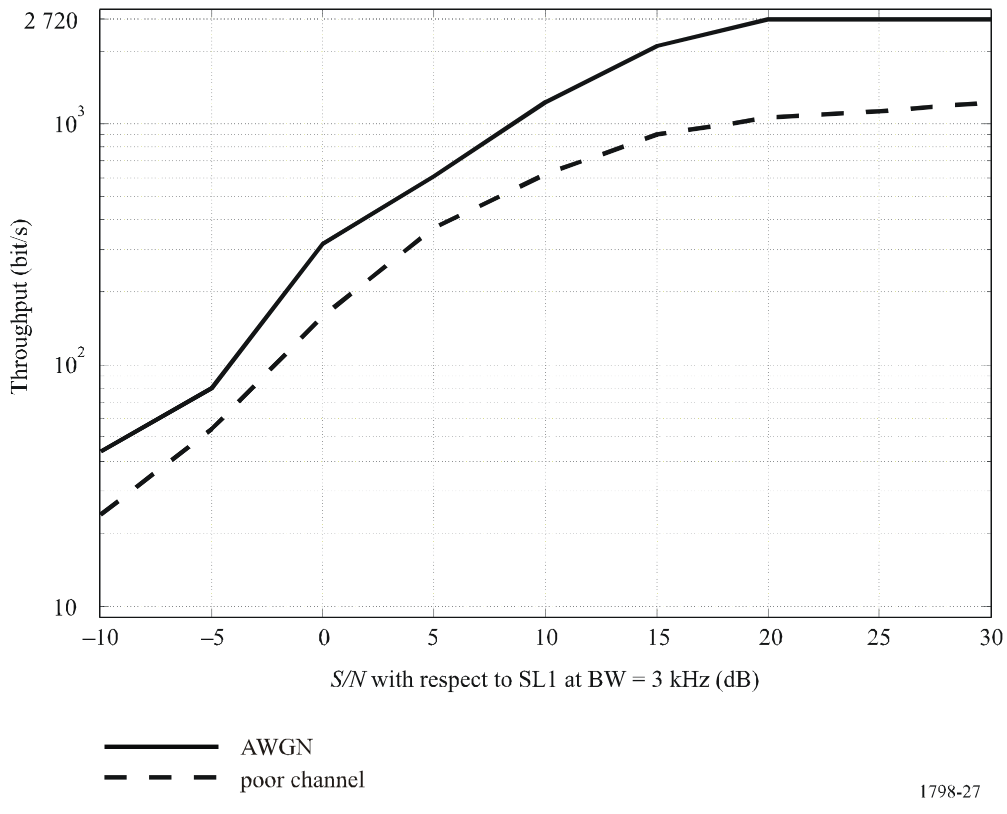


FIGURE 28

BER for the different SLs with respect to the energy per bit

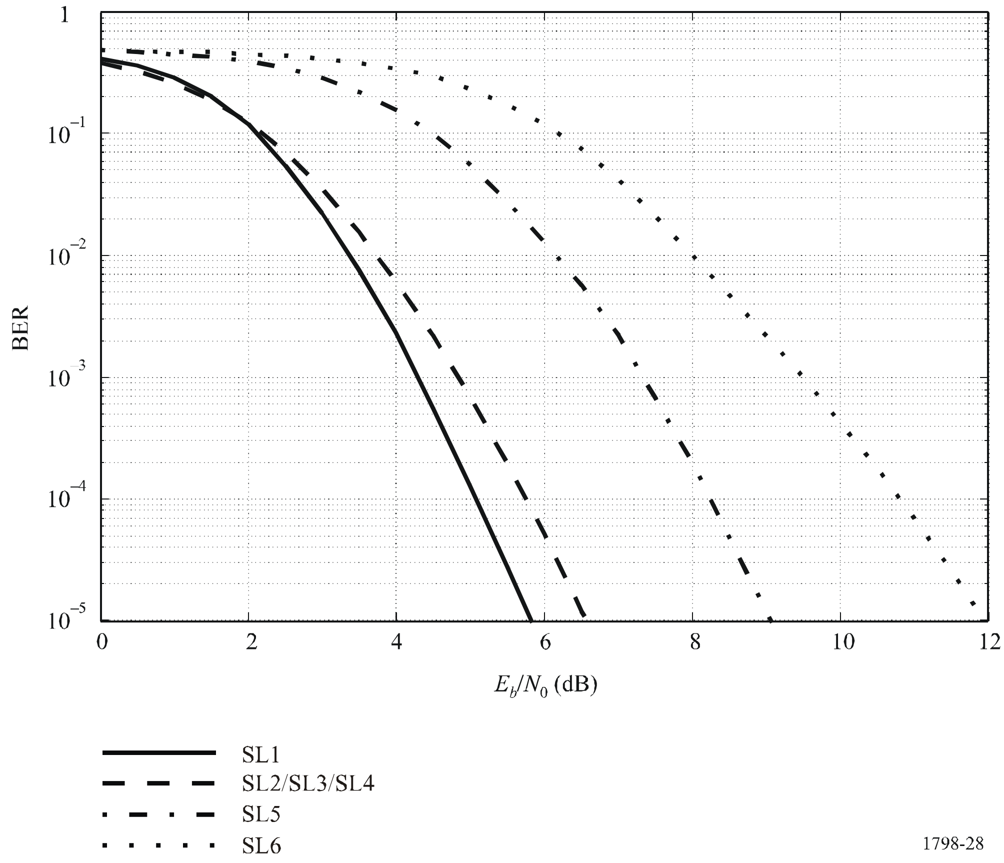
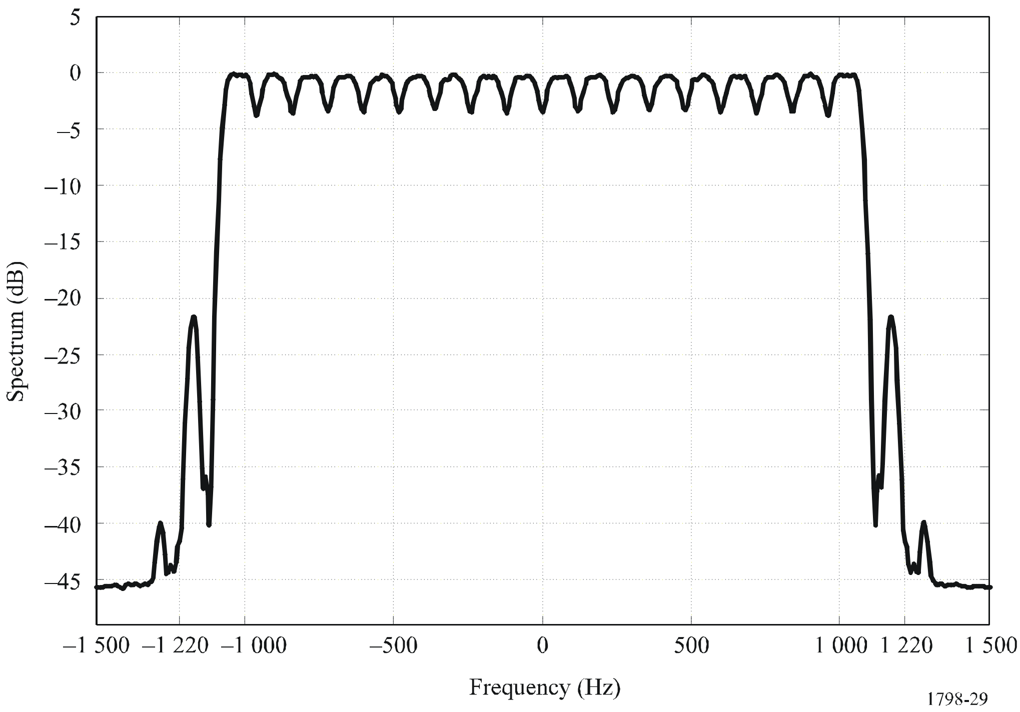


FIGURE 29

Spectrum of a PACTOR-III signal at SL6 with all 18 tones active



4.2 Typical communication protocol (T-BUS)

Interface Protocol

Physical characteristics:

8 data bits,
1 start bit,
1 stop bit,
1 parity bit,
odd parity,
2400 bit/second.

Word formats:

Address word

Sync	T/R	Remote Terminal Addr.

Reserved addresses:

C2h : Receiver
C3h : Transmitter
FFh : Broadcast

Command word

Subadr./mode	Wdcnt/modecod

Reserved commands:

00h : Reset
14h : Telex mode & frequency input
*) 24h : USB mode & frequency input
*) 34h : AM mode & frequency input
*) 44h : CW mode & frequency input
*) 85h : Set scan table entry & radiomode/entry nr. & frequency input
*) 90h : Step to next entry
*) A0h : Empty table
*) B1h : Go to table entry & entry nr.

*) Commands concerning DSC.

Data words

Frequency input:

10 MHz	1 MHz
100 kHz	10 kHz
1 kHz	100 Hz
10 Hz	1 Hz

Radio mode + entry nr. :

Radio mode	Entry nr.

1h : Telex mode Entry nr. = { 0h .. Fh }
2h : USB mode
3h : AM mode
4h : CW mode

Entry nr. :

Not used	Entry nr.

Entry nr. = { 0h .. Fh }

Status word

Err			Remote Term Addr.

Err : Error return status.

Message Format:
A message consists of an Address word followed by a Command word and possible corresponding Data words.

Example: TX 19.1201 MHz in Telex mode.

C3h
14h
19h
12h
01h
00h

4.3 Global Link Network (GLN)

General overview

The GLN is a network of cooperating coastal radio stations (CRSs) offering data access for the maritime mobile service. Due to the increasing demand for e-mail transfer and internet access on sea going vessels and the decreasing use of narrow-band direct-printing (NBDP) and radio telex, these radio stations now offer data services on shortwave.

Organizational structure

All CRSs are operated by independent companies. These companies have joined together to form the GLN. They use common technology and common modulation. The CRS are free to offer their own additional services depending on local requirements. If the connection to the network control centre (NCC) fails for political, military or other reasons, every station is able to operate independently. In such cases the CRSs may also offer long-distance communications outside the main communication networks.

Technical structure

The GLN is based on the so called Pactor IP-Bridge (PIB). PIB enables transparent data connections based on the TCP/IP protocol over 2k4 radio channels in all the maritime MF/HF bands. PIB may be used for any type of data service with a maximum transfer speed up to 5 600 bit/s compressed. All network servers operate with a Linux OS and additional software packages which ensure a high fail-safe performance.

NCC

The NCC is operated under an agreement with the CRSs. It is responsible for data bases, accounting, backup, data security and development. The NCC also operates a mail server for small stations without their own data infrastructure. The NCC offers basic data services like weather information, e-mail online compression, web mail, tracking and crew e-mail to all customers of the GLN network.

CRS

The CRSs hold one or more radio channels on standby for automated data links between vessels and the internet. They may offer additional services like data transfer (FTP), credit card services, web hosting and wireless server administration to specific customers. All CRSs continue to function if the connection to the NCC fails. The CRSs are responsible for their site installations, frequency assignments through their national authorities, power fail systems and solid IT infrastructures at their own locations. They are also responsible for all regulations, endorsements and licenses required by the local authorities. All CRSs may be operated remotely.

The CRSs use fixed frequencies in semi-duplex or simplex mode. They transmit a 100-baud FSK beacon signal on those channels which are not occupied. The beacon signal contains channel quality information, an appropriate call sign and information about channel availability. A Morse identifier may be inserted into the beacon signal if required.

Traffic lists are transmitted at regular intervals by all CRSs.

Ship earth station (SES)

The application required to join the GLN should be forwarded to a CRS. This application allows the SES to access any CRS inside the GLN without any additional registrations. To obtain an automated link, the SES may use existing MF/HF radios or a dedicated radio. The radio is connected to a specific communication server or the control software of the communication server may be integrated into new GMDSS terminals. The communication server may be connected to a ships data network and is a standard e-mail and web server. The server automatically selects the best free channel if data transfer is requested by the user. It also offers fallback capabilities if no radio channel are available.

Internet

All interconnections between the CRSs are via the internet. The CRSs may be connected to the internet by any available service like SDSL, ADSL, ISDN, or Dialup modems as well as Wi-Fi and satellite link. The total bandwidth per radio channel should not be less than 10 kbit/s. A fixed IP is not required for the radio sites. The GLN offers direct access to any web server world wide.

Interfacing

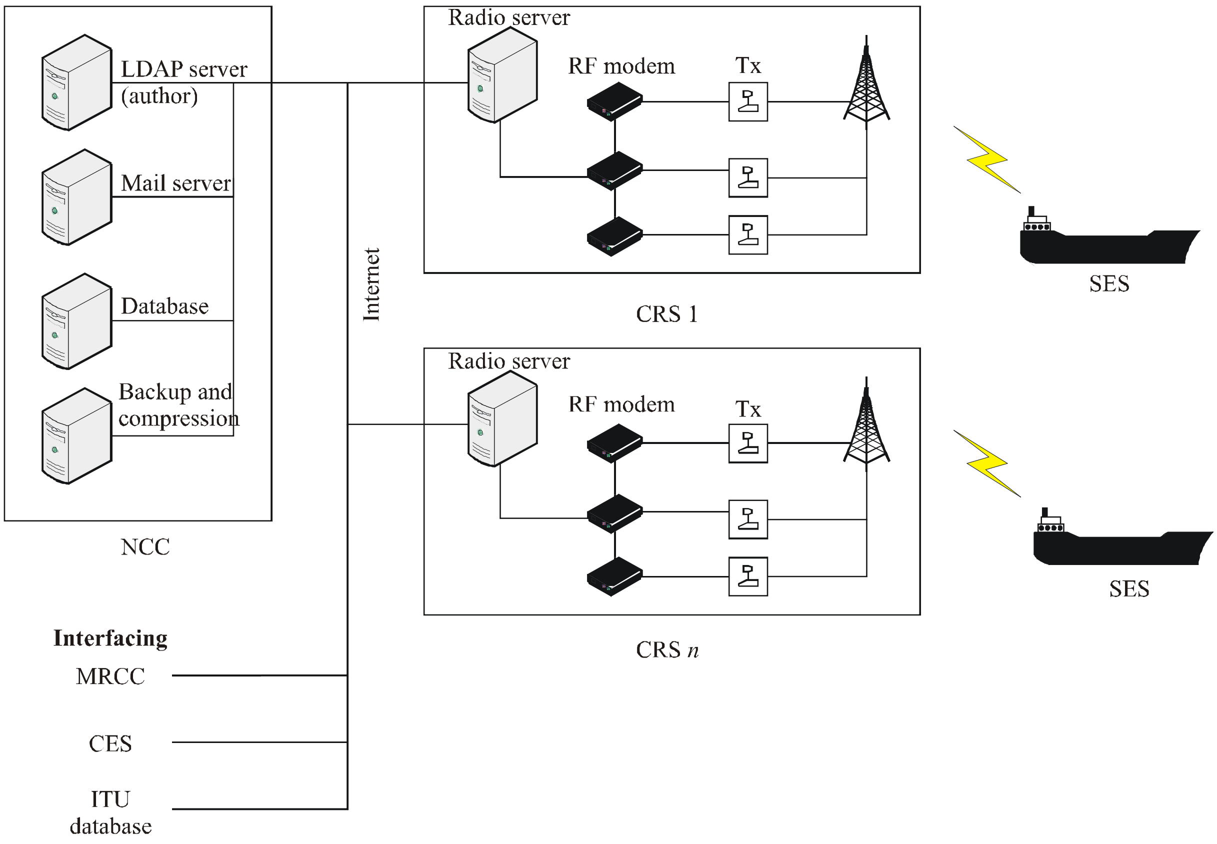
Due to the employment of standard internet technology at any part of the network, the GLN network is open for any additional services such as transfer of telemetric data, chat communications with other networks, position transfers and also for ship/ship and ship/shore communications.

Data security

Data is encrypted on all segments of intercommunication between the CRSs, SES and the NCC. Moreover, the data transferred on the radio link can not be read by other radio listeners. Firewalls, spam filters, virus scan and other security facilities are self-evident.

FIGURE 30

GLN general overview



MRCC: maritime rescue coordination centre

1798-30

Services

The GLN offers commercial communications as well as all types of communications which are currently covered by the radio telex system as part of the GMDSS. Because the PIB is able to transfer data below a S/N of 0, links are established under difficult conditions.

E-mail service

The GLN enables access to any e-mail server on the world wide web. Attachments and documents can be forwarded via GLN to and from shore. All data will be compressed online and interrupted connections will resume automatically with no double data transfer.

Weather information service

The GLN offers free weather download to all SESs. This includes weather faxes and forecasts as well as ice cards and grip data.

Vessel tracking

Position information is transmitted with every connection from SES to NCC and may be forwarded to any tracking service or e-mail address. An NMEA 0183 port is implemented to the system.

Crew mail

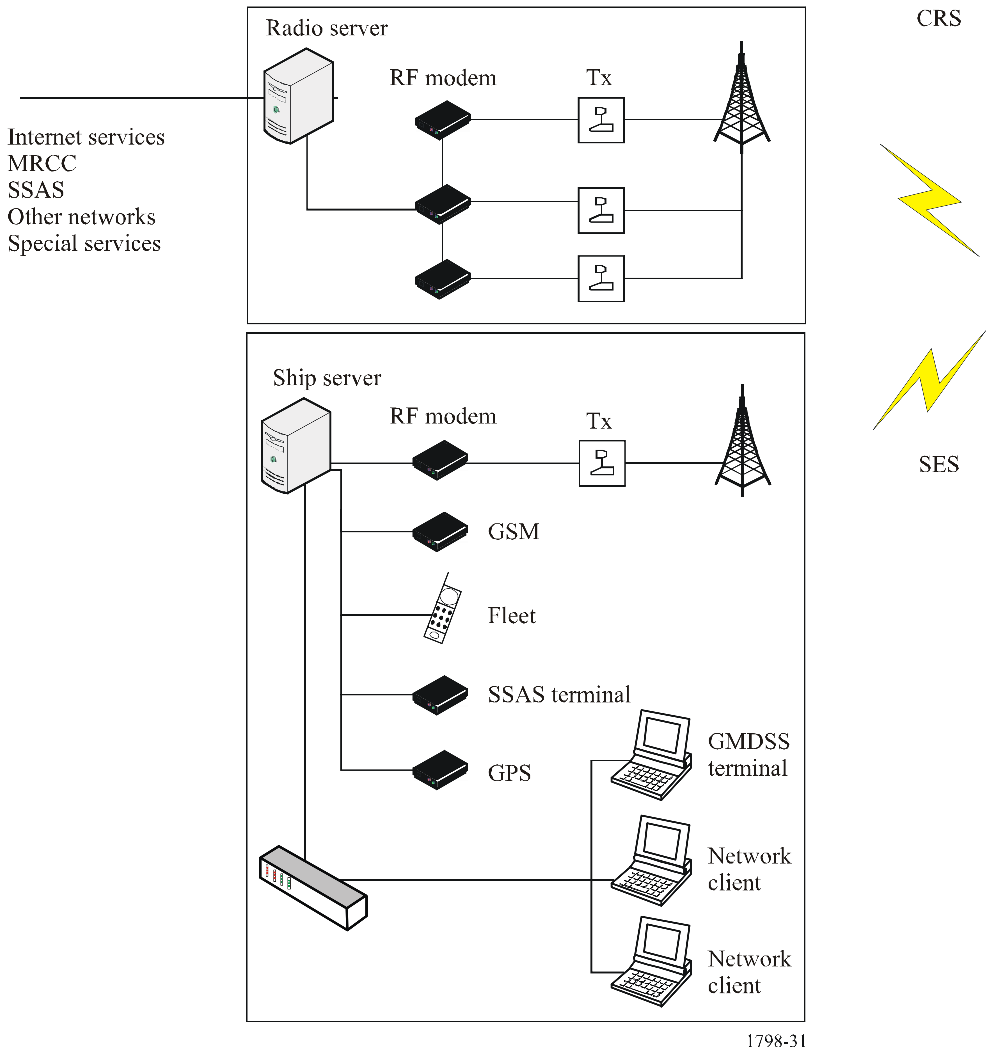
Up to 255 e-mail accounts may be implemented per vessel. They may be charged to the shipping company or the crew may pay by credit card directly to the CRS.

Ship security alert system (SSAS)

SSAS capability is implemented in the system.

FIGURE 31

Coastal radio and ship station overview



Coverage

The GLN offers world wide coverage. It is not a closed network and is open for new sites at any time. New stations inside the network benefit from worldwide coverage for vessels from the beginning. This is made possible by roaming technologies.

Range

Depending on their location and quality of radio equipment, environmental noise, antennas and transmission power used, the average range of each station is between 1.750 and 2.500 nautical miles.

FIGURE 32

GLN radio stations worldwide
(August 2006)

1798-32

Locations (August 2006 – subject to change)

Norway, 3 sites, up to 12 channels, 6 MHz, 8 MHz, 12 MHz

Germany, 1 site, 9 channels, 4 MHz, 6 MHz, 8 MHz, 12 MHz, 17 MHz

Switzerland, 1 site, 10 channels, 4 MHz, 6 MHz, 8 MHz, 12 MHz, 17 MHz

Kenya, 1 site, 15 channels, 4 MHz, 6 MHz, 8 MHz, 12 MHz, 17 MHz

Republic of South Africa, 1 site, 15 channels, 4 MHz, 6 MHz, 8 MHz, 12 MHz, 17 MHz

Angola, 1 site, 15 channels, 4 MHz, 6 MHz, 8 MHz, 12 MHz, 17 MHz

China, 1 site, 5 channels, 4 MHz, 6 MHz, 8 MHz, 12 MHz, 17 MHz

Philippines, 1 site, 5 channels, 4 MHz, 6 MHz, 8 MHz, 12 MHz, 17 MHz

Australia, 1 site, 5 channels, 4 MHz, 6 MHz, 8 MHz, 12 MHz, 17 MHz

Argentina, 1 site

Chile, 1 site

USA, RI, 1 site, 5 channels, 4 MHz, 6 MHz, 8 MHz, 12 MHz, 17 MHz

USA, WA, 1 site, 5 channels, 4 MHz, 6 MHz, 8 MHz, 12 MHz, 17 MHz

USA, AL, 1 site, 5 channels, 4 MHz, 6 MHz, 8 MHz, 12 MHz, 17 MHz