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International Telecommunication Union
Radiocommunication Sector

REPORT ITU-R M.2530-0

(09/2023)

M Series: Mobile, radiodetermination, amateur and related satellite services

Digital voice communication in the VHF maritime frequency band

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Note*: This ITU-R Report was approved in English by the Study Group under the procedure detailed in Resolution ITU-R 1.*

Electronic Publication
Geneva, 2023

REPORT ITU-R M.2530-0

Digital voice communication in the VHF maritime frequency band

(2023)

Scope

The purpose of this Report is to investigate a new approach for the possible expansion of the number of VHF maritime voice channels based on the implementation of digital technology. Analyses concerning operational reliability, impacts on the Global maritime distress and safety system (GMDSS), mode of operation (simplex/duplex), bandwidth, range, etc., which are the necessary objectives to determine the feasibility of implementation of digital voice radio telephony in the VHF maritime mobile band.

The Report contains:

– A first part, sections 1 to 6, that gives an overview of the current situation;

– A second part, section 7, that gives the requirements for voice and digital communication in the VHF band;

– A third part, sections 8 to 13, that describes possible ways to address coexistence with existing analogue and digital voice channels.

Keywords

Analogue communications, digital communications, DSC, GMDSS, migration, VHF radio.

Abbreviations/Glossary

ACELP	Algebraic code-excited linear prediction
ADPCM	Adaptive differential pulse-code modulation
AIS	Automatic identification system
AMBE	Advanced multi-band excitation
AMR-WB	Adaptive multi-rate wideband
ASM	Application-specific messages
ATIS	Automatic transmitter identification system
CELP	Code excited linear prediction
CS-ACELP	Conjugate-structure algebraic-code excited linear prediction
DCME	Digital circuit multiplication equipment
DMR	Digital mobile radio
dPMR	Digital private mobile radio
DSC	Digital selective calling
ETSI	European Telecommunications Standards Institute
EU	European Union
FDMA	Frequency division multiple access
FEC	Forward error correction
FRAND	Fair, reasonable and non-discriminatory

FSK	Frequency shift keying
GMDSS	Global maritime distress and safety system
HF	High frequency
IALA	International Association of Marine Aids to Navigation and Lighthouse Authorities
IMO	International Maritime Organization
LD-CEL	Low delay code excited linear prediction
LPC	Linear predictive coding
MF	Medium frequency
MMSI	Maritime mobile service identity
MSC	Maritime Safety Committee
NXDN	Next generation digital narrowband
PCM	Pulse code modulation
POTS	Plain old telephone service
PSTN	Public switched telephone network
RAINWAT	Regional Arrangement on the Radiocommunication Service for Inland Waterways
RALCWI	Robust advanced low complexity waveform interpolation
RR	Radio Regulations
SB-ADPCM	Sub-band adaptive differential pulse code modulation
SMS	Short message service
SNR	Signal to noise ratio
SOLAS	Safety of Life at Sea
TETRA	Trans-European trunked radio system
TDMA	Time division multiple access
TWELP	Tri-wave excited linear prediction
Tx	Transmission
VDES	VHF data exchange system
VHF	Very high frequency
VTS	Vessel traffic service
WMOPS	Weighted million operations per second
WRC	World Radiocommunication Conference

Related ITU Recommendations and Reports

Recommendation ITU-R M.493 – Digital selective-calling system for use in the maritime mobile service

Recommendation ITU-R M.541 – Operational procedures for the use of digital selective-calling equipment in the maritime mobile service

Recommendation ITU-R M.585 – Assignment and use of identities in the maritime mobile service

Recommendation ITU-R M.1084 – Interim solutions for improved efficiency in the use of the band 156-174 MHz by stations in the maritime mobile service

Recommendation ITU-R M.1171 – Radiotelephony procedures in the maritime mobile service

Recommendation ITU-R M.1309 – Digitally coded speech in the land mobile service

Recommendation ITU-R M.1808 – Technical and operational characteristics of conventional and trunked land mobile systems operating in the mobile service allocations below 869 MHz to be used in sharing studies in bands below 960 MHz

Report ITU-R M.2010 – Improved efficiency in the use of the band 156-174 MHz by stations in the maritime mobile service

Report ITU-R SM.2022 – The effect on digital communications systems of interference from other modulation schemes

Report ITU-R BT.2140 – Transition from analogue to digital terrestrial broadcasting

Report ITU-R M.2231 – Use of Appendix **18** to the Radio Regulations for the maritime mobile service

Report ITU-R M.2288 – Digital voice communication system on MF/HF radio channels of the maritime mobile service for shore-to-ship/ship-to-shore applications

Report ITU-R M.2474 – Conventional digital land mobile radio systems

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1 Introduction

Voice radio telephony in the VHF maritime mobile frequency band is the most important communication for shipping. At this moment the congestion in the VHF maritime mobile frequency band has become a serious problem around the world and is continuing to grow. Also, as a consequence of the implementation of digital selective calling (DSC), Automatic identification system (AIS), and VHF data exchange system (VDES) the availability of voice channels in the VHF maritime mobile frequency band has been reduced rapidly and exacerbates congestion problems.

The first intent to cope with this problem was the reduction to 12.5 kHz/6.25 kHz bandwidths as indicated in Recommendation ITU-R M.1084, but it has never been implemented.

Recommendation ITU-R M.1084 provides ways to improve efficiency in the use of the frequency band 156-174 MHz by stations in the maritime mobile service; it specifically describes technical characteristics when using channels spaced by 12.5 kHz and 6.25 kHz, migration to narrow-band channels, an example method for implementing interleaved narrowband channels at 12.5 kHz or 6.25 kHz offset spacing and assignment of channels numbers to interleaved channels and simplex operation of duplex channels.

This ITU-R Report provides analysis and investigates the feasibility of implementation of digital voice radio telephony in the VHF maritime mobile frequency band. This Report could be the base for future discussions.

2 Current regulation

2.1 International Telecommunication Union

A review and information provided by the ITU members produced the following list of possible ITU documents related to the subject.

2.1.1 Regulations/Recommendations/Reports

Radio Regulations Appendix **18**.

See the section Related ITU Recommendations and Reports of this Report.

The Radio Regulations Appendix **18** describes which frequencies and bandwidth are designated to which channel and what should be the use of the channel, for instance, analogue or digital. Together with the performance standards and resolutions of the International Maritime Organization (IMO), this is the bases of the use of VHF radio channels by analogue or digital communications.

Recommendation ITU-R M.1084 recommends how to assign channel numbers when 25 kHz channels are divided into 12.5 or 6.25 kHz sub-channels. This Recommendation with minor adjustments could be used when assigning new channel numbers to the digital voice channels and depending on the chosen technique used to automatically switch to analogue if required.

2.2 International Maritime Organization

As stated in the previous paragraph, VHF radio communication performance standards for maritime radio are defined by IMO in the International Convention for the Safety of Life at Sea (SOLAS) and IMO resolutions. This section will focus on these performance standards and resolutions.

2.2.1 Regulations as of July 2023

IMO performance standards:

– Resolution A.694(17): General requirements for shipborne radio equipment forming part of the Global Maritime Distress and Safety System (GMDSS) and for electronic navigational aids;

– Resolution MSC.511(105): Performance standards for shipborne VHF radio installations capable of voice communication and digital selective calling;

– SOLAS Convention Chapter IV/5.1.3;

– SOLAS IV 7.1.1;

– SOLAS IV 7.1.2.

2.2.2 Performance standards

The main performance standards for VHF shipborne radio installations are IMO Resolution MSC.511(105).

These performance standards define the minimal requirements for the use, availability, installation, robustness, etc. of VHF radios on board ships.

Below is an overview of the most important requirements from IMO Resolution MSC.511(105).

General:

The installation, which may consist of more than one piece of equipment, should be capable of operating on single-frequency channels or on single- and two-frequency channels.

The equipment should provide for the following categories of calls using both voice and digital selective calling (DSC):

– distress, urgency and safety;

– ship operational requirements; and

– general radiocommunications.

The equipment should provide for the following categories of communications using voice:

– distress, urgency and safety;

– ship operational requirements; and

– general radiocommunications.

The equipment should comprise at least:

– a transmitter/receiver including an antenna;

– an integral control unit or one or more separate control units;

– a microphone with a press-to-transmit switch, which may be combined with a telephone in a handset;

– an internal or external loudspeaker;

– an integral or separate digital selective calling facility; and

– a dedicated DSC watchkeeping facility to maintain a continuous watch on channel 70.

The installation may also include additional receivers.

A distress alert should be activated only by means of a dedicated distress button. This button should not be any key of an ITU-T digital input panel or an ISO keyboard associated with the equipment and should be physically separated from functional buttons/keys used for normal operation. This button should be a single button for no other purpose than to initiate a distress alert.

The dedicated distress button should:

– be clearly identified;

– be protected against inadvertent operation.

The distress alert initiation should require at least two independent actions.

The equipment should indicate the status of the distress alert transmission.

It should be possible to interrupt and initiate distress alerts at any time.

Class of emission, frequency bands and channels:

– the equipment should be designated for operation on channels selected from and in accordance with Appendix **18** of the Radio Regulations;

– the radiotelephone facility should be capable of operating as follows:

  • in the frequency band 156.025 MHz to 157.425 MHz on single-frequency channels as specified in Appendix **18** of the Radio Regulations;

  • in the frequency band 156.025 MHz to 157.325 MHz for transmitting and the frequency band 160.625 MHz to 161.925 MHz for receiving on two-frequency channels as specified in Appendix **18** of the Radio Regulations;

– the digital selective calling facility should be capable of operating on Channel 70;

– class of emission should comply with Chapter IX of the Radio Regulations;

– the frequency tolerance for ship station transmitters should not exceed 10 parts in 10^6.

Controls and indicators:

– Channel control and switching:

  • change of channel should be capable of being made as rapidly as possible, but in any event within five seconds;

  • the time taken to switch from the transmit to the receive condition, and vice versa, should not exceed 0.3 seconds;

  • an on/off switch should be provided for the entire installation with a visual indication that the installation is switched on;

  • a visual indication that the carrier is being transmitted should be provided;

  • the equipment should indicate the four-digit channel number, as given in the Radio Regulations Appendix **18**, to which it is tuned. It should allow the determination of the channel number under all conditions of external lighting. Where practicable, channels 16 and 70 should be distinctively marked;

  • the equipment should not be able to transmit during a channel switching operation; and

  • operation of the transmit/receive control should not cause unwanted emissions.

– Radiotelephone facility:

  • provision should be made for changing from transmission to reception by use of a press-to-transmit switch. Additionally, facilities for operation on two-frequency channels without manual control may be provided;

  • the receiver should be provided with a manual volume control by which the audio output may be varied; and

  • a squelch (mute) control should be provided on the exterior of the equipment.

Permissible warming-up period:

– the equipment should be operational within five seconds after switching on.

Transmitter output power:

– the transmitter output power should be between 6 and 25 W;

– provision should be made for reducing the transmitter output power to a value of less than 1 W. However, this reduction of the power is optional on Channel 70.

Receiver parameters:

– Radiotelephone facility:
  - the sensitivity of the receiver should be equal to or better than 2 µV e.m.f. for a signal-to-noise ratio of 20 dB.

  Digital selective calling facility:
  - with a DSC modulated input signal having a level of 1 µV e.m.f. to its associated VHF receiver, the DSC equipment should be capable of decoding the received message with a maximum permissible output character error rate of 10^{-2}.

  Immunity to interference:
  - the immunity to the interference of the receiver should be such that the wanted signal is not seriously affected by unwanted signals.

2.3 International Association of Marine Aids to Navigation and Lighthouse Authorities

For the shoreside, the International Association of Marine Aids to Navigation and Lighthouse Authorities (IALA) defines Standards, Guidelines and Recommendations on the use and operation of VHF voice radio to communicate with shipping on their waters/area of responsibility for safety. In the process of adaption and migration to digital VHF radio, all parties involved need to be addressed, to ensure proper and clear communications between ships and ships and shore.

Although, there were initiatives before IALA started their discussion in February 2019 at an IALA ENAV WG3 intercessional meeting with a presentation on digitization of VHF voice radio. This received a follow-up at ENAV23 in Singapore where several presentations were given on possible more efficient use of the spectrum currently in use by VHF radio. The presentations at that moment focussed mainly on one specific international standard, namely digital private mobile radio (dPMR), as an example, but there are other possible solutions.

From this IALA ENAV meeting, a liaison note was sent from IALA to the European Conference of Postal and Telecommunications Administrations FM58 (on Maritime Radio) and International Electrotechnical Commission stating IALA's view on the digitization of VHF radio.

2.3.1 Correspondence

IALA in their liaison note on 4 April 2019 stated the following (IALA ENAV Committee, 2019-04) text:

> "IALA recognises and agrees that the maritime voice service on the maritime VHF band should be digitised. In this respect, IALA recommends that the following shall be included in the consideration:
>
> – That any evaluated technologies should have a clear migration path both from the current analogue voice services to the new digital voice services by allowing both the digital and analogue services to co-exist in the same transceiver for the duration of the entire migration period. This could extend to using the same antenna and other existing physical installation hardware;
>
> – The channel efficiency be a high priority by allowing four (4) or more digital voice channels for each 25 kHz maritime VHF voice channel;

– The digital service includes the capability of transmitting the location of the radio for the entire duration of the digital voice conversation;

– The digital service allows a short message service (SMS) without the need to set up a digital or other voice call;

– The digital voice quality be similar to or better than the analogue voice service especially using weaker radio signals at the extents of the radio coverage.

IALA has evaluated digital Private Mobile Radio (dPMR) as one of the candidate technologies and is able to share high-level evaluation."

2.3.2 IALA Standards, Guidelines and Recommendations

From a review of the possible IALA documents related to the use of VHF radio, the following documents were identified:

– Maritime Radio Communications Plan (MRCP) (Edition 15 December 2017);

– R1012 VTS communications (Edition 1.2 January 2022);

– G1089 Provision of VTS (Edition 2.0 January 2022);

– G1111 (Series) Establishing functional and performance requirements for VTS systems (Edition 2.0 December 2022);

– G1132 VTS voice communications and phraseology (Edition 2.2 January 2022).

2.4 Other bodies/countries/companies

2.4.1 Regional Arrangement on the Radiocommunication Service for Inland Waterways

The Regional Arrangement on the Radiocommunication Service for Inland Waterways (RAINWAT) regulates radiocommunication on some European inland waterways. It is based on Art. 6 of the RR and concluded between administrations.

The current revision of the RAINWAT agreement is from 11 October 2016.

There are some differences between the performance requirements of IMO as stated in paragraph 3.3 and the operational and technical requirements of the equipment as stated in Annex 3 of the RAINWAT agreement. The most important differences are:

– dual watch is not allowed.

– DSC usage is not allowed in radiocommunication on Inland Waterways.

– Identification of the radio by automatic transmitter identification system (ATIS) code is required and the reception of the ATIS signals on the loudspeaker or handset can be suppressed by suitable technical measures.

– The output power for mobile VHF radiotelephone equipment shall be set to a value between 0.5 and 25 W; however:

  – the output power for frequencies designated for service categories ship-to-ship, ship-to port and on-board communications shall be limited automatically to a value between 0.5 and 1 W;

  – for nautical information the Administrations may demand the reduction of the output power to a value between 0.5 and 1 W for vessels within their territory.

– Output power for handheld equipment used on Inland waterways shall be set to a value between 0.5 and 6 W. The following exceptions apply:

– the output power for frequencies designated for service categories ship-to-ship, ship-to port and on-board communications shall be limited automatically to a value between 0.5 and 1 W;
– for nautical information the Administrations may demand the reduction of the output power to a value between 0.5 and 1 W for vessels within their territory.

2.4.2 Cybernetica

Cybernetica made a report on “Analyses of different digital radio protocols for use in maritime communication (Y-399-70)” where they compared different digital radio protocols. This analysis resulted in the following advice:

> “Technically, the standards are comparable and offer similar functionality. However, the dPMR published as open ETSI standard is more preferred.”

The full report can be found in Annex 1.

2.4.3 Estonia

The test in Estonia provided the following insights:

The participants in the test were generally positive about the introduction of digital communication. Also, the range of digital communications was the same (or better) than the range of analogue communication. At maximum distances that the digital dPMR communication was understandable (d = 19.6 NM) – analogue communications experienced very high noise and was not understandable. During the digital switchover period, when the digital station and the analogue station are very close together, the digital station signal will overlap the analogue channel, but this did not cause any interference during testing.

The full report can be found in Annex 2.

2.4.4 The Netherlands

The Netherlands did a trial in The Port of Rotterdam to test digitisation of VHF radio with the main purpose to include users in the process. Before and during the trial several issues were raised to consider:

– An appropriate used voice codec and digital VHF system seems to be a possible solution for a future replacement or co-existence of the present analogue VHF.
– In principle: one analogue voice Channel can be converted into 4 digital voice channels.
– Technologies should have in principle a clear migration path both from current analogue voice services to new digital voice services by allowing both digital and analogue services to co-exist in the same transceiver for the duration of the entire migration period. This could be extended to using the same antenna and other existing physical installation hardware.
– The technology should confirm during the migration period to the standards set by international bodies like IMO and ITU. Next to this additional regional functionality could be applicable.
– Digital services should include the capability to embed additional information such as short messages (SMS) and position information of the radio for the entire duration of the digital voice conversation.
– Digital voice quality should be similar or better than the analogue voice quality, especially at the extent of the radio coverage where the radio signals are weaker.
– From a financial point of view digital radios should have the same price as analogue radios.
– The present techniques of digital equipment should detect an analogue or digital signal and switch over automatically to the right frequency.

– In relation to maritime there should be clear functional requirements, such as: must have, need to have, nice to have, undesirable and compare this to possible different technical standards and solutions.

The full report can be found in Annex 3.

2.4.5 Comité International Radio-Maritime

The view of the Comité International Radio-Maritime is that the following issues need to be considered in the ongoing work on this subject:

– a cost benefit analysis covering the entire maritime community, i.e. VHF coast stations, commercial shipping, recreational boating, ports, harbours and marinas etc.;
– the relative merits of Time division multiple access (TDMA) vs. Frequency division multiple access (FDMA) (e.g. in terms of frequency spectrum efficiency);
– management of co-existence of digital and analogue channels;
– should there be an allocation of a separate channel for digital distress, safety and calling (or should Channel 16 or Channel 70 take on this role);
– what will be the future relationship between Digital Voice and DSC (e.g. could the same technology be used for DSC);
– could digitisation create an opportunity for SMS on VHF channels and position information with low overhead;
– what ITU alignment issues are involved;
– an implementation plan that clearly sets out how digitisation would be introduced and the impact of each stage on the maritime community.

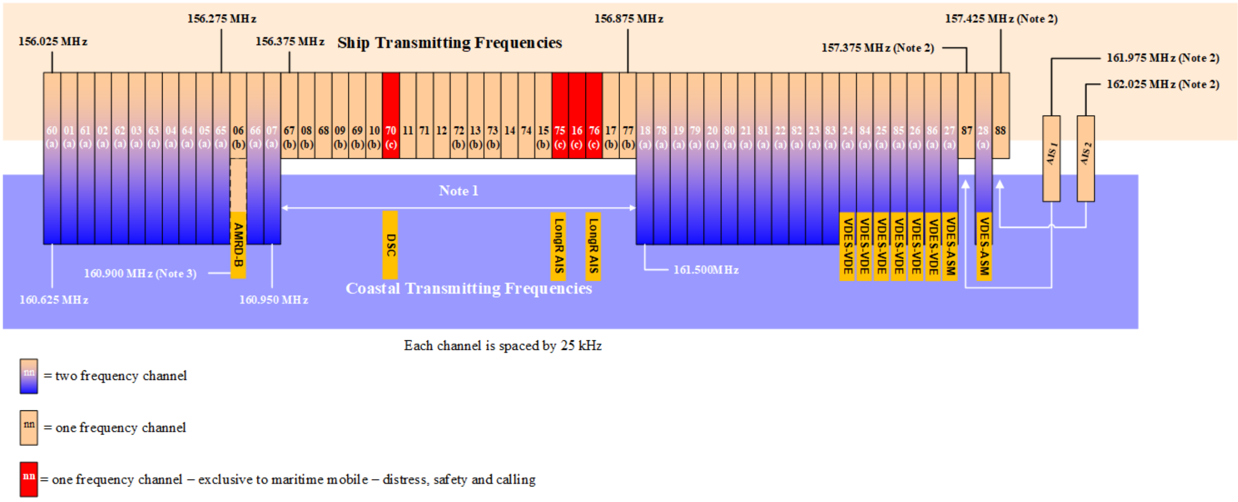
3 Overview of the current situation

This section gives an overview of how the maritime VHF service is built up in general at the moment and the impact of VDES from 1 January 2028 as stated in IMO MSC.1/Circ.1460/Rev.4.

The current use of the VHF frequency band after WRC-19 is shown in Fig. 1.

FIGURE 1

The current VHF frequency plan after WRC-19



Channels; 06, 08, 09, 10, 13, 15, 17, 67, 69, 72, 73, 77 – are identified for inter-ship use (indicated by (b) in the above diagram)

Channels; 01-07 (inclusive), 18-28 (inclusive), 78-86 (inclusive) – are identified for public correspondence (indicated by (a) in the above diagram)

Note 1: Whilst the frequency range/channels appears to be within the scope of RR Appendix 18, it should be noted that the 'gaps' as indicated are general mobile allocations with no footnote priority to maritime use.

Note 2: Change as a result of WRC-97, that split the original channels; 87 and 88, into four single frequency channels. The 'high' frequencies were identified for AIS, with the remaining 'low' single frequencies now designated as channels 87 and 88 (i.e. no longer two frequency, duplex channels, but single frequency simplex channels).

Note 3: This channel, whilst not explicitly referenced in Appendix 18, falls with the bands identified within RR Article 5 footnote, 5.226 and is upper frequency related to channel 6. This channel is now used from WRC2019 as ARMD Group B with or without AIS technology

In the past, several analogue VHF voice channels were given over to other maritime purposes than voice. Examples of these are DSC and AIS. Most recently from 2012 VHF channels are given to the development of IMO's e-Navigation where channels were assigned to VDES and autonomous maritime radio devices group B.

4 Maritime radio use cases

The maritime environment can be divided into two different zones, the open sea and the coastal/inland areas, such as harbours and rivers.

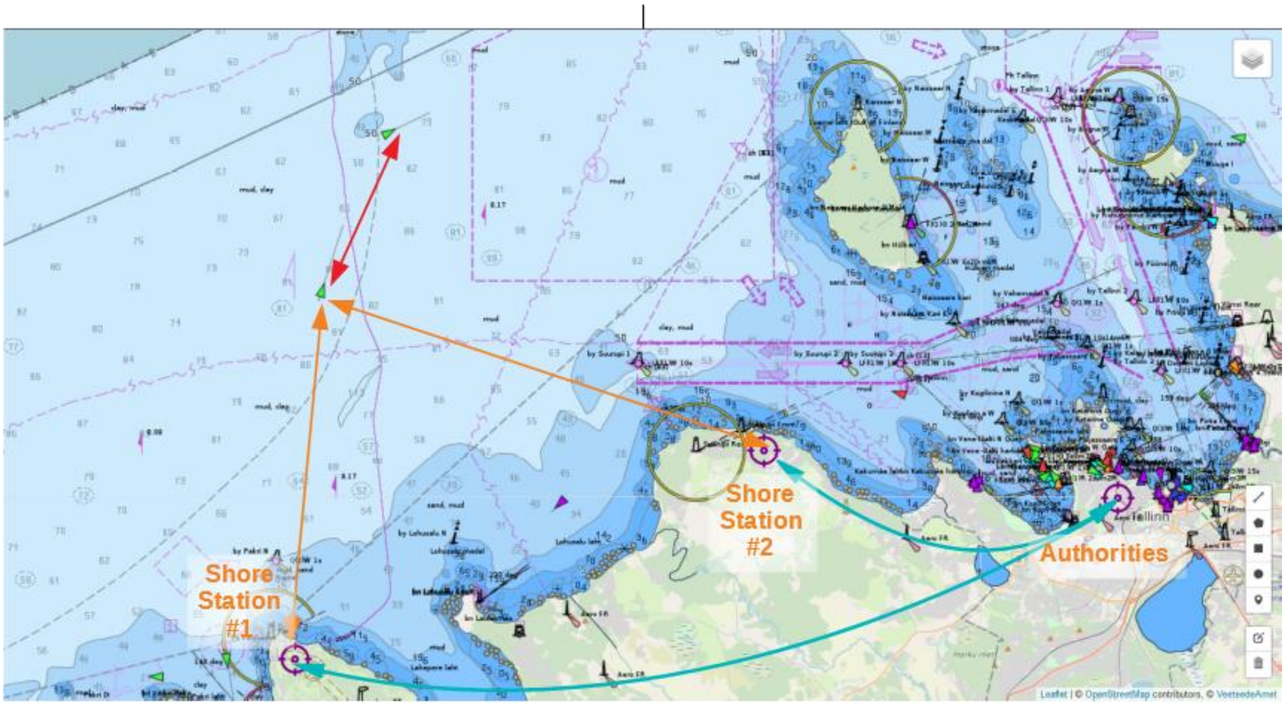
In marine environment, voice communication over VHF frequency band is widely used. There are two main usage cases:

– Ship-to-Ship communication. Normally vessels are listening to Channel 16 (distress and alerting frequency). Therefore, Channel 16 is used as call channel, to negotiate working channel (normally channel 6). If the vessels maritime mobile service identity (MMSI) address is known, a preferable method is to use digital selective calling (DSC) call to negotiate the working channel.

– Ship-to-Shore communication. Protocols are similar. Coastal stations are using fixed traffic channels, Channel 16/DSC can be used to advise vessels to switch to working channel.

FIGURE 2

Maritime communication



1. Ship to Ship direct radio traffic
2. Shore to Ship radio traffic
3. Shore to Ship TCP/IP traffic

Ship-to-Ship communication is implemented using simplex channels (vessels cannot contact each other over duplex channel). Ship-to-Shore communication can be over duplex channel but, in this case, other vessels can hear only the coast part of communication. In some situations (security related communication), it is preferred that all vessels can listen to both parties. When free simplex channels are not available coastal station broadcast vessel TX back to sea on coastal TX frequency (The Gulf of Finland reporting system working channels are used like this).

With satellite and mobile communication development, duplex channels are rarely used to VHF-public switched telephone network (PSTN) connections (commercial service), therefore it will not be taken into account in this Report.

5 Terrestrial radio usage

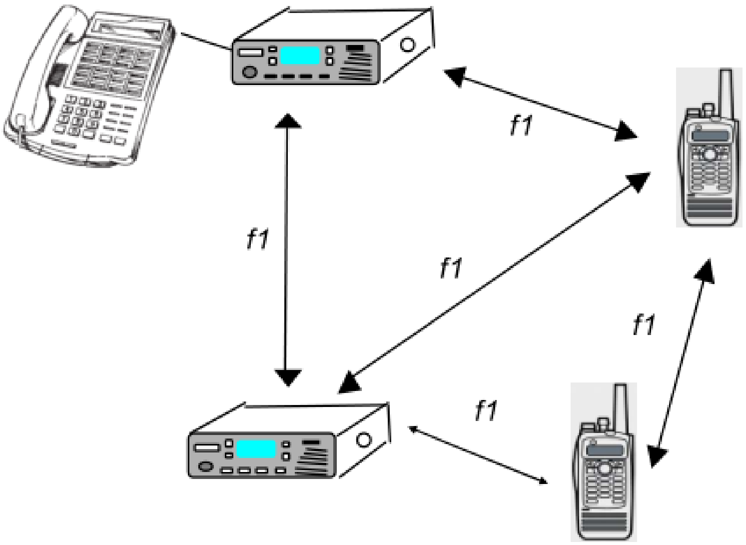
In terrestrial communication, mainly two types of communication are in use: peer-to-peer direct network and centralised repeater network.

In direct network mode, all radios in network work in the same frequency. There are no master-slave relationship and each radio in responsible for channel access rules. This is how maritime communication works today in the analogue VHF frequency band.

In a digital network, communication can be individual (like analogue VHF Ship-to-Ship call in traffic channel), group calls (used in DSC group call) and Broadcast Call.

FIGURE 3

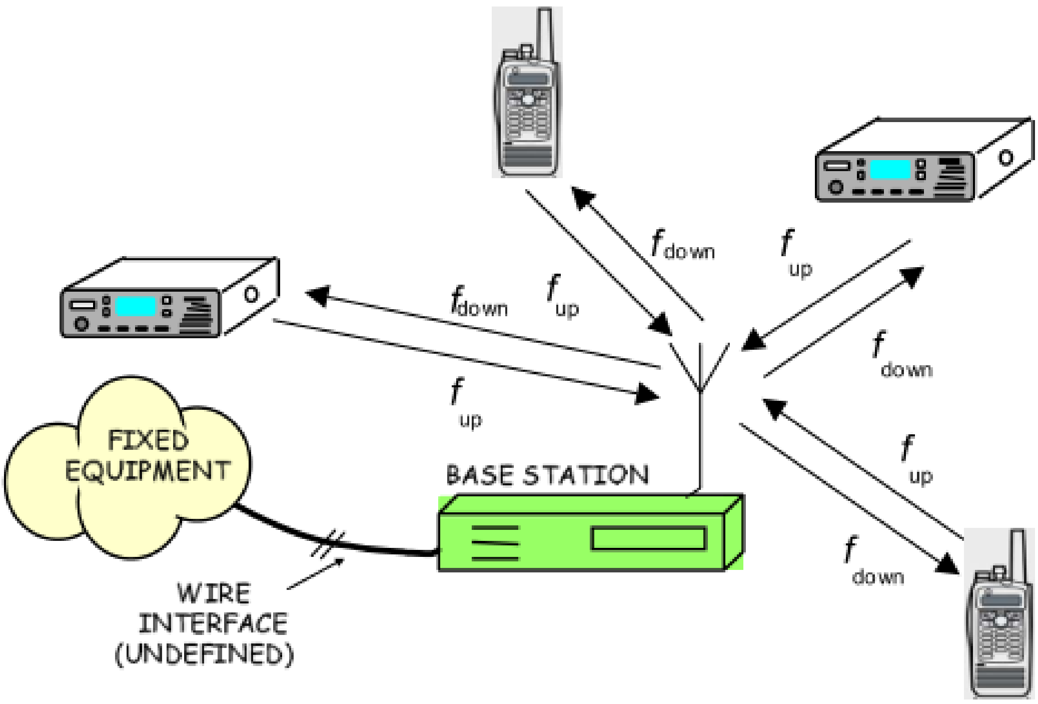
Peer-to-peer direct network



In a centralised repeater network, communication follows a star topology. All communication is between base station and radio. Two frequencies could be in use – Tx and Rx are separate frequencies as in the maritime duplex channels. In the repeater network it is possible to extend the network via additional inter-connected base stations (as communication between radios in different base stations coverage).

FIGURE 4

Centralised repeater network



6 Digital voice VHF Appendix 18 of the Radio Regulations

6.1 Technical characteristics of marine VHF radios

Marine VHF radios with DSC (used for distress calling and general automated calling purposes, and it is required by most administrations) capability are divided into three main classes:

– Class A, mandatory carriage for SOLAS ships under GMDSS SOLAS Chapter IV;

– Class D, voluntary carriage by non-SOLAS vessels;

– Class H, handheld radios for non-SOLAS voluntary carriage.

Some of these have integral Global navigation satellite system, and some also have texting capability (RTCM SC123 standard).

Digital voice-capable radios would need to meet the current requirements of their respective classes.

6.2 Technical characteristics of the VHF maritime frequency band, RR Appendix 18

The VHF maritime frequency band covers the frequency ranges 156.025 to 157.425 MHz, and 160.625 to 162.025 MHz, with a gap in the middle frequency range between 157.425 and 160.625 MHz, and it is channelized in 25 kHz channels with channel numbers in two digits and four digits in accordance with Recommendation ITU-R M.1084-5.

RR Appendix **18** contains simplex and duplex channels, and some of the duplex channels may be used in simplex mode, designated by four-digit channel numbers.

Footnotes are used in RR Appendix **18** to designate how the channels are used, for example:

– for DSC, channel 70, footnotes *j)* and *f)*;

– for AIS, channels AIS 1 and AIS 2, footnotes *f), l),* and *p)*;

– for VDES, numerous channels in both the lower and upper frequency ranges, footnote *w)*;

– for Application specific message (ASM)1 and ASM2, footnote *z)*.

7 Requirements for voice communication and associated digital selective calling in the VHF frequency band

Gathered from the previous chapters, the following requirements could be composed (the source of regulation is given in brackets):

– should be capable of operating on single-frequency channels or on single- and two-frequency channels (IMO);

– should provide four levels of priority of communications using voice (IMO);

– a dedicated DSC watchkeeping facility to maintain a continuous watch on channel 70 (IMO);

– DSC facility should be capable of operating on Channel 70 (IMO);

– should provide four categories of calls using both voice and DSC (IMO);

– should be capable of disabling DSC capabilities (RAINWAT);

– switch time between transmit and receive condition, and vice versa, should not exceed 0.3 seconds (IMO);

– reducing the transmitter output power to a value of less than 1 Watt (IMO);

– reduction of the power is optional on Channel 70 (IMO);

– the sensitivity of the receiver should be equal to or better than 2 μV e.m.f. for a signal-to-noise ratio of 20 dB (IMO);

– with a DSC modulated input signal having a level of 1 μV e.m.f. to its associated VHF receiver, the DSC equipment should be capable of decoding the received message with a maximum permissible output character error rate of 10^{-2} (IMO);
– should allow both the digital and analogue services to co-exist during the migration to future digital services (IALA/NL);
– channel efficiency should have a high priority by allowing four (4) or more digital voice channels for each 25 kHz maritime VHF voice channel. (IALA/NL);
– the ability to regularly transmit the location of the radio for the entire duration of the digital voice communication (IALA/NL);
– Short Message Service (SMS) without the need to set up a digital or other voice call (IALA/NL);
– digital voice quality be similar to or better than the analogue voice service especially using weaker radio signals at the extents of the radio coverage (IALA);
– the capability of transmitting the identification (MMSI) of the radio for the entire duration of the digital voice conversation (ITU);
– the capability of transmitting the identification (ATIS) of the radio for the entire duration of the analogue voice conversation (RAINWAT);
– could/should detect an analogue or digital signal and switch over automatically to the appropriate frequencies (NL);
– should be possible for Coastal stations to relay transmissions (with the same voice quality in digital mode) (NL);
– should be able to work in a network (trunked system) (NL);
– should be able to auto connect to a trunked system (NL);
– mobiles should automatically be activated (transmit SAR position and identification);
– be able of “graceful degradation”.

8 Voice CODECS

It is the view that licensing and patents are important considerations when selecting an appropriate vocoder. Use of technology should not involve patents unless a patent owner was prepared to donate or sell the patent. Although this is the “best” for the maritime community, sometimes a small fee for the patent could be acceptable if a formal declaration acknowledging the patent holder is in place with the right agreements. The desired technology should comply with the ITU policy on intellectual property.

The VHF frequency band today offers channels with a spectrum bandwidth of 25 kHz for analogue speech communications. Using technologies available today this can be split up to improve the spectral efficiency by applying digital encoding techniques to the speech signals. For instance, the Trans-European trunked radio system (TETRA) system allows for four TDMA speech channels to be carried over its 25 kHz radio channel by encoding each speech channel into a 7 200 bit/s data stream, which, after removing the error correction, results in a 4 800 bps voice channel. Adding control and overhead signalling, this results in a total over-air data rate of 36 kbit/s using pi/4 QPSK modulation.

Alternatively, following an FDMA approach and splitting the 25 kHz radio channel into four separate radio channels yields a channel bandwidth of 6.25 kHz. Again, by using conventional technology, this allows for an over air data rate of 4 800 bit/s, using four frequency shift keying (FSK) modulation and still remaining within the adjacent channel power limits. Removing the signalling overhead, this results in a speech channel of 3 600 bit/s, of which approximately 1/3 is used for error correction, so that the data channel available for encoding the speech waveform is approximately 2 400 bit/s.

Two of the most used vocoders for mobile radio at present are AMBE+2 and Algebraic code-excited linear prediction (ACELP). AMBE+2 is used by digital mobile radio (DMR) and dPMR, and ACELP is used by TETRA. Both AMBE+2 and ACELP are covered by patents.

8.1 CML Microcircuits

CML Microcircuits offers various solutions for vocoders for use in low data rate radio channels, among them the CMX7262 and CMX618/638.

Detailed information on these vocoders could be found here: https://www.cmlmicro.com/digital-voice-2/

These are available as hardware devices on the open market with no additional license fees.

The robust advanced low complexity waveform interpolation (RALCWI) algorithm is covered by patents but at the IALA/JCG conference in Tokyo, February 2023, CML announced that it will also be made available under the European Telecommunications Standards Institute (ETSI) fair reasonable and non-discriminatory (FRAND) policy (Fair, Reasonable And Non-Discriminatory)

RALCWI has been successfully implemented on dPMR equipment in place of AMBE+2.

8.2 Digital voice systems

Digital Voice Systems offers various solutions for vocoders, among them the AMBE+2 series which could be found on https://www.dvsinc.com/products/compare.shtml

AMBE+2 is covered by patents, but it is made available under licences by their developers under the ETSI FRAND policy (Fair, Reasonable And Non-Discriminatory). Whilst this does impose a cost on each radio produced, it is consistent for all manufacturers; however, this may make smaller manufacturers hesitate to enter the market.

8.3 VoiceAge

VoiceAge Corporation has developed the ACELP which is a patented speech coding algorithm.

The ACELP method is widely employed in current speech coding standards such as: AMR, EFR, AMR-WB (G.722.2), VMR-WB, EVRC, EVRC-B, SMV, TETRA, PCS 1900, MPEG-4 CELP and ITU-T G-series standards G.729, G.729.1 (first coding stage) and G.723.1. The ACELP algorithm is also used in the proprietary ACELP.net codec. More information could be found on: http://www.voiceage.com/Overview.html.

ACELP is covered by patents, but is made available under licences by their developers under the ETSI FRAND policy (Fair, Reasonable and Non-Discriminatory). Whilst this does impose a cost on each radio produced, it is consistent for all manufacturers; however this may make smaller manufacturers hesitate to enter the market. The licence for ACELP is significantly lower than that charged for the AMBE+2 and is a single one-off charge which makes it much simpler to manage and cost.

8.4 ITU-T Codecs

ITU-T has described a number of Codecs as listed below. Details are contained in Annex 4.

– ITU-T Codecs – G.711;

– ITU-T Codecs – G.722 and G.722.1;

– ITU-T Codecs – G.723.1;

– Recommendation ITU-T G.726: (ADPCM);
– Recommendation ITU-T G.728: Coding of speech at 16 kbit/s using LD-CELP;
– Recommendation ITU-T G.729: Coding of speech at 8 kbit/s using CS-ACELP;
– Recommendation ITU-T G.729.1: ITU-T G.729 based embedded variable bit-rate coder: An 8-32 kbit/s, scalable wideband, coder-bitstream interoperable with ITU-T G.729 codecs.

8.5 Opus (free open audio codec)

Opus is a totally open, royalty-free, highly versatile audio codec. More details can be found on http://opus-codec.org. It supports bit rates from 6 to 510 kbit/s.

8.6 Comparison of voice CODECS

Extensive studies have been performed, and the results of these studies to date are summarized in Table 1.

TABLE 1

Comparison of voice CODECS

Codec (5)	**Audio bandwidth** (1)	**Samples**	**Speed (bit/s)**	**FEC**	**Use policy**	**Quality**
CMX 7262 (TWELP) (2)	300-3 400		2 400	add	Patent	POTS (4)
CMX 618/638 (RALCWI) (2)	300-3 400		2 400	add	Patent	POTS (4)
AMBE+2 (3)	300-3 400		2 450	add	Patent (FRAND)	POTS (4)
ACELP	300-3 400		7 200	Incl.	Patent (FRAND)	POTS (4)
G.711.0 – G.711.1	50-4 000	40-320	6 400	add	Free	
G.722 – G.722.1 – G.722.2	50-7 000	320	6 400	add	Free	
G.723.1			5 400	add	Free	
G.726			16 000	add	Free	
G.728			16 000	add	Free	
G.729	50-4 000		8 000	add	Free	
G.729.1	50-4 000		8 000	add	Free	
Opus	Below 3 000		6 000	add	Free	Poor

(1) The audio bandwidth for analogue FM is 300-3 000 Hz.

(2) CMX7262 and CMX618/638 are commercially available as chip devices in volume. In this case there is no additional licence fee.

(3) AMBE+2 is available as both low-volume commercial chip devices or as a software package for popular micro controllers under a licensing agreement.

(4) POTS = "Plain Old Telephone System".

(5) Noise Reduction: Voice codecs that use a model of the human vocal tract as the basis for their algorithm have an inherent facility to reject background noise and other non-voice signals (this includes AMEBE+2, ACELP, RALCWI, TWEWLP and OPUS). Audio codecs that convert an analogue signal (which could be voice, music, tones etc) do not have this facility and so could require additional noise reduction (this includes most of the ITU G.7xx codecs).

The ability of a radio channel to transfer a significant number of bits whilst still remaining within the spectrum mask defined for its channel is limited. In the case of AIS, one 25 kHz marine channel can only transfer 9 600 bit/s.

By using a multi-level modulation scheme, such as 4-FSK, this can be doubled to approximately 19 200 in a 25 kHz channel and by extrapolation, 9 600 bit/s in a 12.5 kHz channel and 4 800 bit/s in a 6.25 kHz channel. This is the over-air bit rate, and to get to the bit rate available to the vocoder, both the signalling overhead and forward error correction must be deducted – generally these would account for about half the over-air rate to ensure a robust and usable system, so the maximum bit rates available to the vocoder is effectively 2 400 bit/s per voice channel. This basic engineering fact then rules out a significant number of the vocoders detailed here. 4-FSK can be received and transmitted by existing VHF FM radio modulators, PA’s and demodulators with only minor adjustments to filtering.

Using even higher level modulation schemes, such as 8-PSK or 16-QAM, would allow higher bit rate vocoders to be used. However, the hardware to implement these schemes requires linear transmitters and receivers with much more complex decoders and so are significantly more expensive, power hungry and with reduced receiver sensitivity.

9 Overview existing digital radio protocols

There are multiple digital radio protocols in use worldwide, some of which are noted below, for different purpose and goals, most of them proprietary.

9.1 Digital private mobile radio

The dPMR memorandum of understanding was established in February 2007 to develop an open, non-proprietary European Union (EU) standard for 6.25 kHz channel audio protocol (dPMR Association, 2012). It is published by ETSI under the reference License-free version (ETSI TS 102 490, 2013-02) and Licensed version (ETSI TS 102 658).

TABLE 2

Technical specifications of digital private mobile radio

Access method	FDMA
Channel spacing	6.25 kHz
Transmission rate	4 800 bit/s
Modulation	4-level FSK
Vocoder	AMBE+2
Codec rate	3 600 (Voice 2 450 + Error Correction 1 150 bit/s)

Protocol defines four modes. Tier1 – License Exempt dPMR or dPMR446. Tier 2: Licensed dPMR Mode 1 for operations without repeater, Licensed dPMR Mode 2 for operations with repeater and Licensed dPMR Mode 3 for multi-site, multi-channel trunked repeaters.

dPMR classifies itself as “specifically target highly functional, spectrum efficient solutions employing proven, low cost and low complexity”.

Next generation digital narrowband (NXDN) and dPMR are similar protocols but are not compatible. dPMR is widely known in many of the world’s markets, and it is documented as an ETSI standard.

9.2 Next generation digital narrowband

Icom and JVC KENWOOD began the collaboration in 2003 to develop protocol to provide voice and/or data at 6.25 kHz or an "equivalent" bandwidth. Following the announcement of the NXDN protocol in 2005, the NXDN forum was established in July 2008, (Next Generation Digital Narrowband (NXDN) ™ Forum, 2014).

TABLE 3

Technical specifications of next generation digital narrowband

Access Method	FDMA
Channel spacing	6.25 kHz or 12.5 kHz
Transmission rate	4 800 bit/s (at 6.25 kHz channel)
Modulation	4 l FSK
Vocoder	AMBE+2, 3 600 bit/s (at 6.25 kHz channel)
Codec rate	3 600 bit/s (Voice 2 450 bit/s + Error Correction 1 150 bit/s)

Protocol defines three levels: Conventional (with and without repeater); Type-C trunking (with control channel) and Type-D trunking (without control channel).

NXDN protocol documentation is downloadable from NXDN forum.

9.3 Digital mobile radio

Digital communication standard to allow easy migration from existing 12.5 kHz narrowband FM analogue channel. 12.5 kHz channel size allows re-use of existing frequency licenses and site infrastructure and doubles network capacity (6.25 kHz channel efficiency) (ETSI TR 102 398).

DMR is non-proprietary EU standard. It is first published in 2005 by ETSI under the reference TS 102 361-1 ETSI TS 102 361-1.

TABLE 4

Technical specifications of digital mobile radio

Access method	2 slot TDMA
Channel spacing	12.5 kHz
Transmission rate	9 600 bit/s (symbol rate of 4 800 symbols/sec)
Modulation	4-FSK
Vocoder	AMBE+2
Codec rate	3 600 bit/s (Voice 2 450 bit/s + Error Correction 1 150 bit/s)

DMR defines three tiers: Tier-I – license free 446 MHz operation, power limited to 0.5 W; Tier-II – direct replacement for analogue conventional radio, can be used with repeaters; Tier-III – trunking mode and data services.

9.4 Trans-European trunked radio system

TETRA is European standard for trunked radio system, designed for government agencies, emergency services and public safety networks. TETRA is a European version of trunked radio similar to Project 25.

Tetra is EU open standard, published by ETSI under the reference ETS 300.392, ETSI TS 300 396-1, ETSI TS 100 392-2.

TABLE 5

Technical specifications of trans-European trunked radio system

Access method	4 slot TDMA
Channel spacing	25 kHz
Transmission rate	7.2 kbit/s per timeslot
Modulation	π/4 DQPSK
Vocoder	ACELP 4.567 kbit/s
Codec rate	7 200 bit/s (Voice 4 567 bit/s + Error Correction, 2 633 bit/s)

TETRA system works mainly as a trunking mobile radio network, but it is possible to use TETRA radio as conventional radio (direct mode operation mode).

9.5 Comparison of digital radio protocols

TABLE 6

Comparison of digital radio protocols

Protocols	Over-air rate (bit/s)	Access method	Bit rate for vocoder (bit/s)	Modulation	Vocoder	Channel spacing (kHz)
dPMR	4 800	FDMA	3 600	4 FSK	AMBE+2	6.25
NXDN	4 800	FDMA	3 600	4 FSK	AMBE+2	6.25
DMR	9 600	TDMA 2 slot	3 600	4 FSK	AMBE+2	12.5
TETRA	7 200	TDMA 4 slot	7 200	π/4 DQPSK	ACELP	25

9.6 Other issues raised

The Ministry of Infrastructure and Water Management (Rijkswaterstaat) in the Netherlands have asked about the ability of dPMR to support ATIS operation on Inland Waterways – the significant difference between dPMR and marine operations is the use of the MMSI in marine operations which demands the use of a 30-bit addressing field, whereas dPMR today can only support 24-bits. ETSI ERM TG MARINE have recognised that there are significant differences between the investigated technologies and marine operations, and have already been working on the modifications to the dPMR addressing protocol to support 30-bit modes, which should make it suitable for both MMSI and ATIS addressing modes.

It was recognised that support of legacy functionality is needed during the full migration period.

All radio protocols listed will require some adaption to comply with requirements of the maritime environment.

10 Implementing digital voice in marine VHF frequency bands

The introduction of narrowband (6.25 kHz) channelisation should be considered to support digital voice in the marine VHF frequency band. Principally to alleviate the current over-crowding issues posed by both the increase in marine radio traffic generally and the reduction in available RF spectrum for voice communication following the introduction of AIS, ASM and VDE data transmissions. The principal advantage of the digital voice system is that it allows up to four digital voice channels to be available in the same spectrum as one current analogue voice channel. Recommendation ITU-R M.1084-5 suggests the basis of a channel plan that has to be adjusted to support four subchannels in a 25 kHz channel, but rolling this out over the RF spectrum allocated to marine VHF communications across the world whilst still keeping essential emergency and safety services available is a complicated task, especially in a market where equipment may only get replaced or updated after ten years or more in service.

There are both technical, operational and logistical challenges to be considered, from the performance of cost-effective (affordable) radio equipment as well as how best to integrate it with the existing systems. The marine market is significantly more complex due to its world-wide nature, the requirement to maintain safety critical services at all times, the number of units involved (both commercial and leisure markets probably amount to over 10 million units) and the low cost of the current analogue solutions (in comparison to other digital technologies such as cell phones, TETRA and DMR radios, for instance).

10.1 Possible scenarios

In order of increasing complexity, the possible scenarios[1] considered here are:

– interleave channels;

– re-farm a specific block;

– re-farm the entire marine VHF frequency band except for the GMDSS;

– re-farm the entire marine VHF frequency band including the GMDSS.

In the first three cases, it is assumed that new VHF radio equipment will be capable of “dual” mode operation, i.e. it will operate in either analogue mode or digital mode as instructed by the user based on the channel selected. The channel numbering scheme (based on Recommendation ITU-R M.1084) determines if the channel is digital or analogue and would be published in the same way that the current analogue-only channel allocations are promulgated to current users.

In the final case, an all-digital solution would be feasible, but only after the whole network has migrated to digital and would then provide scope for savings in equipment design and cost. However, given the longevity of existing equipment and the massive task to migrate the entire world to solely digital operation, this would only be likely to become possible in decades to come.

For the sake of this Report, the term GMDSS refers to those channels reserved in the Radio Regulations for dedicated services as described in Report ITU-R M.2231. The channels listed below are dedicated to special purposes, some of which are safety services and others are noted in SOLAS Chapters 4 and 5, the remaining channels in the list are calling channels used by all ships and shore stations. These channels below should possibly remain reserved in the near term to ensure compatibility with current services:

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[1] There are likely to be other scenarios depending on the circumstances of any given radio system.

– Channel 06;
– Channel 16 distress and calling;
– Channel 13;
– Channel 15;
– Channel 17;
– Channel 70 DSC signalling;
– Channel 75;
– Channel 76.

The AIS, ASM and VDES (in accordance with footnote *w)* of RR Appendix **18**) channels are already allocated for data services and so cannot be allocated for digital voice.

10.2 Inter-leaving channels

Recommendation ITU-R M.1084 shows a possible "inter-leaved" channel plan where the new digital channels sit in between the existing analogue channels. In this case, it is assumed that the digital modes could operate in tandem with the existing analogue and be used as and when needed or appropriate and so seamlessly increase the number of channels available by making use of the "gaps" in between the existing analogue channels. Technical constraints on the design of radio equipment make this scenario very difficult to achieve without compromising the performance of one or both systems, especially if the transmitters are expected to be close or co-located. With suitable physical separation between transmitters, (probably around a mile or so – this needs to be tested by trials) there may still be situations where this is feasible, but it would not be a universal solution.

Effect on GMDSS: none.

10.3 Re-farm a specific block

In a similar manner that ITU/IMO have allocated blocks of RF spectrum for data use, it could make a channel block available for digital voice. In the first instance, digital voice would have to share the spectrum with any incumbent analogue services, but over time as the digital service become more widely used, they could be elevated to primary usage, and ultimately sole usage. To be most efficient, the block should be a continuous multiple of 25 kHz, as the current Recommendation ITU-R M.1084 numbering scheme allows for a small guard band at the edge of each channel block, so a single 25 kHz analogue channel would only support three digital channels, whereas two contiguous 25 kHz channels would provide seven digitals channels and four contiguous analogue 25 kHz channels would yield 15 digital channels.

NOTE – This is a consequence of the numbering scheme adopted by Recommendation ITU-R M.1084 which allows for interleaved channels. If inter-leaving is not required, there is no technical reason why a direct four digital to one analogue channel transition cannot be supported.

One variation on this option would be to concentrate on moving port operations and VTS services onto digital first, as many of the stations using these channels tend to remain "local" to the area and so can be more easily monitored and controlled by the local administration. This would then leave the analogue channels for stations that operate internationally, even those that do not have digital capability.

Effect on GMDSS: none.

10.4 Re-farm the entire maritime VHF frequency band except for the GMDSS

This is continuation from the earlier scenario, as more and more stations become equipped with digital-compatible equipment, then it may become feasible to extend the specific digital voice blocks to cover more of the spectrum and reduce the analogue allocation to solely those channels used for safety critical operations. This may be difficult to achieve in some geographic areas due to the re-allocation of parts of the maritime VHF frequency band to support other services or private operators.

Effect on GMDSS: none.

10.5 Re-farm the entire maritime VHF frequency band including the GMDSS

This is a continuation from the earlier scenario but would now include digitising the operation of the GMDSS services. The two schemes would need to operate in tandem during the switch-over period which could cause some confusion, so would need to be carefully managed. DSC operation could remain unchanged during the switch-over, but a digital equivalent message would be transmitted following the DSC signalling. For distress calls made on channel 16, it would be technically feasible for the radio to record the initial distress call (made either on analogue or digital) and then repeat it automatically in the other mode, either on the same channel or an alternative one. However, this would consume precious time which may not be available.

Effect on GMDSS: significant.

10.6 Extension of Recommendation ITU-R M.1084 numbering

Interim solutions for improved efficiency in the use of the frequency band 156-174 MHz by stations in the maritime mobile service will not support full channel indications as required for addressing the 6.25 kHz channels. After studying the current Recommendation ITU-R M.1084, the proposal to add a new Annex to it was made.

10.6.1 Proposed new Annex to Recommendation ITU-R M.1084-5 – Assignment of channel numbers for digital voice communications on 25 kHz channels in the VHF maritime frequency band

1 Channel number assignments for implementing digital voice communications with 6.25 kHz channel spacing between the four subchannels on 25 kHz channels in the VHF maritime frequency band should be in accordance with Table 7.

2 The channel centre frequency retains its channel number in the VHF maritime frequency band, and it is used exclusively 25 kHz voice communications. Digital voice communications should not be used when the 25 kHz channel is in its normal 25 kHz use.

TABLE 7

Example of channel number assignments with 6.25 kHz channel spacing

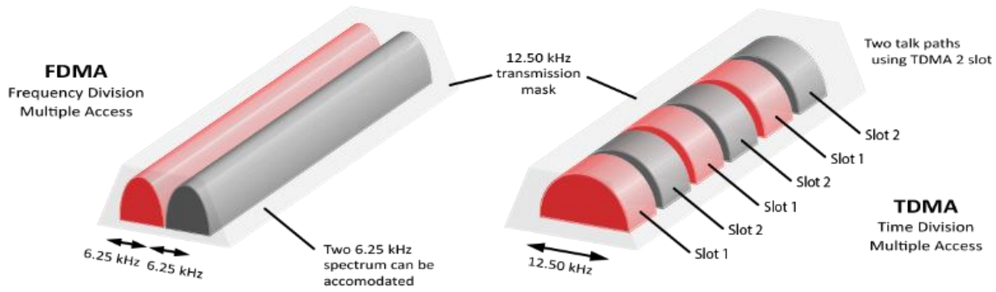
Channel No. (6.25 kHz spacing between subchannels)					Ship	Ship and coast	Coast
		801			156.059375		160.659375
	701				156.053125		160.653125
01					156.050		160.650
	601				156.046875		160.646875
		501			156.040625		160.640625

11 Summary of existing digital radio protocols

Current maritime VHF radio is inherently FDMA in nature, one radio channel carries one voice channel. TETRA and DMR are TDMA systems in which one radio carrier can carry multiple voice channels, however the timing of these is critical to its operation and so is limited to systems where there is a “master” transmitter that can define the slot timing accurately for all units in the network. Whilst this may be feasible close to coast, clearly this is not possible in the high seas or areas not covered by a coast station.

FIGURE 5

Frequency division multiple access vs. Time division multiple access



TETRA is designed for large public safety networks and requires expensive infrastructure and although its spectrum efficiency meets the “6.25 kHz per voice channel” requirement, its deployment requires 25 kHz channelization and its TDMA nature make it unsuitable for the maritime environment.

DMR is designed to replace 12.5 kHz analogue commercial channels and does meet the “6.25 kHz per voice channel” requirement, but, like TETRA, its TDMA nature makes it unsuitable for maritime use.

dPMR and NXDN suits well to replace analogue audio in maritime environment. They both provide:

– standardised minimal cost digital radio solution, working on 6.25 kHz channels;

– group call and individual call;

– short data transmission;

– security Services (location and status transmission);

– coexistence with analogue audio.

They both claim: improved audio quality in weak signal conditions; better range performance (this is taken to mean a good quality of service out to the range boundary rather than much greater absolute range).

NXDN uses 16 bit user and 16 bit group address space. dPMR uses 24 bit address, both need modification to support 32 bit MMSI address as radio ID. Both are using AMBE+2 codec. This codec is proprietary, usage needs license from Digital Voice Systems, Inc.

dPMR is ETSI open standard, NXDN is standard published by NXDN forum.

Technically, the standards are comparable and offer similar functionality. However, the dPMR published as open ETSI standard is more preferred.

12 Possible advice for digital public mobile radio marine

ETSI has published a Technical Report (ETSI TR 103 784)[2] for using digital voice calls in the marine VHF frequency band. It is based on the modified dPMR protocol to use 32 bit address space instead 24 bit. (ETSI TS 102 658, 2019-01; ETSI, 2019).

The following sections are a short summary about changes to TS 102 658 and comments about technical details.

12.1 Proposed changes to ETSI TS 102 658

12.2 Address field related changes

For 32 bit address support frame encoding is changed. Total frame length is the same, but some fields are changed:

– “Frame Number” – unchanged;

– “Called ID (lower 16 bits)” – changed from 12 bits to 16;

– “Communications mode” – unchanged;

– “Category” – new field, 2 bit long;

– “Version”, “Comms format”, “Reserved” – fields removed, total 6 bits.

Similar changes are in Header frame and ACK frame content. Address fields are changed, added “category” and “Version”, “Comms format”, “Reserved” fields are removed. Total frame size is unchanged.

12.3 “Communication mode” field related changes

“Communication mode” field defines following:

– voice communication + slow data (slow data contains position latitude and longitude);

– Data Communication Type 2 (user data with forward error correction (FEC));

– Data Communication Type 2 (user data with FEC) + Appended data;

– reserved for future use.

12.4 “Category” field related changes

New “Category” field defines the order of priority of communications according to Article **53** of the RR:

– distress;

– urgency;

– safety;

– other communications.

13 Summary assessment of the options noted in this report

This Report on the digitisation of the analogue maritime VHF voice channels and further developments will be closely monitored. This ITU-R Report could potentially be the base for future discussions in ITU and IMO.

[2] Digital VHF Maritime Radio; Air interface for voice and data services using FDMA in 6.25 kHz bandwidth.

The intention of this work was to investigate the possible expansion of the number of VHF voice channels based on the implementation of digital technology. A plan for change over from analogue to digital will be required.

Analysis observations:

– Current maritime VHF radio is inherently FDMA in nature, one radio channel carries one voice channel. TETRA and DMR are TDMA systems in which one radio carrier can carry multiple voice channels, however the timing of these is critical to its operation and so is limited to systems where there is a “master” transmitter that can define the slot timing accurately for all units in the network. Whilst this may be feasible close to coast, clearly this is not possible in the high seas or areas not covered by a coast station.
– dPMR and NXDN suit well to replace analogue audio in maritime environment. Technically, the standards are comparable and offer similar functionality. However, the dPMR published as open ETSI standard and therefor is preferred.

Two test trials were carried out by two ITU member states. Full reports of their trials can be found in Annex 2 and Annex 3.

Further observation on this subject:

– the Digital channelling arrangements need to fit within the existing analogue channel plan. (Compatible with Recommendation ITU-R M.1084);
– practical trials undertaken in Estonia and the Netherlands demonstrate that digital systems operated successfully in coexistence with analogue systems in these locations;
– similar technologies exist already and are used in other communities but need to have minor modifications for the use in the maritime community;
– analogue and digital channels may need to exist in parallel during a migration period and some analogue channels may need to be retained in perpetuity.

Annex 1

Report from CYBERNETICA
Analyses of different digital radio protocols for use in maritime communication

A1.1 IALA recommendations

IALA in their Liaison Note on 04 April 2019 stated following recommendations (IALA ENAV Committee, 2019-04):

– 6.25 kHz should be used (one 25 kHz channel divided into for channels);
– new networks should support Short Messages (SMS);
– new network should support transmitting radios location duration voice call;
– digital voice quality should be similar to or better than analogue voice.

A1.2 Transceivers addressing

In digital networks radio stations normally have Identity. In maritime digital communication (DSC) there is already ID in use – MMSI number. Same number is used by AIS network. It is preferable when same ID can be used in new digital network.

Overview of existing digital radio protocols: There are multiple digital radio protocols in use worldwide for different purposes and goals, most of them proprietary.

A1.3 Digital private mobile radio

The dPMR memorandum of understanding was established in February, 2007 to develop open, non-proprietary EU standard for 6.25 kHz channel audio protocol (dPMR Association , 2012). It is published by ETSI under the reference License-free version (ETSI TS 102 490, 2013-02) and Licensed version ETSI TS 102 658.

Technical specifications of dPMR (dPMR Association , 2012):

– Access method: FDMA;

– Transmission rate: 4 800 bit/s;

– Modulation: 4-level FSK;

– Vocoder: AMBE+2;

– Codec rate: 3 600 bit/s (Voice 2 450 bit/s + Error Correction 1 150 bit/s).

Protocol defines four modes. Tier1 – License Exempt dPMR or dPMR446. Tier 2: Licensed dPMR Mode 1 for operations without repeater, Licensed dPMR Mode 2 for operations with repeater and Licensed dPMR Mode 3 for multi-site, multi-channel trunked repeaters.

dPMR classifies itself as “specifically target highly functional, spectrum efficient solutions employing proven, low cost and low complexity”.

NXDN and dPMR is similar protocols but not compatible. dMPR is not known in US market.

A1.4 Next generation digital narrowband

Icom and JVC KENWOOD began the collaboration in 2003 to develop protocol to provide voice and/or data at 6.25 kHz or an “equivalent” bandwidth. The announcement of the NXDN protocol in 2005, the NXDN forum was established in July, 2008 (dPMR Association , 2012), (NXDN ™ Forum, 2014).

TABLE 8

Technical specifications of next generation digital narrowband

Access Method	FDMA
Vocoder	AMBE+2, 3 600 bit/s (at 6.25 kHz channel)
Channel Spacing	6.25 kHz or 12.5 kHz
Transmission Rate	4 800 bit/s (at 6.25 kHz channel)
Codec Rate	3 600 bit/s (Voice 2 450 bit/s + Error Correction 1 150 bit/s)

Protocol defines three levels: Conventional (with and without repeater); Type-C trunking (with control channel) and Type-D trunking (without control channel)

NXDN protocol documentation is downloadable from NXDN forum.

A1.5 Digital mobile radio

Digital communication standard to allow easy migration from existing 12.5 kHz narrowband FM analogue channel. 12.5 kHz channel size allows re-use of existing frequency licenses and site infrastructure and doubles network capacity (6.25 kHz channel efficiency) ETSI TR 102 398.

DMR is non-proprietary EU standard. It is first published 2005 by ETSI under the reference TS 102 361-1 - TS 102 361-4 (ETSI TS 102 361-1, 2006-01).

TABLE 9

Technical specifications of digital mobile radio

Transmission Rate	9 600 bit/s (symbol rate of 4 800 symbols/s)
Modulation	4-level FSK
Access Method	2 slot TDMA
Modulation	4-level FSK Modulation
Vocoder	AMBE+2
Channel Spacing	12.5 kHz

DMR defines three tiers: Tier-I – license free 446 MHz operation, power limited to 0.5 W; Tier-II – direct replacement for analogue conventional radio, can be used with repeaters; Tier-III – trunking mode and data services.

A1.6 Trans-European trunked radio system

TETRA is European standard for trunked radio system, designed for government agencies, emergency services and public safety networks. TETRA is European version of trunked radio similar to Project 25.

Tetra is EU open standard, published by ETSI under the reference ETS 300.392 - ETS 300.396 (ETSI TS 300 396-1, ETSI TS 100 392-2).

TABLE 10

Technical specifications of Trans-European trunked radio system

Transmission rate	7.2 kbit/s per timeslot
Modulation	π/4 DQPSK
Access Method	4 slot TDMA
Vocoder	ACELP 4.567 kbit/s
Channel Spacing	25 kHz

TETRA system works mainly as a trunking mobile radio network, but it is possible to use TETRA radio as conventional radio (PTT mode).

A1.7 Summary

TETRA is designed for large networks and requires expensive infrastructure and does not provide the 6.25 kHz channel requirement.

DMR is meant to replace 12.5 kHz analogue channel and so does not allow to make separate 6.25 kHz channels. Lot of maritime audio traffic is broadcast (or all vessel), so TDMA does not offer advantages here.

dPMR and NXDN are well suited to replace analogue audio in maritime environment. They both provide:

– standardised minimal cost digital radio solution, working on 6.25 kHz channels;
– group call and individual call;
– short data transmission;
– security Services (location and status transmission);
– coexistence with analogue audio.

They both claim: improved audio quality in weak signal conditions; better range performance (this is taken to mean a good quality of service out to the range boundary rather than much greater absolute range).

NXDN uses 16 bit user and 16 bit group address space. dPMR uses 24 bit address, both need modification to support 32 bit MMSI address as radio ID. Both are using AMBE+2 codec. This codec is proprietary, usage needs license from Digital Voice Systems, Inc.

dPMR is ETSI open standard, NXDN is standard published by NXDN forum.

Technically, the standards are comparable and offer similar functionality. However, the dPMR published as open ETSI standard is more preferred.

A1.7.1 Digital private mobile radio marine

ETSI has proposed draft document for using digital voice for Routine category calls in the marine VHF frequency band. It is based modified dPMR protocol to use 32 bit address space instead 24 bit.

Last documents version is V 1.2.0 created 09.04.2019. Document is based license free dPMR ETSI TS 102 490.

Next chapters are short summary about changes from TS 102 490 and comments about technical details.

A1.7.2 Proposed changes from ETSI TS 102 490

For 32 bit address support frame encoding is changed. Total frame length is same, but fields are changed.

TABLE 11

Address field related changes

Frame Number	Unchanged
Called ID (lower 16 bits)	changed from 12 bits to 16
Communications mode	Unchanged
Category	new filed, 2 bit long
Version	“Comms format”, “Reserved” – fields removed, total 6 bits

Similar changes are in Header frame and ACK frame content. Address fields are changed, added “category” and “Version”, “Comms format”, “Reserved” fields are removed. Total frame size is unchanged.

Comment: It remains unclear whether this protocol sufficiently “future proof” when there are no “Version” field.

“Communication mode” field related changes

The “Communication mode” field defines the settings as set out in Table 12.

TABLE 12

Communications mode field related changes

Voice communication	No user data in SLD field
Voice + slow data	Position data in SLD field
Voice and appended data	Type 2
Data communication type 2	Payload is user data with FEC

“Category” field related changes

TABLE 13

Category field related changes

New “Category” field defines:
Routine
Safety
Urgency
Distress

Channel Code

ETSI TS 102 658 Chapter 6.1.5 Channel Code describes algorithm to calculate 24 bit value of “channel code” from channels centre frequency.

Algorithm gives integer value from 0-63.

Comment: Recommendation ITU-R M.1084 recommends how to assign channel number when 25 kHz channels are divided 12.5 or 6.25 kHz sub-channels. There is an example for channel 01 and surrounding 60 and 61 channels:

TABLE 14

Channel frequency

Channel number	Ship frequency	Coast frequency
60	156.025	160.625
160	156.03125	160.63125
260	156.0375	160.6375
360	156.04375	160.64375
01	156.050	160.650

TABLE 14 *(end)*

Channel number	Ship frequency	Coast frequency
101	156.05625	160.65625
201	156.0625	160.6625
301	156.06875	160.66875
61	156.075	160.675

An open question remains on whether Channel Code be generated from ITU-R M.1084 channel number.

Call types

It should be explained how DSC (Recommendation ITU-R M.493-15, 2019) calls should be used to make call dPMR channel. There are defined "Individual, Routine, Data" for individual call and no information about group calls (the Recommendation does not specify Data telecommand for group call).

Frequency field should be proposed 6.25 kHz channel.

Comment: It remains unclear whether audio call be "All mode RT" (technically, it is digital data), or whether future version of Recommendation ITU-R M.493 should add "digital RT" option. Channel numbers should be as in Recommendation ITU-R M.1084 channel number. It is open whether this channel number is related to "Channel Code".

Subscriber mapping

It should be explained how to use MMSI address as radio protocols address field. There are defined that address can be individual or group MMSI address.

Comment: This chapter refers to Recommendation ITU-R M.585-7.

In maritime communication, most calls are "All Call" (s.t broadcasted to all listeners in current channel).

Recommendation ITU-R M.585-7 does not define "All Call" MMSI. There is MMSI address 009990000 to address all coastal stations. For DSC "All ships" calls in ITU-R M.493-15 are defined special messages, without destination MMSI address.

One option to consider is whether some legal MMSI address (for example 000000000) should be selected and documented in the future versions of ITU-R M.585-7.

Annex 2

Report from Estonia

A2.1 Summary of the maritime communication test

Previous measurements

Sensitivity of transceivers:

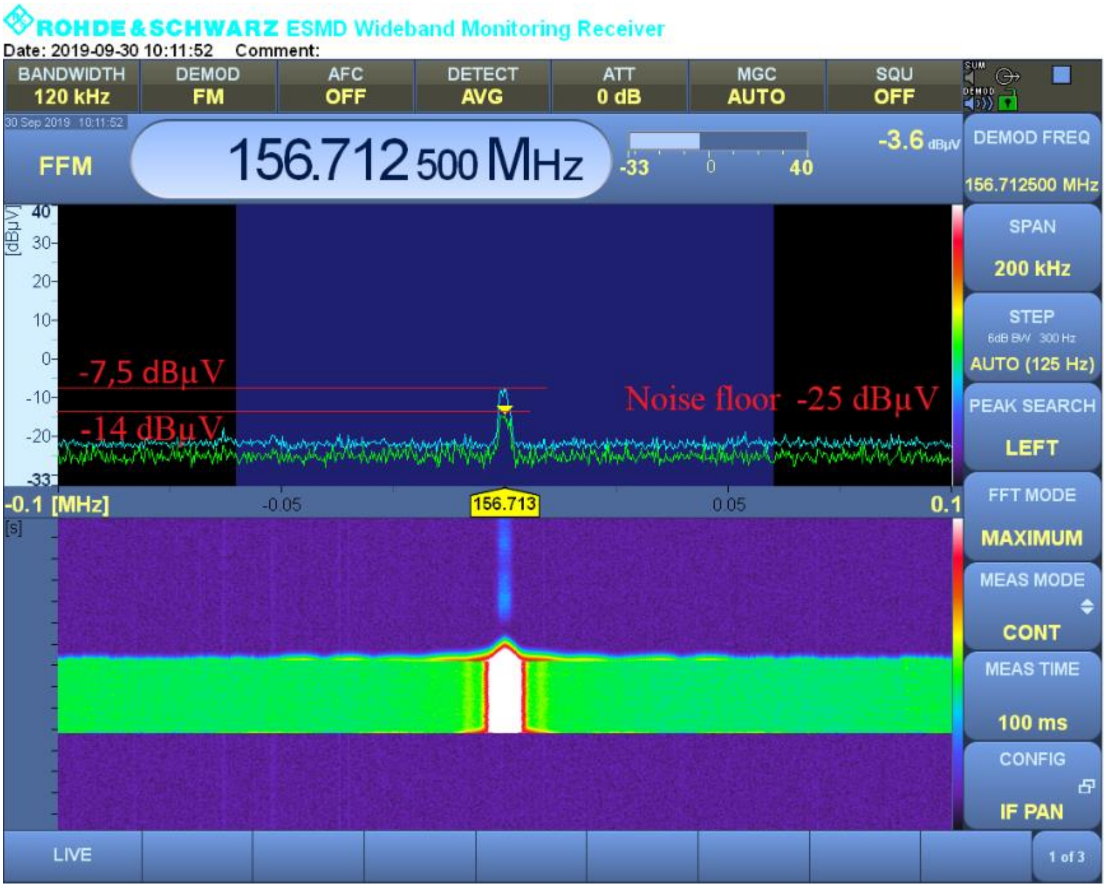
– AT 20 dB SINAD 4 dB µV;

– AT 5% BER 8 dB µV.

Figure 6 describes a signal for which the perceived quality of the communication is 2, i.e. interrupted and poorly understood. The data of the manufacturer and the measured signal are consistent. $V_v = -7.5... 14$ dB µV (at speed $V = 20$ Knots).

FIGURE 6

Signal voltage at the receiver inlet as the perceived quality of the communication is 2



The instrument’s noise is −25 dB µV. It is not possible to estimate how much the received radio transmitter signal must be more powerful than the noise signal. The actual noise signal strength is unknown.

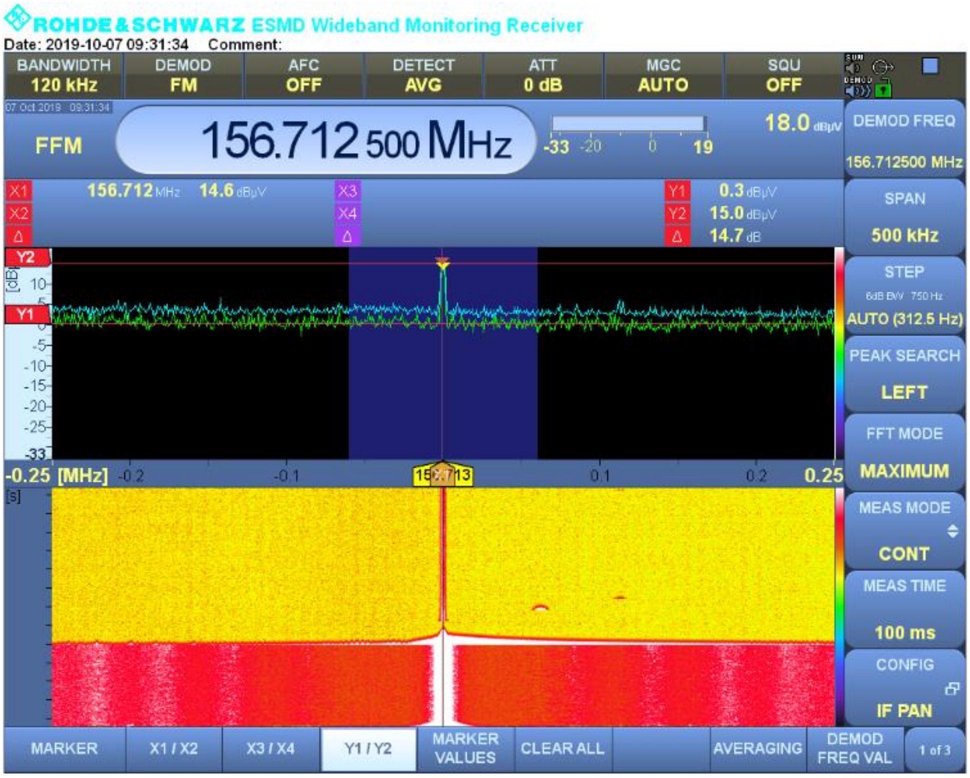
An additional measurement was performed. The receiver was introduced into a noisy environment and an approximate signal of noise in relation to the interruption of the communication was checked. Measurement locations are shown in Fig. 10 (the source of noise is a LED screen mounted on the Shopping Centre).

FIGURE 7

Measurement location for assessing environment with high radio interference

FIGURE 8

Signal vs interference level during signal interruption situation

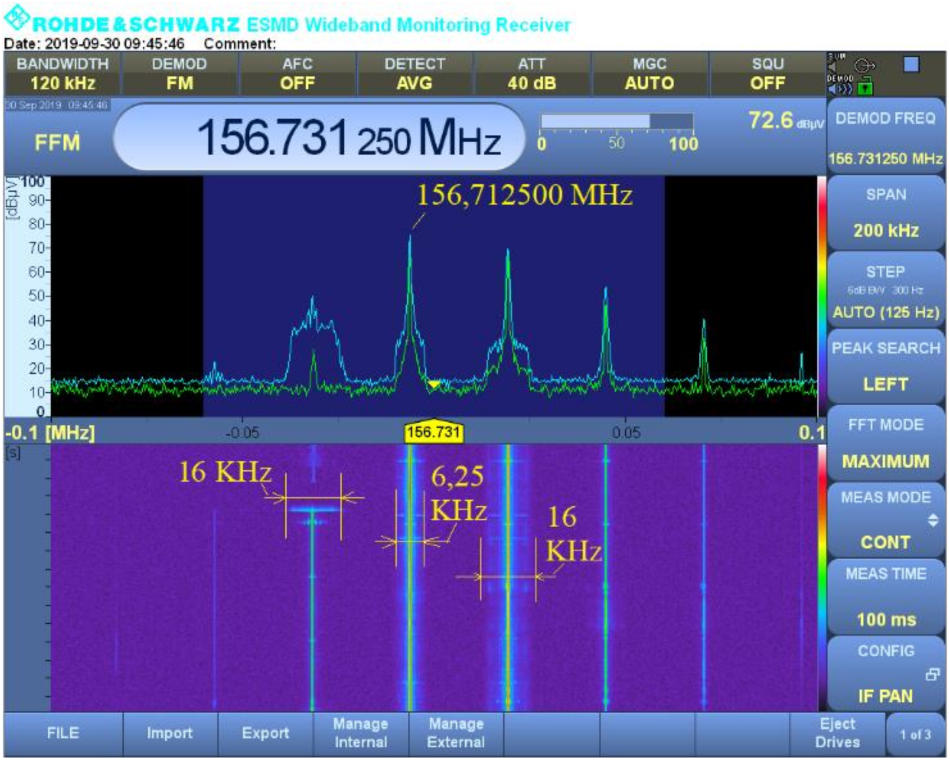


As a result of the measurement, the communication was interrupted when the signal to-noise ratio was less than SNR < 14.7 dB. The power of the signal must exceed noise by more than 30 times.

The measurement was carried out on the 214th channel (156.7125 MHz).

FIGURE 9

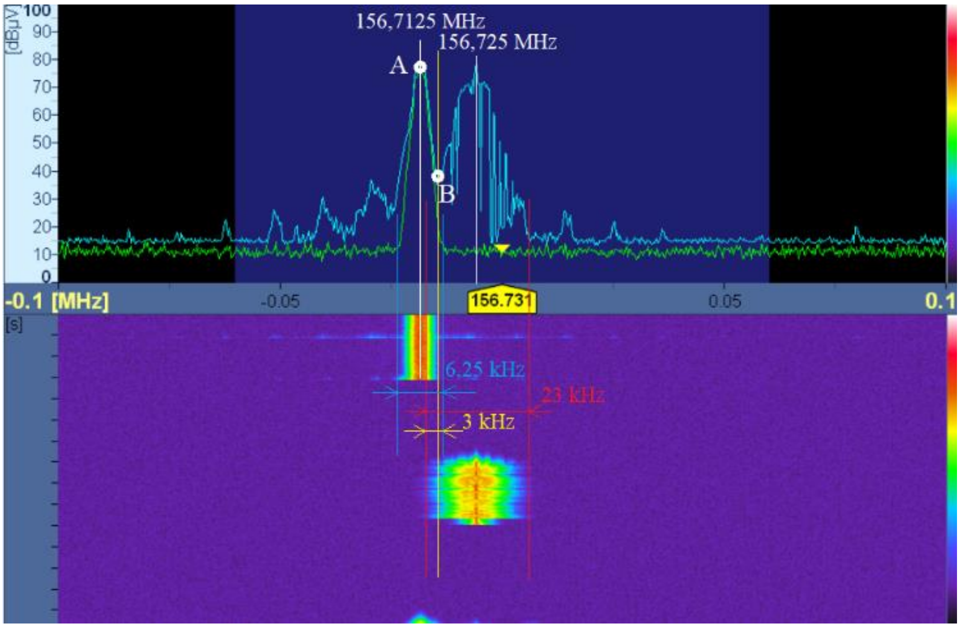
6.25 kHz channel spacing between two 25 kHz step channels as used during the transition period



The 74th analogue channel (K_s = 25 kHz) and the 214th digital channel, the frequency bands may overlap.

FIGURE 10

Example of frequency overlap ($f_{overlap}$ = 3 kHz) between 74th analogue and 214th digital channel



In the case described in Fig. 10, the digital signal at point A must be at least ΔE = 14.7 dB (µV/m) above the yellow line point B (explanation in Fig. 8 above).

The communication was carried out using the vessel on-board analogue station and temporarily installed a digital station with an additional antenna:

– the digital station installed on board the vessel: Icom IC-F5400D;

– antenna: Celwave CX4 146-162.5 MHz.

For taking measurements vertically polarised dipole antenna was fitted to the ship:

– measuring antenna Rohde & Schwarz HZ-12;

– the length of the dipole element: L = 0.913 m (f = 156 MHz), gain G = 2.15 dBi.

FIGURE 11

Antennas installed on a vessel

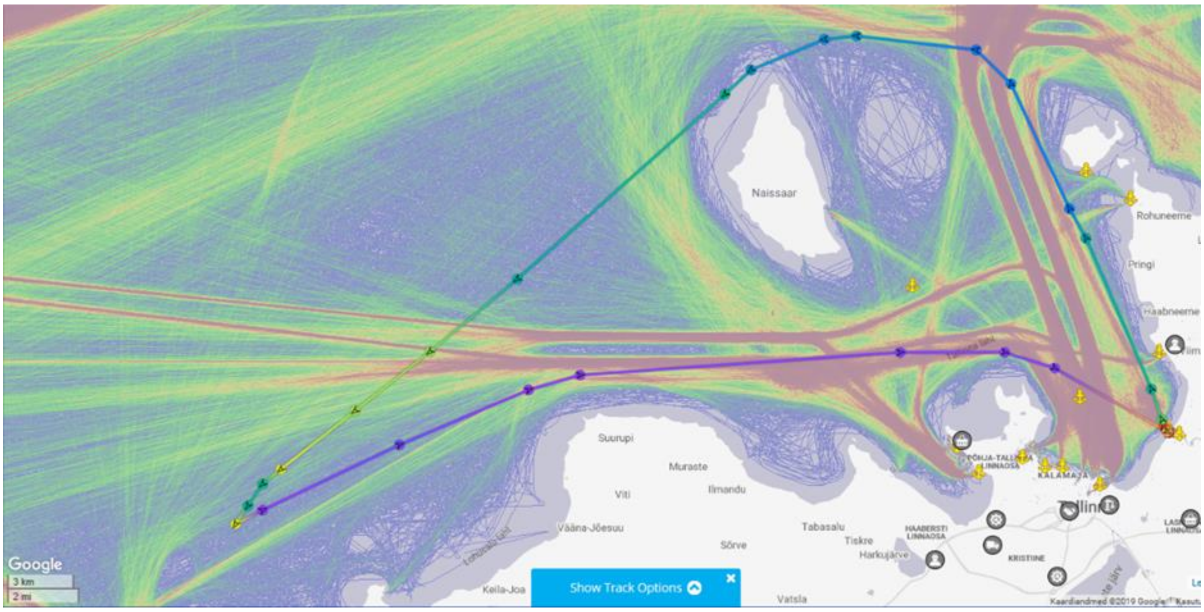
FIGURE 12

On-board measurement instrumentation Rohde & Schwarz

The distance between Pirita and Hundipea ports d = 3.2 NM (5.9 km) and the estimated quality of communication between them on a scale of 5 is 2 for analogue and 5 for digital stations.

FIGURE 13

Path of the vessel



The vessel (callsign – Jaam1) set out from Pirita along the edge of the Viimsi peninsula around Naissaar island to the direction of the Pakri peninsula. When leaving Pirita, the quality of the digital communication with Hudipea port (callsign – Jaam2) was assessed until the connection was lost. Estimates are shown on a map (Fig. 17) with green symbols. After navigating around Naissaar direct visibility (line of sight LOS) with the tip of the Pakri peninsula was achieved from a distance of 21 NM (39 km), since the tip of the peninsula is elevated 23 m above sea level. At the Pakri peninsula, in the viewpoint parking lot, there was a vehicle with a digital and analogue station (call sign – Jaam3 stationary) and two handheld stations (switchable between analogue and digital) one in the Pakri lighthouse (Jaam3 handheld-tower) and another on observation platform (plateau) (Jaam3 handheld-plateau).

TABLE 15

On the evaluation of the quality of digital communications

Digital						
Quality of communications [0... 5] Vessel – Pakri					**Coordinates**	
Distance [NM]	**Stationary**	**Handheld-tower**	**Handheld plateau**	**Handheld on vessel**	**Latitude**	**Longitude**
19.6	2		0		59°37,031 ′	24°30,303 ′
18.9	4		0		59°36,64 ′	24°29,5 ′
16.8	5	5	0		59°35,18 ′	24°26,3 ′
8.5	5	5	5	3	59°29,72 ′	24°13,85 ′
6.3	5	5	5	4	59°28,46 ′	24°10,67 ′
2.7	5	5	5	5	59°26,06 ′	24°05,64 ′
Communication between Hundipea and Pirita						
	Distance [NM]		**Handheld on vessel**			
	3.2		5		59°45,891 ′	24°71,8338 ′
Communication between Hundipea and the vessel						
	7.7	Intermittent	3		59°34,661 ′	24°42,529 ′
	9.8	No signal	0		59°37,009 ′	24°40,24 ′

TABLE 16

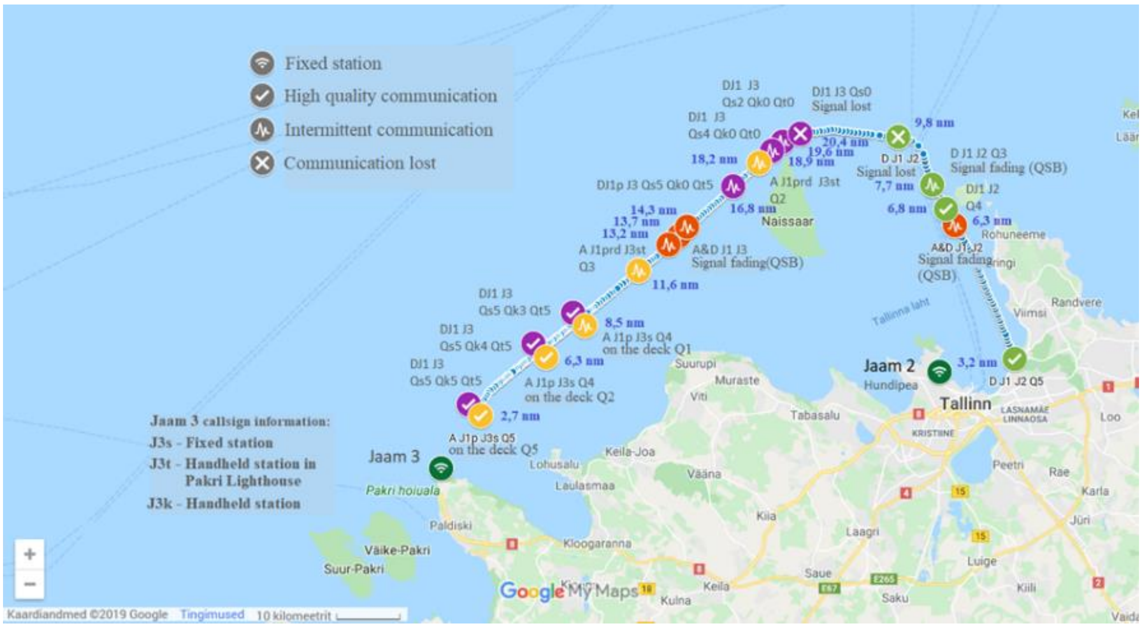
Quality of analogue communication

Analogue				
Quality of communication [Graded 0 to 5]			**Coordinates**	
Distance [NM]	**Onboard station on vessel**	**Handheld on vessel**	**Latitude**	**Longitude**
18.2	2		59°36,14 ′	24°28,4 ′
11.6	3		59°31,7 ′	24°18,59 ′
8.5	4	1	59°29,72 ′	24°13,85 ′
6.3		2	59°28,46 ′	24°10,67 ′
2.7		5	59°26,06 ′	24°05,64 ′
Communication between Hundipea and Pirita				
Distance	Handheld on board		59°45,891′	24°71,8338 ′
3.2	2			

Measurement results in the above Tables were recorded on the map (Fig. 14). Map in Google: https://www.google.com/maps/d/viewer?mid=1kaQC1VE_env6Mac5T9k2pGHsrseyaCME&ll=59.50154582929907%2C24.322173566386027&z=11

FIGURE 14

Measuring points and results on a map



Colour codes:
Green – measuring point for digital communication
Violet – measuring point for digital communication
Yellow – measuring point for analogue communication
Dark green – shore station.

After leaving Pirita harbour there were interruptions in communications between Jaam1 and Jaam2, at the distance d = 6.3 NM (Vessel – Hundipea port), and between Jaam1 and Jaam3, at a distance d = 13.4 – 14.3 NM (Vessel – Pakri fixed station). Such weakening of the signal may be caused due to the multipath propagation, as described in the measurement methodology (§ A2.3), influencing the quality of both analogue and digital communications. The waterfall diagram on measurement screenshots (Figs 15 and 16) shows the variation of signal strength (measured electric field strength) within 10 seconds. On the map, variations are marked with a zigzag symbol. At the place where the communication appears – there is no change in the colour of the symbols. Orange symbols mark places where communication was lost due to radio propagation attributes.

In the case of analogue voice, it is possible to distinguish and understand the speech even in case of a very weak signal, for example by storing and processing (using ear-muffs). The Digital signal is completely interrupted, thus the operator does not know that someone is trying to start communicating (Fig. 17). As a consequence, in one case there is the risk of integrity loss, but in the second case, the loss of availability is not guaranteed and it is also the biggest weakness of digital communication. As soon as it is possible to decode the signal, digital communication ensures significantly better understanding than analogue signals. Analogue and digital communications quality became comparable only at the last measurement point d = 2.7 NM.

FIGURE 15

Variation of the signal strength due to propagation attributes between Jaam1 and Jaam2

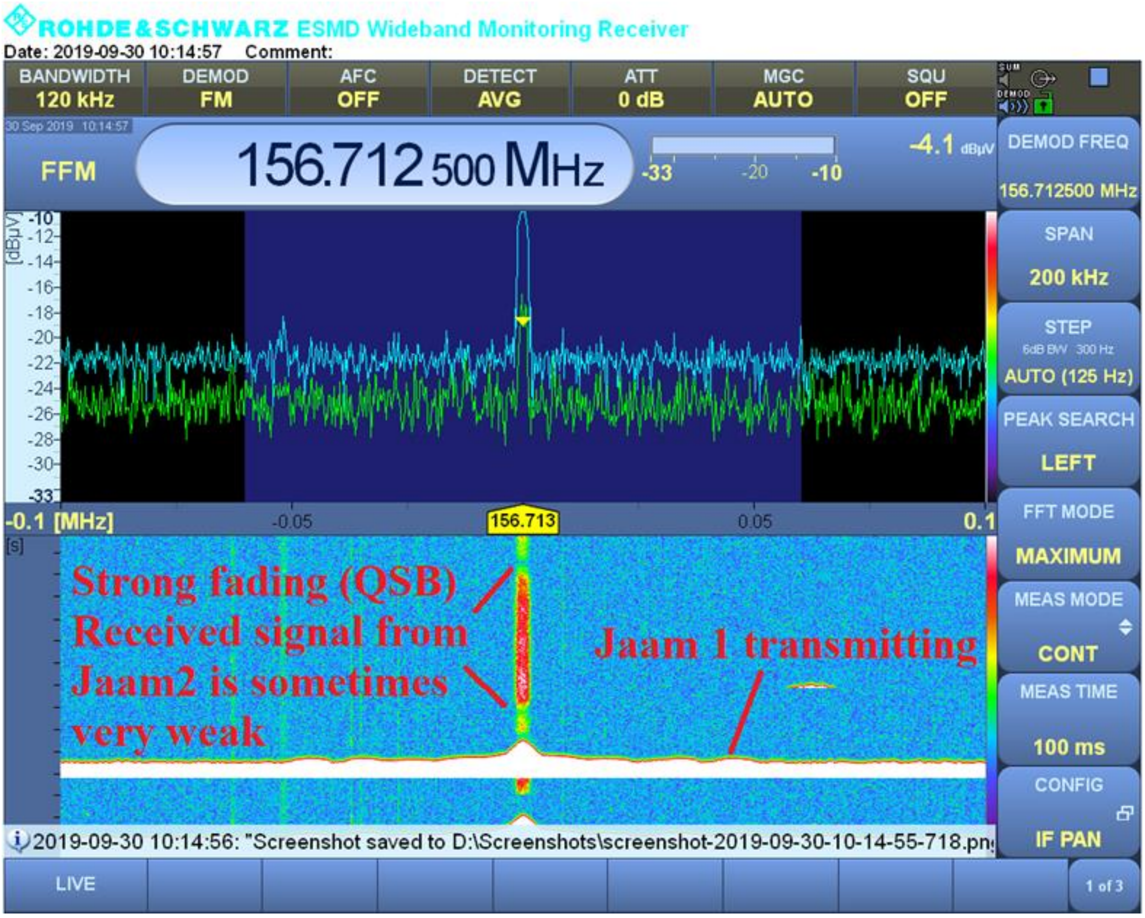


FIGURE 16

Variation of the signal strength due to propagation attributes between Jaam1 and Jaam3

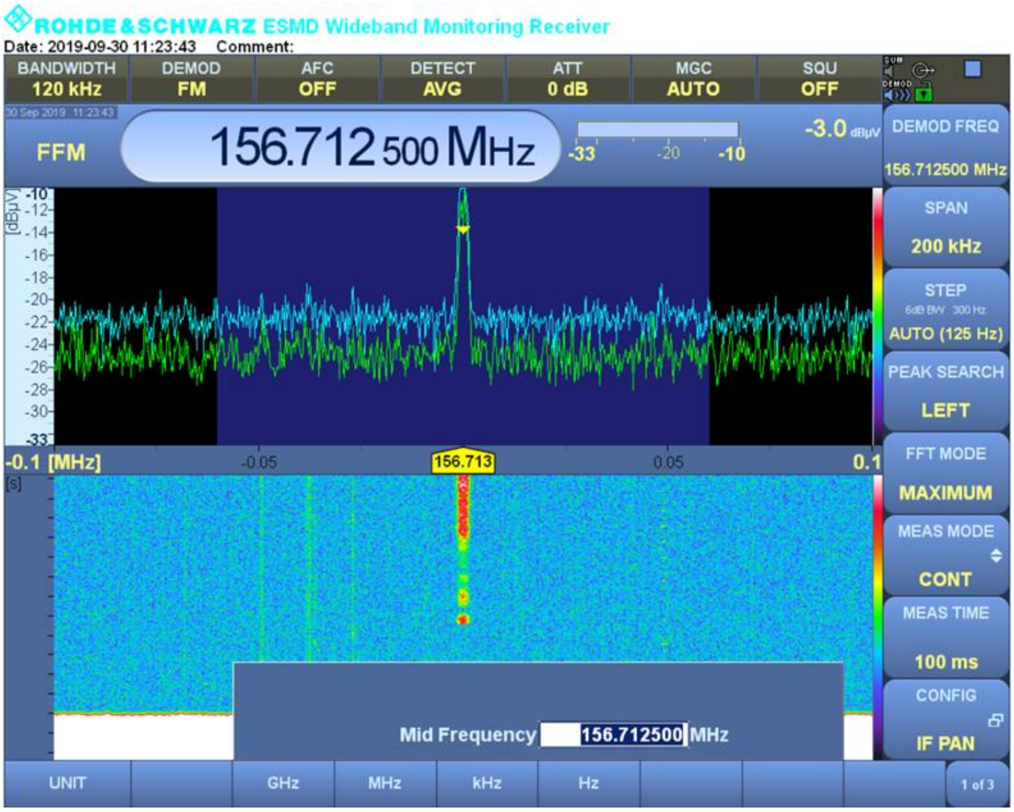
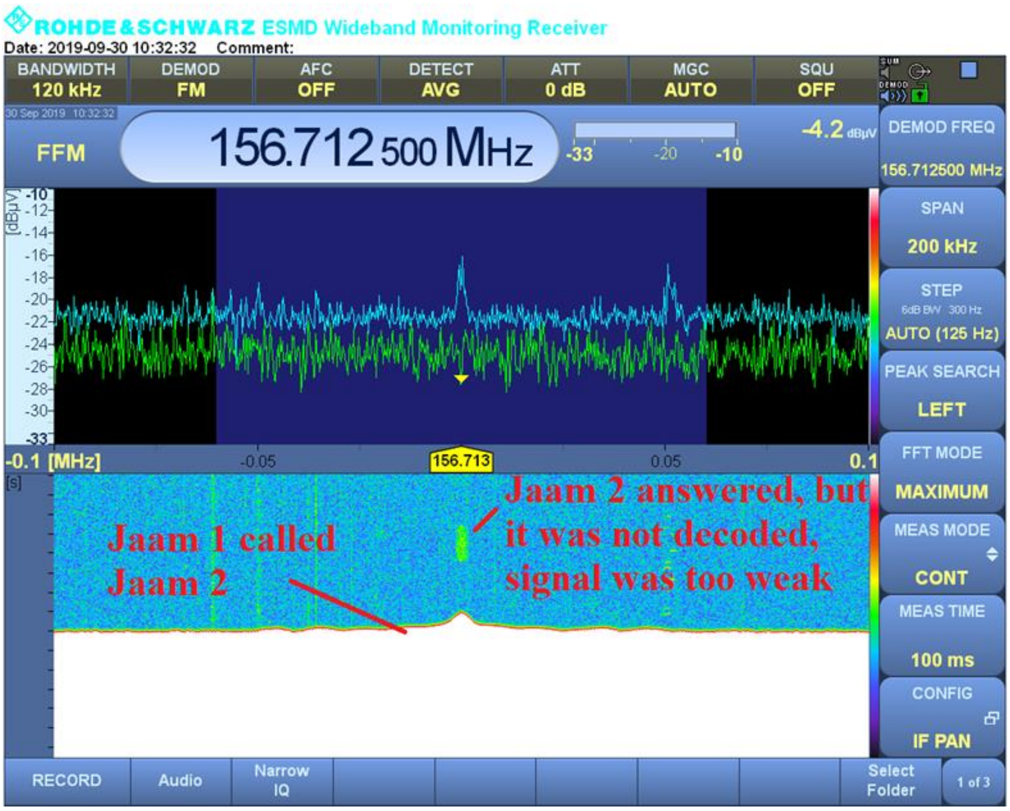


FIGURE 17

Jaam1 calling Jaam2, Jaam2 not decoded



A2.2 Brief summary

Participants in the test were generally positive about the introduction of digital communication. The range of digital communications was the same (or better) than the range of analogue communication. At the maximum distances that the digital communication was understandable (d = 19.6 NM) – analogue communications experienced very high noise and were not understandable. During the digital switchover period, when the digital station and the analogue station are very close together, the digital station signal will overlap the analogue channel, but this did not cause any interference during testing.

A2.2.1 Description of transceivers

Icom IC-F3162T

Switching between analogue and digital mode:

1 P 0 (for analogue – digital switch)


2 Analogue – Digital choice


3 P 0 (for analogue – digital switch)

FIGURE 18

Icom IC-F3162T

FIGURE 19

Icom IC-F5122D

Figure 20 shows an Icom IC-F1000D that works only in digital mode.

FIGURE 20

Icom IC-F1000D

Figure 21 shows an Icom IC-F5400D that works only in digital mode.

FIGURE 21

Icom IC-F5400D

A2.3 Description of marine VHF communication digitalisation testing methodology

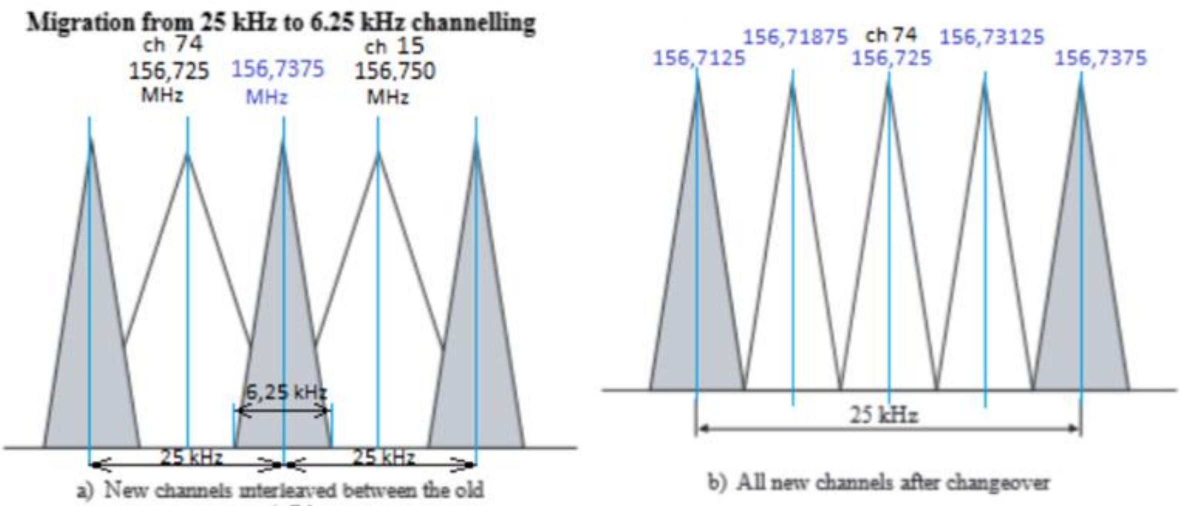
Introduction

This measurement methodology is designed to test possibilities to increase the number of VHF voice channels by switching from analogue to digital channels and reducing channel bandwidth (channel step) K_s = 25 kHz to K_s = 6.25 kHz. This method of measurement is intended to evaluate the effects of the digitalisation of marine radio communications.

Evaluation measurement shall be carried out in the frequency range f = 156.7-156.8 MHz (or more precisely f = 156.709375–156.7625 MHz; ITU Radio Regulations Appendix **18** channel designator K = 15;74).

FIGURE 22

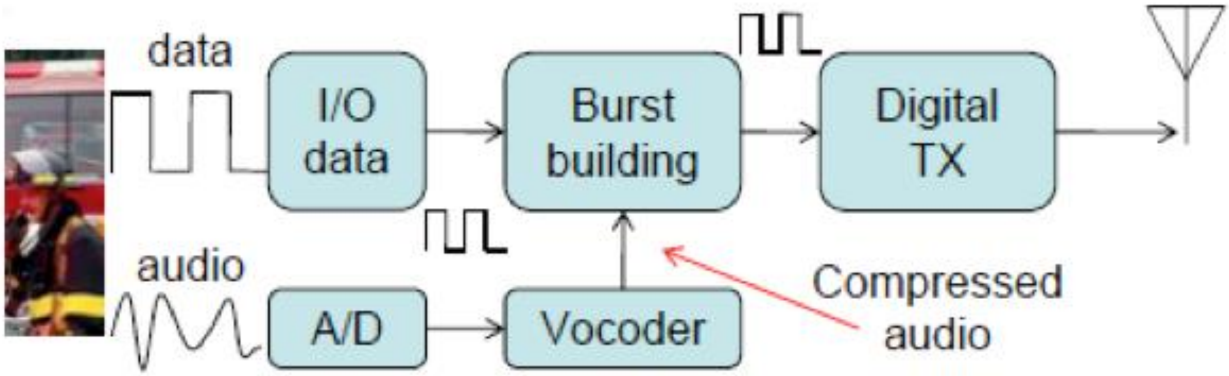
Distribution of ducts in the frequency range



A speech contains certain redundant information that is not reasonable to be sent over the channel. The dPMR standard uses the source to codec AMBE+2 (Advanced Multiband Excitation) or more advanced.

FIGURE 23

Channel block-diagram



The dPMR standard uses Hamming code (Hamming code can correct single errors) in combination with interleaving. However, these are effective up to a certain point. In case of fading, long link or noise, bit errors can reduce the channel's capacity and make it incomprehensible. To assess the quality of the communication calls are made from various distances (d). In break points (d):

– quality of the speech is assessed in the 5-point scale (good without interruption; good with disruptions; distorted without interruption; distorted with interruption; impossible to understand);

– measure the signal to-noise ratio SNR (Signal power or averaged level during transmission/noise level without transmission). The electric field strength E is measured and signal power is calculated;

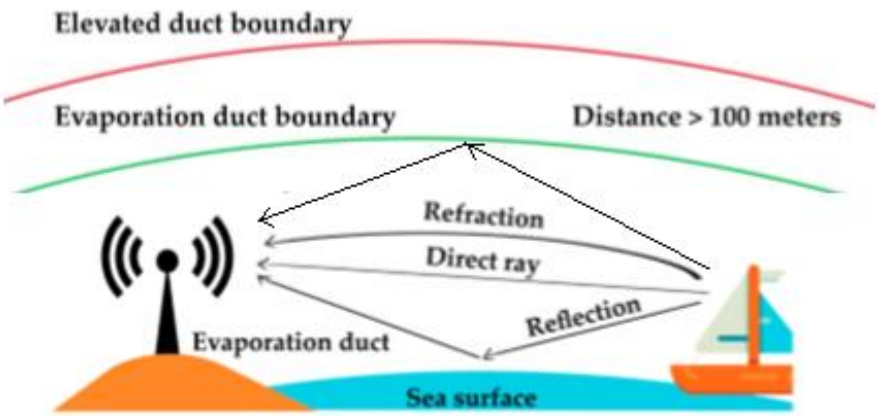
– signal to-noise ratio SNR provided that adjacent analogue channels (K_S = 25 kHz) are working, and digital communication is interrupted. Pictures of spectrum and measurement results are recorded in the event of a disruption.

The measurement shall be carried out between shore station, vessel and handheld stations.

The distance *d* shall be determined using the multi-ray propagation model for over-the-sea communication.

FIGURE 24

Three ways the VHF radio waves are propagated at sea



The reflection of radio waves on the surface of the sea will have a significant effect on the loss of the link. Therefore, only the marine propagation model can be used for this calculation.

Theoretical dual-beam distribution model at sea for line-of-sight:

$$L_{2-ray} = -10\log_{10}\left\{\left(\frac{\lambda}{4\pi d}\right)^2\left[2\sin\left(\frac{2\pi h_t h_r}{\lambda d}\right)\right]^2\right\} \quad (1)$$

where:

$L_{2\text{-}ray}$ is the net loss of the link for two-ray propagation (dB)

λ is the wavelength (m)

d is the distance between the sender and the receiver (m)

h_t and h_r are the height of the transmitter and receiver antenna (m)

Antenna heights:

h_b = 100 m and h_{mk} = 1.5 m (handheld station, boat) where h_b – base station antenna height and h_m – mobile transceiver antenna height.

Antenna gain:

G_b = 7 dBi (base station on shore); G_m = 2.15 dBi (craft); G_k = 0 dBi (handheld transceiver).

Transceiver gain:

P_{WK} = 5 W = 36.9 dBm (handheld transceiver)

P_{wb} = 25 W = 43.9 dBm (base station)

P_{WM} = 25 W = 43.9 dBm.

Radio receiver sensitivity $P_{Rmin} = -115$ dBm.

Transmission power including antenna:

Base station P_b = 43.9 + 7 = 50.9 dBm and $P_r = -115-7 = -122$ dBm

Vessel (G = 2.15 dBi and G = 3 dBi) P_m = 43.9 + 2.2 = 46.1 dBm

Handheld transceiver: P_k = 36.9 dBm.

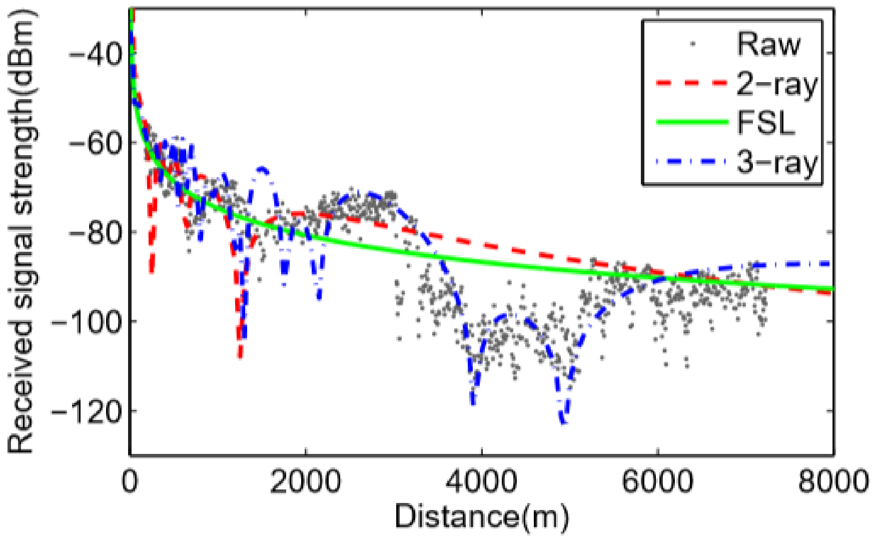
Information about evaporation duct

In case of sunny weather or changes in temperature a layer of steam may appear over the water. The layer may start from h_e = 7 m but normally it starts between 25 and 40 m, in this case radio wave is transmitted in the channel between stream and surface of sea and model of three-ray propagation must be used.

Figure 25 shows that FSL (Free Space Loss) and the two-ray propagation model are not usable at distances longer than d_{break} (break point $d_{break} \approx 2 - 3$ km).

FIGURE 25

Comparison between two- and three-ray propagation models (*f* = 5 GHz, h_r = 10 m)



In this link budget calculations, the two-ray propagation model is used.

Link loss

The calculation allows selecting a lower measurement distance for the test, analyse the measured and theoretical results.

Link loss in the case of two-ray propagation, where the antenna height h_t = 100 m and h_r = 1.5 m and d = 37 km (≈ 20 nautical miles).

$$L_{2-ray} = -10\log_{10}\left\{\left(\frac{\left(\frac{300}{156}\right)}{4\pi 37000}\right)^2\left[2\sin\left(\frac{2\pi 1{,}5\cdot 100}{\left(\frac{300}{156}\right)37000}\right)\right]^2\right\} = 139.2\text{ dB} \quad (2)$$

TABLE 17

Antenna elevation and distance

Link loss depending on antenna heights *d* = 30 NM (dB)				
$H_{/h\,s}$ [m]	1.50	4	10	100
1.50	182.57086	174.05148	166.09268	146.0928
4	174.05148	165.53211	157.57332	137.57413
10	166.09268	157.57332	149.61456	129.61963
Link loss depending on antenna heights *d* = 20 NM (dB)				
$H_{/h\,s}$ [m]	1.50	4	10	100
1.50	175.68442	167.16504	159.20625	139.2065
4	167.16504	158.64567	150.68689	130.68868
10	159.20625	150.68689	142.72818	122.7394

TABLE 17 *(END)*

Antenna elevation and distance

Link loss depending on antenna heights *d* = 13.5 NM (dB)				
$H_{/h\,s}$ [m]	1.50	4	10	100
1.50	168.87395	160.35458	152.39578	132.39633
4	160.35458	151.83521	143.87644	123.88037
10	152.39578	143.87644	135.91785	115.94242
Link loss depending on antenna heights *d* = 8 NM (dB)				
$H_{/h\,s}$ [m]	1.50	4	10	100
1.50	160	151.48063	143.52184	123.52338
4	151.48063	142.96127	135.00256	115.01348
10	143.52184	135.00256	127.04434	107.11266
Link loss depending on antenna heights *d* = 5.4 NM (dB)]				
$H_{/h\,s}$ [m]	1.50	4	10	100
1.50	152.95635	144.43698	136.47821	116.48166
4	144.43698	135.91764	127.95905	107.98362
10	136.47821	127.95905	120.00155	100.15559
Link loss depending on antenna heights *d* = 2.7 NM (dB)				
$H_{/h\,s}$ [m]	1.50	4	10	100
1.50	140.91515	132.3958	124.43711	104.45093
4	132.3958	123.87656	115.91859	96.017048
10	124.43711	115.91859	107.965	88.58803

Finding measurement distances

Sea area-1 (IMO Resolution A.801 (19)).

h \ *H*	50 m	100 m
4 m	23 nm	30 nm

where:

h : height of the mobile station antenna

H : height of the base station antenna

d : = 30 NM = 55.56 km (NM – nautical mile)

d : = 20 NM = 37.04 km.

From Table 17, it can be found that:

a) link loss between mobile and base station at 30 NM from PL = 137.6 dB.

P_r = 46.7 − 137.6 = −90.9 dBm. This condition $P_r > P_{rmin}$ ensures connection (P_{rmin} = −122 dBm that is receiver sensitivity limit).

b) link between two mobile stations (antenna heights h_m = 4 m) is limited by direct visibility, but the transmission power of 25 W allows the connection to be achieved from d = 20 NM (37 km):

PL = 158.6 dB

P_r = 46.7 − 158.6 = −111.9 dBm. This condition $P_r > P_{rmin}$ provides connection (P_r = 117.2 dBm).

Line-of-site $D_{los} = 4.12 \cdot (\sqrt{h_1} + \sqrt{h_2}) = 4{,}12 \cdot (\sqrt{4} + \sqrt{4}) = 16.48$ km = 8.9 NM

c) link between two handheld transceivers (height of antenna 1.5 m)

d = 10 km

PL = 153 dB; P_{rmin} = −115 dBm (antenna gain G = 0 dB)

P_r = 36.9 − 153 = −116.1 dBm This condition $P_r \approx P_{rmin}$ may ensure communication.

d) link between the handheld transceiver and the mobile station (antenna heights 1.5 m and 4 m) d = 18 km = 9.7 NM

PL = 153 dB; P_{rmin} = −117.2 dBm (antenna gain G_k = 0 dB and Gm = *2.15* dB)

P_r = 36.9 − 154.6 = −117.7 dBm This condition $P_r \approx P_{rmin}$ *may* ensure connectivity from distance d = 18 km = 9.7 NM.

e) link between the handheld transceiver and base station (height of 1.5 m and 100 m for aerials) d = 55 km = 30 NM.

Line-of-site distance:

$d_{los} = 4.12 = \cdot (\sqrt{h_1} + \sqrt{h_2})\ 4.12 \cdot (\sqrt{1.5} + \sqrt{100}) = 46.24$ km = 25 NM

PL = 146 dB; P_{rmin} = −122 dBm (antenna gain G_k = 0 dB and G_m = 7 dB)

P_r = 36.9 − 146 = −109.1 dBm, thus $P_r > P_{rmin}$ and link could be assured to distance d = 55 km, but actually there is no link out of line-of-site d_{los} = 46.24 km = 25 NM.

Testing distance between mobile and base stations

Distance needed is derived to ensure the same condition (P_s = 25 W, d = 30 NM = 55.56 km) for the mobile transceiver with transmit power P_s = 5 W. Sender + antenna gain: P_m = 36.9 + 2.2 = 39.1 dBm.

The necessary power at the receiver input P_r = −90.9 dBm.

Therefore, link loss PL =| −90.9 – 39.1| = 130 dB.

For verifying maximum distance, transmitter output power is reduced P_s = 1 W = 30 dBm: d = 30 NM = 55.56 km) P_m = 30 + 2.2 = 32.2 dBm.

The minimum power at the receiving antenna should be P_{rmin} = −122 dBm (antenna gain G_k = 0 dB and G_m = *7* dB).

Thus, the maximum link loss can be PL_{max} = |−122 − 32.2| = 154.2 dB.

Line-of-site $d_{los} = 4.12 \cdot (\sqrt{h_1} + \sqrt{h_2}) = 4{,}12 \cdot (\sqrt{100} + \sqrt{4}) = 49.44$ km = 23.5 NM.

Power spectral density for the digital transceiver.

According to Recommendation ITU-R M.489-2, the radio channel with step K_s = 25 kHz used to transmit 3 kHz voice signal needs bandwidth B = 16 kHz (max deviation $dev_{max} = \pm 5\ kHz$)

Measurement plan

The electric field measurement must be performed from a tripod using a measurement antenna with linear vertical polarisation and circular direction diagram.

Icom *dPMR* (digital maritime mobile radio) protocol shall be used to assess the quality of communications. Two analogue radio transceivers with channel step K_s = 25 kHz and transmit power up to $P_{s\,=}$ 25 W.

Transceivers output power is programmed to be switchable between values P_s = 1 W, 5 W, 25 W (for hand-held transceivers P_s = 1 W, 5 W).

Each antenna used in the test shall be vertical with linear vertical polarization and circular direction diagram.

Antennas mounted on vessel or with magnetic mount having amplification G = 2.15 dBi are used. Antenna height h_m = 4 m. Onshore station antenna height h_m = 100 m with amplification G = 7 dBi.

Handheld transceiver antennas have amplification G = 0 dBi.

One vessel and one boat or vessel with a lower deck (where handheld transceiver with antenna height of h_s = 1.5 m above sea level can be used) are required for testing. On shore there should be a handheld transceiver and base station. The test between the two boats (height $h_s = 1.5$ m) shall be carried out between the boat and the onshore handheld transceiver. As described in 2-c, connection is expected to break at d = 10 km. In case there is not a boat, one handheld transceiver shall remain ashore, and communication should be tested between the ship and the shore-based handheld transceiver (according to point 2-d – communication is expected to break at d = 18 km) should be tested. Between vessel and handheld transceiver maximum length of the link d (tested until the link is broken) will be measured. In other cases, the electrical field strength (noise floor vs radio signal) shall be measured.

The calculation of the measurement points does not take into account any rain loss. In the case rain, loss adds to link loss and distances will change (probably marginally).

Important measurement points in clear weather are:

– Communications test with handheld transceiver until the link is lost, including speech quality evaluation. When the link is broken, electric field strength E can be measured from tripod height. The testing also includes the situation when adjacent analogue channels (in steps K_s = 25 kHz) are occupied. It is expected that the communication will be lost at d = 18 km = 9.7 NM. In this section, the measurement will certainly be carried out.

– In addition, the quality of the speech at the shoreline base station is assessed with reduced power $P_{smobile}$ = 5 W, 1 W. Communication test between boat and handheld transceiver will be done from distance d = 10 km = 5.4 NM. The electric field strength shall be measured on board. If the link is not lost, the adjacent analogue channels (in steps $K_s = 25$ kHz) shall be occupied, and the test will be repeated.

  Also, the quality of the speech with the shoreline base station will be assessed.

– After the vessel is moved to distance d = 20 NM = 37 km from the base station while continuous monitoring quality of the speech using reduced power P_s = 5 W. In case of change the electric field strength E shall be measured and adjacent analogue channels (with steps K_S = 25 kHz) will be occupied and quality of the speech evaluated.

– If possible, proceed further until the link is lost or d = 30 NM = 55 km;

  (Line-of-site d_{los} = 49 km = 26.5 NM for antenna heights h = 4 m and 100 m);

  • The quality of the communication is monitored continuously with reduced output power P_s = 5 W and, in case of changes, the electric field strength E will be measured adjacent analogue channels (with steps K_S = 25 kHz) will be occupied and quality of the speech evaluated.

- If the link is lost transmitting power shall be increased to P_s = 25 W.

Mobile communications engineering, William C. Y. Lee, Chapter “Path loss over flat terrain” [40].

Annex to measurement methodology

Three-ray propagation model:

$$L_{3-ray} = 10\log_{10}\left\{\left(\frac{\lambda}{4\pi d}\right)^2 [2(1+\Delta)]^2\right\}$$

$$\Delta = 2\sin\left(\frac{2\pi h_t h_r}{\lambda d}\right)\sin\left[\frac{2\pi(h_e-h_t)((h_e-h_r)}{\lambda d}\right] \qquad (3)$$

Annex 3

Report from The Netherlands
Digital private mobile radio trial – port of Rotterdam

A3.1 Foreword

Communication between ships and shore has taken place traditionally through the use of VHF radio. VHF radio equipment is used for shipping both at sea and inland. Over the years, the use of VHF radio, and the wish to communicate and be sure that your message is received and understood, has grown. Digitisation in other areas of communication has improved the way to communicate (GSM, LTE, etc.). But, in the marine frequency bands the introduction of new digital communication channels for data has put pressure on the availability of VHF voice channels.

With the introduction of VDES (VHF Data Exchange System) a problem arises that this would not be an easy task for the Netherlands to contend with. The ITU (International Telecommunication Union) has taken the decision that the frequencies for VDES are available from 1 January 2017 in the World Radiocommunications Conference of 2015. Because the Netherlands has foreseen the same problems as they encountered, they sent in a paper to MSC97 to raise awareness of this problem. Due to this, International Maritime Organisation (IMO) agreed that from 1 January 2024 these frequencies should be freed by Contracting States and VDES could then use these frequencies. In the World Radiocommunications Conference of 2019 (WRC-19), ITU also decided on the use of frequencies for VDES satellite communication.

During the IALA (International Association of Lighthouse Authorities) eNAV communications workgroup intersessional meeting in Sydney a possible technical way of a more efficient use of VHF frequencies was presented. This should at least have the same performance standards (functionality) as the current VHF radio.

There are multiple ways to achieve this, but the technical candidate solution presented is called dPMR, currently used in land mobile communications as a replacement for analogue FM voice communication in both VHF and UHF frequency bands. There was, as far as it is known, no specific test done for maritime use of dPMR as a candidate technology to replace analogue VHF radio. Replacing analogue VHF radio needs to be done in such a way that both the “old” and new technology could be used next to each other and therefore an important task is the possible migration plan.

This new candidate technology could also have a place within Maritime Safety Information and/or Smart Shipping because it is possible to embed small data/text with the voice transmission. This information could contain the intentions of ships or information about hazards.

For situational awareness, it is commonly known that eye-sight, VHF radio and radar are the main tools to accomplish this. Next to this, the use of AIS and by transmitting the position of the VHF (digital) radio could complete the picture by showing identification, size, location and which ship is transmitting.

A3.2 Goals

The purpose of the trial was to identify if dPMR could be a possible candidate technology to replace and possibly improve the current voice communication by VHF radio by digitising the voice.

Inquiry goal 1 current functionality

The first inquiry goal was to identify if:

– the quality of the speech was equal or better than with current VHF radio under various ranges;
– migration strategy from current situation to a mixed and maybe a full digital situation;
– possible (harmful) interference of current communication;
– possible (harmful) interference of new digital communication;
– are multiple systems from different vendors capable of working together.

Inquiry goal 2 new functionality

Because dPMR is a ETSI standard, there are already extra functionality embedded in this standard that could be used:

– could position information embedded with the signal be used;
– is it possible to identify the transmitting station;
– could short messages for Maritime Safety Information and/or broadcasting your intention (Smart Shipping) be sent;
– are there possibilities to check the validity of transmissions;
– is there more functionality needed (must have, need to have and nice to have).

Tracks

To ensure that these goals would be reached there were identified two tracks during the trials.

Work together with the users of VHF radio (mariners, operators, skippers, etc.) if the quality of the speech and range is enough and if possible new features are a possible asset.

Check with national ITU organisation (Agentschap Telecom), waterway users and authorities if dPMR will interfere with the current communication, discuss a possible migration strategy and possible adjustments to the standard to make it more appropriate for maritime.

A3.3 Setup

Groups

Before the trial started two main groups were identified. One group of users that actively participated to the trial and one group observers that would be informed about the technology and developments.

The group that actively participated to the trial were technicians, VTS operator, skipper and law enforcement agency for frequency (national ITU organisation).

At first, the idea was to have a small group of observers for the trial. After defining this group, and sending out the invitation, there was a lot of additional interest. This group consists of policy makers, managers, advisors and technicians from different governmental and non-governmental organisations. In total, during the day, 40 people visited the trial.

Area

During the planning of the trial a suitable area needed to be chosen. The Netherlands is very flat with almost no mountains, as a result radio signals carry far and will possibly interfere with other signals. Next to this, the complete area of the Netherlands has about 7 500 ships sailing on a daily base where most are concentrated around the big ports like Port of Rotterdam. The Netherlands also have a lot of infrastructure such as locks and bridges where communication is needed.

Therefore, an ideal place to test interference would be around the Port of Rotterdam.

Locations

For the test, the Port of Rotterdam was contacted for their assistance and to use some of their assets and personnel. The Port of Rotterdam was willing to help us with this trial and offered a de-commissioned VTS centre in the middle of the city and one of their assistance vessels. Also they provided us with an experienced VTS operator and crew for the vessel. These employees of the Port of Rotterdam had been working for at least 30 years at the Port on the vessel or VTS centres.

Figure 26 shows the trial area, VTS centre, vessel and monitoring setup.

FIGURE 26

Trial area

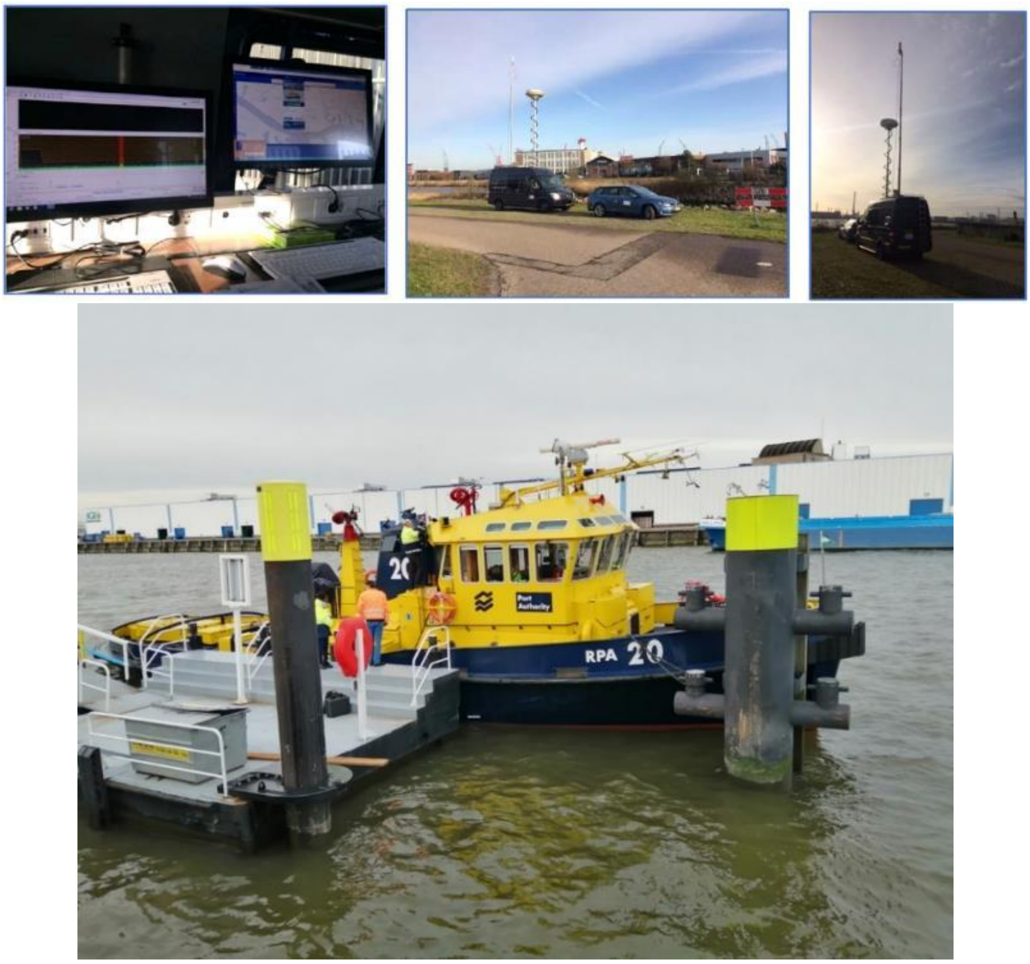
On the right side is the VTS centre where one of the antennas was placed on a height of 16 m. The dots show how the vessel sailed and the time the vessel was on a specific location.

FIGURE 27

Vessel traffic service centre in port of Rotterdam

FIGURE 28

Vessel used belonging to the port of Rotterdam



The monitoring setup/site of the ITU organisation (Agentschap Telecom).

A3.4 Used hardware

The antenna installation on the ship was 0 dB omni-directional antenna (Procom CXL2) with 6-metre RG214 coax cable.

The antenna installation on the VTS centre was a 0 dB omni-directional antenna (Procom CXL2) with a 15-metre RG213 and 5-metre Ecoflex coax cable between the two cables was a lightning security.

During the test the following equipment was used:

– Mobile station Kenwood NEXEDGE NX720 (VTS centre);

– Mobile station Icom IC-F5400DP (ship);

– Portable Kenwood NX220;

– Portable Icom IC-F3400DPT.

The transmit power measurements at the connector of the station were:

– 43 dBm and 30 dBm (Kenwood NEXEDGE NX720);

– 43 dBm, 39 dBm and 36 dBm (Icom IC-F5400D).

FIGURE 29

Pictures of the equipment

A3.5 Used frequencies

A trial license for the period from the 1 December 2019 till the end of 1 July 2020 was obtained. This was done so more trials could take place. Next to this it would provide the opportunity to give different users and interested people to hear and experience the quality of voice and use of the equipment. Plans were made to give participants of the VTS-ENAV Symposium (25 to 29 May 2020) a chance to experience the digitisation of VHF radio (dPMR).

Although a license has been received to use all maritime frequencies, the test was made on frequencies used by Rijkswaterstaat only, so as not to affect the normal port operations.

FIGURE 30

Rijkswaterstaat frequencies used in the trial

	162.5				162.5				
analogue FM voice	162.500 channel 2038				162.525 channel 2098				
Direct replacement	162.490625 D1	162.496875 D2	162.503125 D3	162.509375 D4	162.515625 D5	162.521875 D6	162.528125 D7	162.534375 D8	
ITU-style	162.487500 I1	162.493750 I2	162.500000 I3	162.506250 I4	162.512500 I5	162.518750 I6	162.525000 I7	162.531250 I8	162.537500 I9

A3.6 Considerations

IALA

That any evaluated technologies have a clear migration path from the current analogue voice services to the new digital voice services by allowing both the digital and analogue services to co-exist in the same transceiver for the duration of the entire migration period. This could extend to using the same antenna and other existing physical installation hardware;

The channel efficiency should be a high priority, by allowing four (4) or more digital voice channels for each 25 kHz maritime VHF voice channel;

The digital service includes the capability of transmitting the location of the radio for the entire duration of the digital voice conversation;

The digital service allows a Short Message Service (SMS) without the need to set up a digital or other voice call;

The digital voice quality be similar to, or better than, the analogue voice service, especially using weaker radio signals at the extent of the radio coverage.

Before trial

– that the candidate technologies are easy to use by the users and limit the possibility of the users to make mistakes;
– that the candidate technology is independent of other (supporting) technology (like GPS) and manufacturer (no vendor lock);
– support the current functionality of VHF radio (DSC/ATIS);
– costs of the equipment are around the same as current;
– impact on current regulations is minimal (RR Appendix **18**);
– the candidate technology, with most of the requested functionality, should already be available;
– the candidate technology should be future-proof;
– could support (Cyber) security for instance to check your own transmissions;
– support of Smart Shipping, for instance sending small data packets with the intentions of the ship;
– harmonised;
– should be implemented using open standards.

A3.7 Results

Start trial

Before the trial was started, the equipment and installation (antenna and cabling) were tested. The first results were that the antennas initially used were not good enough and needed to be replaced.

The next test was to test the installation. On the ship, this was all good after replacing the antenna but on the VTS centre a problem with the cabling was encountered. This problem with the cabling needed us to replace the cabling in the VTS centre. This was done very quickly by the technical staff of the Port of Rotterdam.

After the installation was tested again, there were no problems with cabling or antennas.

During the testing of the installation both technicians of CML and Koning and Hartman checked the configuration of the stations and tested them. These tests passed ok but not all of the envisioned features could not be tested at that moment because of some missing parts. These parts were delivered after the test but did not jeopardise the main purpose of the test.

FIGURE 31

Checking the configuration of settings

After everything was tested the scenarios were checked again. The red line in the scenario's was to go sailing and test on different distances with different power levels, digital and analogue communication and two languages English and Dutch. During the test, the time, distance, power, language and possible (harmful) interference were recorded. For the last test, the equipment of our ITU organisation was used and of course informed the VTS operators of the Port of Rotterdam to inform us if something unusual within communication happened.

Test day

FIGURE 32

Questions and observations

On the test day (10 December 2019) there was a short instruction and roles. The main group of the attendees would be busy with the trial while two other staff would accompany the observers. There was a presentation about the trial to the observers after which the observers were asked to post up their questions and observations.

Use of equipment

FIGURE 33

Monitoring of equipment during the trial

After a short introduction and demonstration, the use of the equipment was easy to use. They are similar as VHF radio equipment. For the trial, there were buttons programmed to switch between high and low power and change between all channels, analogue and digital. Also, the display showed which channel you were using, transmitting level, reception level and the station on the ship recorded the voice communications.

Next to the default installation, an emergency button function on the Kenwood equipment was also programmed. This as a possible feature to show to the observers.

Voice quality

During the test all voice was recorded on the ship side only. This was because the equipment at the VTS centre did not have this functionality. Therefore, only recordings coming from the VTS centre and recorded on the ship are accountable for the test.

Both users on the ship and the VTS centre were enthusiastic about the quality of the digital voice and said that especially on the edges of the transmitting range the sound was clear and less tiring to listen to. Also the interference was much less.

During the test, the reception quality of the digital transmissions was higher than on the same range when compared with analogue ones.

A question arose that if the operator could distinguish if the mariner would still be able to operate a ship by the sound of his/her voice was answered: that to determine this would not only be done by the voice quality but also by the sentence structure, response time and ability to response.

A3.8 Interference

During the test the ITU organisation (Agentschap Telecom) monitored if the equipment stayed in the standard (ETSI TS 102.658 and RR Appendix **18**) and if it caused (harmful) interference. They measured that frequency 162,500 MHz was used for analogue communication and frequency 162.534375 MHz for digital communication (dPMR).

The two positions where the monitoring took place from was about 4 km from the VTS centre (because monitoring next to the transmitter has no use). Locations were Wilhelminahaven/Nieuwe Waterwegstraat and the Karel Doormanweg, both near the waterway.

The monitoring equipment consists of a broadband-omni-directional antenna at a height of 10 m connected to a Tektronix RSA real-time spectrum analyser. At the same time this was monitored with a Rohde & Schwarz direction-tracker. This antenna height was 6 m. During the test, no irregularities were discovered. A small note was that some of the communication could not be received due to the lower height of the antennas what could be explained.

On the ship, signals were also monitored with a Rohde & Schwarz FSH-6 spectrum-analyser if they noticed any interference in the spectrum when they were unable to connect to the VTS centre and ship. Also, no irregularities were found. A note was that they only sailed on the main fairways and not all the inlets/basins at the Port of Rotterdam.

During the test there were some findings these were:

– On lager distances, quality deteriorated when sailing. This was probably caused by the horizon/line of sight that current VHF radio also encounters.
– On a specific area of the fairway the quality deteriorated because of large storage tanks standing between the VTS and vessel. The large storage tanks probably blocked or reflected the signal. (multipath).
– When the quality deteriorated it presented itself by losing the connectivity or a “metallic/robotised” sound. When it happened with analogue VHF communication it caused noise).

During the trial, it was not possible to test the adjacent and nearby channel rejections and this was tested later for both the current 25 kHz analogue FM (voice) and 6.25 kHz dPMR (digital voice) channels. This test is to quantify the interference potential of an adjacent/near dPMR channel on an existing analogue voice channel.

Also additional qualitative testing was also done to establish the closest distance an interfering dPMR radio would need to be before affecting the analogue reliever. This is particularly important when it comes to channel planning and migration strategies.

These tests were done by using the ETSI specification procedure, measurements were made using two different instruments to perform the SINAD measurement, the 2955R having a flat filter response, whereas the 8 903 measurement uses the psophometric filter as defined in ETSI EN 300 086. A third set of measurements were made using the TIA procedure.

Two channel plans have been proposed for implementing the replacement of analogue voice with digital voice:

– direct replacement – where a 25 kHz analogue channel is split exactly into four dPMR channels;
– ITU-style, where the channel centre of digital channels is aligned with the channel of the analogue channel, so that the extreme digital channels overlap into the adjacent analogue channels.

These two channel plans are indicated in the channel selections as D5 to D8 and I5 to I9 respectively.

A3.9 After trial considerations/questions

During and after the trial the following additional observations/considerations came out:

– be able of shutting down a transmitter remotely;
– be able to limit the maximum time of one conversation;
– be able of integration GMDSS (DSC, MSI);
– capable of dual watch functionality;
– identification integration embedded in signal like MMSI, Callsign or ATIS for the entire duration of the digital voice conversation;
– be able to switch automatically between analogue and digital voice transmissions;
– capable of detection of poor signal;
– capable of detecting of interference;
– to be able to cope with multicast/diversity;
– support (half) duplex;
– possible of to dedicate one digital channel for data only;
– support multi languages by voice and user interface;
– tests were in perfect weather/communication conditions, so how does it operate when not;
– must have an interface to connect to other bridge equipment for instance obtaining ships position from its centralised positioning system;
– how can you detect destruction of the signal, like with analogue;
– would the repeater and trunking possibilities enhance communication and safety in a Port or traffic dense area.

FIGURE 34

Users appeared content with the operation of the digital radio system

A3.10 Conclusions

Use equipment

During the test the users had no problem with operating the equipment. There might be some slight adjustments to the user interface when integrating DSC/dual watch functionality.

Voice quality

Both the users and the observers found the voice quality the same or better than analogue. The users reported back that listening to digital voice with the noise reduction made it easier and less intensive to listen. Concluding that with digital transmission of voice, if a mariner is still capable of operating his ship is equal as analogue.

The bad reception or failure of digital VHF that caused a "metallic/robotised" sound of losing the connection is similar of analogue VHF where it causes noise. The impact and acceptance of this against the gains has to be analysed and decided.

Frequencies

In the lab tests, the equipment exceeds the requirements of both ETSI and TIA standards by some margin and the rejection of the dPMR channels in excess of 70 dB in the direct frequency replacement format indicates that that the same adjacent channel practices can be applied to both analogue and digital implementations.

In the case of the ITU channel plan, the 61 dB result on Channel I5 indicates that this arrangement could be marginal and would need very careful consideration before implementing.

The Walk Test was provided to illustrate the difference in range of the interferers that could be expected in a typical deployment. It shows that the use of Channel I5 in close proximity to the wanted analogue channel will produce more interference than the existing analogue channel and so calls into question its usefulness in a real-world scenario. Although Channel D5 does interfere slightly more than the analogue, it is not significantly so (only 11 m compared to 10 m). All the other channels showed that they would introduce less interference than the existing analogue channel and so could be deployed using the same (or possibly stricter) channel planning criteria as currently used for analogue channels.

The field test did not show any (harmful) interference.

Additions

The use of the candidate technology dPMR for maritime use could be a good option. Some possible functionalities need to be defined by IMO in their performance standards.

Annex 4

International Telecommunication Union Codecs

A4.1 Recommendation ITU-T G.711

Recommendation ITU-T G.711.0 describes a lossless compression scheme of ITU-T G.711 bitstream, mainly aimed for transmission over IP (e.g. VoIP).

The coder operates on frame lengths of 40, 80, 160, 240 and 320 samples, has a maximum algorithmic delay equals to the frame length, and has a worst-case computational complexity of less than 1.7 weighted million operations per second (WMOPS) for encoder plus decoder.

This Recommendation includes an electronic attachment containing the ANSI C code (fixed-point arithmetic implementation of the specification), as well as a non-exhaustive set of test signals for use with it.

Recommendation ITU-T G.711.1 describes embedded wideband speech and audio coding algorithm operating at 64, 80 and 96 kbit/s.

The encoder input and decoder outputs are sampled at 16 kHz by default, but 8-kHz sampling is also supported. When sampled at 16 kHz, the output of the ITU-T G.711.1 coder can encode signal with a bandwidth of 50-7 000 Hz at 80 and 96 kbit/s, and for 8-kHz sampling, the output may produce signal with a bandwidth ranging from 50 up to 4 000 Hz, operating at 64 and 80 kbit/s (the bandwidth of the narrowband signal output from the decoder is characterised by the built-in split-band filterbank which has cut-off frequency of 4000 Hz). At 64 kbit/s, Recommendation ITU-T G.711.1 is compatible with Recommendation ITU-T G.711. The coder operates on 5 ms frames, has a maximum algorithmic delay of 11.875 ms, and has a worst-case computational complexity of 8.70 WMOPS.

The encoder produces an embedded bitstream structured in three layers corresponding to three available bit rates: 64, 80 and 96 kbit/s. The bitstream can be truncated at the decoder side or by any component of the communication system to adjust the bit rate to the desired value, but since it does not contain any information on which layers are contained, an implementation would require outband signalling on which layers are available.

The underlying algorithm has a three layer coding structure: log companded pulse code modulation (PCM) of the lower frequency band including noise feedback, embedded PCM extension with adaptive bit allocation for enhancing the quality of the base layer in the lower frequency band, and weighted vector quantisation coding of the higher frequency band based on modified discrete cosine transformation.

A4.2 Recommendations ITU-T G.722 and ITU-T G.722.1

Recommendation ITU-T G.722 describes the characteristics of an audio (50 to 7 000 Hz) coding system which may be used for a variety of higher quality speech applications. The coding system uses sub-band adaptive differential pulse code modulation (SB-ADPCM) within a bit rate of 64 kbit/s. The system is henceforth referred to as 64 kbit/s (7 kHz) audio coding. In the SB-ADPCM technique used, the frequency band is split into two sub-bands (higher and lower) and the signals in each sub-band are encoded using ADPCM. The system has three basic modes of operation corresponding to the bit rates used for 7 kHz audio coding: 64, 56 and 48 kbit/s. The latter two modes allow an auxiliary data channel of 8 and 16 kbit/s respectively to be provided within the 64 kbit/s by making use of bits from the lower sub-band.

Recommendation ITU-T G.722, Appendix II describes digital test sequences for the verification of the ITU-T G.722 64 kbit/s SB-ADPCM 7 kHz codec. This guide gives information concerning the digital test sequences which should be used to aid verification of implementation of the ADPCM codec part of the wideband coding algorithm.

Recommendation ITU-T G.722.1 describes a low complexity encoder and decoder that may be used for 7 kHz bandwidth audio signals working at 24 kbit/s or 32 kbit/s. Furthermore, this algorithm is recommended for use in hands-free applications such as conferencing where there is a low probability of frame loss. It may be used with speech or music inputs.

The digital input to the coder may be in a 14-, 15- or 16-bit 2’s complement format, at a sampling rate of 16 kHz (handled in the same way as in Recommendation ITU-T G.722). The analogue and digital interface circuitry at the encoder input and decoder output should conform to the same specifications described in Recommendation ITU-T G.722. The algorithm is based on transform technology, using a modulated lapped transform. It operates on 20 ms frames (320 samples) of audio. Because the transform window (basis function length) is 640 samples and a 50 percent (320 samples) overlap is used between frames, the effective look-ahead buffer size is 20 ms. Hence the total algorithmic delay of 40 ms is the sum of the frame size plus look-ahead. All other delays are due to computational and network transmission delays.

Recommendation ITU-T G.722.1 includes a software package which contains the encoder and decoder source code and a set of test vectors for developers. These vectors are a tool that can provide an indication of success in implementing this codec.

Recommendation ITU-T G.722.2 describes the high quality adaptive multi-rate wideband (AMR-WB) encoder and decoder that is primarily intended for 7 kHz bandwidth speech signals. AMR-WB operates at a multitude of bit rates ranging from 6.6 kbit/s to 23.85 kbit/s. The bit rate may be changed at any 20-ms frame boundary.

Annex C includes an integrated C source code software package which contains the implementation of the ITU-T G.722.2 encoder and decoder and its Annexes A and B and Appendix I.

A set of digital test vectors for developers is provided in Annex D. These test vectors are a verification tool that can provide an indication of success in implementing this codec. Digital test sequences are necessary to test for a bit-exact implementation of the adaptive, multi-rate wideband (AMR-WB) speech-transcoder; voice-activity detection; comfort noise generation; and source controlled rate operation.

A4.3 Recommendation ITU-T G.723.1

Recommendation ITU-T G.723.1 specifies a coded representation that can be used for compressing the speech or other audio signal component of multimedia services at a very low bit rate. In the design of this coder, the principal application considered was very low bit-rate, visual telephony as part of the overall ITU-T H.324 family of Recommendations. This coder has two bit rates associated with it (5.3 and 6.3 kbit/s).

A4.4 Recommendation ITU-T G.726

The characteristics below are recommended for the conversion of a 64 kbit/s A-law or Mu-law PCM channel to and from a 40, 32, 24 or 16 kbit/s channel. The conversion is applied to the PCM bit stream using an ADPCM transcoding technique. The relationship between the voice frequency signals and the PCM encoding/decoding laws is fully specified in Recommendation ITU-T G.711.

The principal application of 24 and 16 kbit/s channels is for overload channels carrying voice in digital circuit multiplication equipment (DCME).

The principal application of 40 kbit/s channels is to carry data modem signals in DCME, especially for modems operating at greater than 4 800 kbit/s.

Appendix II describes the test sequences (vectors) for the ADPCM algorithms of Recommendation ITU-T G.726 at the four fixed bit rates (16 kbit/s, 24 kbit/s, 32 kbit/s, 40 kbit/s) for both A-law and Mu-law.

NOTE – Recommendation ITU-T G.726 is the consolidation of Recommendation ITU-T G.721 (1988) and Recommendation ITU-T G.723 (1988), which are now superseded as individual Recommendations.

A4.5 Recommendation ITU-T G.728: Coding of speech at 16 kbit/s using low delay code excited linear prediction

Recommendation ITU-T G.728 contains the description of an algorithm for the coding of speech signals at 16 kbit/s using low-delay, code-excited, linear prediction.

The LD-CELP algorithm consists of an encoder and a decoder. The essence of code excited linear prediction (CELP) techniques, which is an analysis-by-synthesis approach to codebook search, is retained in LD-CELP. The LD-CELP however, uses backward adaptation of predictors and gain to achieve an algorithmic delay of 0.625 ms. Only the index to the excitation codebook is transmitted. The predictor coefficients are updated through linear predictive coding (LPC) analysis of previously quantised speech. The excitation gain is updated by using the gain information embedded in the previously quantised excitation. The block size for the excitation vector and gain adaptation is five samples only. A perceptual weighting filter is updated using LPC analysis of the unquantized speech.

A4.6 Recommendation ITU-T G.729: Coding of speech at 8 kbit/s using conjugate-structure algebraic-code excited linear prediction

Recommendation ITU-T G.729 contains the description of an algorithm for the coding of speech signals at 8 kbit/s using conjugate-structure algebraic-code-excited linear-prediction (CS-ACELP). This coder is designed to operate with a digital signal obtained by first performing telephone bandwidth filtering (Recommendation ITU-T G.712) of the analogue input signal, then sampling it at 8 000 Hz, followed by conversion to 16-bit linear PCM for the input to the encoder. The output of the decoder should be converted back to an analogue signal by similar means. Other input/output characteristics, such as those specified by Recommendation ITU-T G.711 for 64 kbit/s PCM data, should be converted to 16-bit linear PCM before encoding, or from 16-bit linear PCM to the appropriate format after decoding. The bitstream from the encoder to the decoder is defined within this Recommendation.

Recommendation ITU-T G.729 and its Annexes and Appendices offer different functionalities in terms of various bit rates and/or DTX operations using either fixed point or floating point arithmetic. Table 18 summarises these functionalities.

TABLE 18

Recommendation ITU-T G.729 functionalities

	Annex										
Functionality	**-**	**A**	**B**	**C**	**D**	**E**	**F**	**G**	**H**	**I**	**C+**
Low Complexity		X	X								
Fixed-point	X	X	X		X	X	X	X	X	X	
Floating-point				X							X
8 kbit/s	X	X	X	X	X	X	X	X	X	X	X
6.4 kbit/s					X		X		X	X	X
11.8 kbit/s						X		X	X	X	X
DTX			X				X	X		X	X

A4.7 Recommendation ITU-T G.729.1: ITU-T G.729 based embedded variable bit-rate coder: An 8-32 kbit/s, scalable wideband, coder-bitstream interoperable with ITU-T G.729 codecs

Recommendation ITU-T G.729.1 describes an 8-32 kbit/s, scalable, wideband speech and audio coding algorithm interoperable with ITU-T G.729, ITU-T G.729A and ITU-T G.729B codecs. The output of the ITU-T G.729.1 coder has a bandwidth of 50-4 000 Hz when operated at 8 and 12 kbit/s and 50-7 000 Hz when operated from 14 to 32 kbit/s. At 8 kbit/s, ITU-T G.729.1 codecs are fully interoperable with codecs conforming to Recommendation ITU-T G.729, Recommendation ITU-T G.729 Annex A and Recommendation ITU-T G.729 Annex B. The coder operates on 20 ms frames and has an algorithmic delay of 48.9375 ms. By default, the encoder input and decoder output are sampled at 16 kHz. The encoder produces an embedded bitstream structured in 12 layers corresponding to 12 available bit rates from 8 to 32 kbit/s. The bitstream can be truncated at the decoder side or by any component of the communication system to adjust “on the fly” the bit rate to the desired value with no need for outband signalling. The underlying algorithm is based on a three-stage coding structure: embedded CELP coding of the lower band (50-4 000 Hz), parametric coding of the higher frequency band (4 000-7 000 Hz) by time-domain bandwidth extension, and enhancement of the full frequency band (50-7 000 Hz) by a predictive transform coding technique referred to as time-domain aliasing cancellation.