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Electromagnetic compatibility and Radio spectrum Matters (ERM); digital Private Mobile Radio (dPMR) General System Design

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Foreword

This Technical Report (TR) has been produced by ETSI Technical Committee Electromagnetic compatibility and Radio spectrum Matters (ERM).

Introduction

The present document has been produced to provide an introduction to dPMR for potential system purchasers, network operators and service users.

It is in relation to TS 102 490 [i.1] and TS 102 658 [i.2] covering the technical requirements for digital Private Mobile Radio (dPMR), as identified below:

- TS 102 490 [i.1].
- TS 102 658 [i.2].

It provides an overview, a description of the dPMR services and facilities, technical background and radio aspects, protocol and service performance, and guidance on numbering and addressing.

It should be understood that, as in all standard setting activities, there is an inherent conflict between the wish to have as broad a standard as possible and at the same time wanting to have as much of that broad standard available and implemented right from the beginning. Potential system purchasers, network operators and service users should make sure they influence the suppliers to have their required functionality available when they need it.

Equipment manufacturers will use the broad flexibility provided within the standard to develop and implement systems in various ways, and still be conforming according to the standard. This broad availability of systems, each optimized around certain features and functionalities, needs to be carefully analysed by a network operator and system user to find the supplier with a system suited best for their needs.

1 Scope

The present document covers digital Private Mobile Radio (dPMR) equipment using FDMA technology with channel spacing of 6,25 kHz supporting voice and data applications capable of operating in the existing licensed land mobile service frequency bands below 1 000 MHz.

The present document includes the baseband signal processing parameters of the Physical Layer (PL) and the protocol structure at the air interface. The protocol supports different levels of functionality from peer to peer mode to managed base station access mode: The equipment is based on FDMA with channel spacing of 6,25 kHz supporting voice and data applications.

dPMR equipment is designed to be compliant with the appropriate harmonized standard for spectrum use, EN 301 166-2 [i.4].

1.1 Scope of TS 102 490

The present document covers digital private mobile radio equipment operating in peer-to- peer mode only. It covers only handportable equipment complying with EN 301 166-2 [i.4] and having an integral antenna.

This equipment is for use:

i) In accordance with ECC/DEC/(05)12 [i.7] on harmonized frequencies, technical characteristics, exemption from individual licensing and free carriage and use of digital PMR446 applications operating in the frequency band 446,100 MHz to 446,200 MHz.

   The equipment conforms to the technical requirements for Digital PMR 446 included in ECC/DEC/(05)12 [i.7]. This permits operation in the frequency range 446,100 MHz to 446,200 MHz, maximum e.r.p of 500 mW, and a maximum transmitter time-out-time of 180 seconds.

ii) In the frequency band 149,01875 MHz to 149,11875 MHz under exemption from individual licensing. This permits a maximum e.r.p of 500 mW, and a maximum transmitter time-out-time of 180 seconds.

1.2 Scope of TS 102 658

The present document supports different levels of functionality from peer to peer mode to managed base station access mode:

Mode 1 Peer to peer (direct mode) operation without Base Stations or infrastructure.

Mode 2 dPMR systems incorporating one or more Base Stations for repeating or providing system gateways.

Mode 3 dPMR systems operating under a managed access mode in systems incorporating one or more Base Stations.

All three modes of operation of the present air interface are designed to be compliant with the appropriate harmonized standard for spectrum use, EN 301 166-2 [i.4]. A polite spectrum access protocol for sharing the physical channel has also been specified.

2 References

References are either specific (identified by date of publication and/or edition number or version number) or non-specific. For specific references, only the cited version applies. For non-specific references, the latest version of the reference document (including any amendments) applies.

Referenced documents which are not found to be publicly available in the expected location might be found at http://docbox.etsi.org/Reference.

NOTE: While any hyperlinks included in this clause were valid at the time of publication ETSI cannot guarantee their long term validity.

2.1 Normative references

The following referenced documents are necessary for the application of the present document.

Not applicable.

2.2 Informative references

The following referenced documents are not necessary for the application of the present document but they assist the user with regard to a particular subject area.

[i.1] ETSI TS 102 490: "Electromagnetic compatibility and Radio spectrum Matters (ERM); Peer-to-Peer Digital Private Mobile Radio using FDMA with a channel spacing of 6,25 kHz with e.r.p. of up to 500 mW".

[i.2] ETSI TS 102 658: "Electromagnetic compatibility and Radio spectrum Matters (ERM); Digital Private Mobile Radio (dPMR) using FDMA with a channel spacing of 6,25 kHz".

[i.3] IEC EN 61162-1 (2008): "Maritime navigation and radio communications equipment and systems - Digital Interfaces - Part 1: Single talker and multiple listeners".

[i.4] ETSI EN 301 166-2: "Electromagnetic compatibility and Radio spectrum Matters (ERM); Land Mobile Service; Radio equipment for analogue and/or digital communication (speech and/or data) and operating on narrow band channels and having an antenna connector; Part 2: Harmonized EN covering essential requirements of article 3.2 of the R&TTE Directive".

[i.5] CEPT Recommendation T/R 25-08: "Planning criteria and coordination of frequencies in the Land Mobile Service in the range 29.7-921 MHz".

[i.6] CEPT ERC Report 25: "The European table of frequency allocations and utilizations covering the frequency range 9 kHz to 275 GHz".

[i.7] CEPT ECC/DEC/(05)12: "ECC Decision of 28 October 2005 on harmonized frequencies, technical characteristics, exemption from individual licensing and free carriage and use of digital PMR 446 applications operating in the frequency band 446,1-446,2 MHz".

[i.8] Draft CEPT ECC Decision (06)06 (WGFM, Cavtat, April 2006): "ECC Decision on the availability of frequency bands for the introduction of Narrow Band Digital Land Mobile PMR/PAMR in the 80 MHz, 160 MHz and 400 MHz bands".

[i.9] ETSI TS 102 726-1: "Electromagnetic compatibility and Radio spectrum Matters (ERM); Conformance testing for Mode 1 of the digital Private Mobile Radio (dPMR) Part 1: Protocol Conformance Implementation Statement (PICS) proforma".

[i.10] ETSI TS 102 726-2: "Electromagnetic compatibility and Radio spectrum Matters (ERM); Conformance testing for Mode 1 of the digital Private Mobile Radio (dPMR); Part 2: Test Suite Structure and Test Purposes (TSS&TP) specification".

[i.11] ETSI TS 102 726-3: "Electromagnetic compatibility and Radio spectrum Matters (ERM); Conformance testing for Mode 1 of the digital Private Mobile Radio (dPMR); Part 3: Interoperability Test Suite Structure and Test Purposes (TSS&TP) specification".

[i.12] ETSI TS 102 587-1: "Electromagnetic compatibility and Radio spectrum Matters (ERM); Peer-to-Peer Digital Private Mobile Radio; Part 1: Conformance testing; Protocol Implementation Conformance Statement (PICS) proforma".

3 Definitions, symbols and abbreviations

3.1 Definitions

For the purposes of the present document, the following terms and definitions apply:

active_hang_time: time during which a Mode 2 BS preserves the channel for the parties involved in a call

Appended_Data: message carrying principally data that is formatted according to the present document

Base Station (BS): fixed end equipment that is used to obtain dPMR services

beacon channel: channel that carries synchronous beacon frames timed from a BS

bearer service: type of telecommunication service that provides the capability for the information transfer between user network interfaces, involving only low layer functions (layers 1 to 3 of the OSI model)

NOTE: Confirmed Data and Unconfirmed Data are examples of bearer services.

burst: smallest predefined block of continuous bits containing information or signalling

NOTE: The burst may include a guard time at the beginning and end of the burst used for power ramp-up and ramp-down.

call: complete sequence of related transactions between MS

NOTE: Transactions may be one or more bursts containing specific call related information

Caller Line Identity (CLI): ability to see who is calling you before answering the telephone

call_hang_time: time during which a Mode 1 or Mode 2 channel is available for an emergency pre-emption

complementary service: dPMR service that enables complementary data to be passed between MS and BS as part of the call set-up phase of another service (such as voice)

Control plane (C-plane): part of the protocol stack dedicated to control and data services

downlink: transmission from BS to MS(s)

extended address: address of an entity that is not a native MS/BS individual/group identity

feature: attribute intrinsic to a station, e.g. MS has an address

intrinsic service: service which is inherent within a voice or data service

late entry: where receiving stations that have missed the start of a transmission are able to recover all information about the call from subsequent message frames

line connected: call whereby one end of the call is connected to the radio system that does not use the DMR Air Interface

NOTE: Examples may be connection to the PSTN or a PABX.

logical channel: distinct data path between logical endpoints

Manufacturers ID (MID): 8 bit identifier assigned to a particular manufacturer

Mobile Station (MS): physical grouping that contains all of the mobile equipment that is used to obtain dPMR mobile services

mode: class of operation of a dPMR system

multi-part call set-up: call set-up procedure whereby the full information to be exchanged between entities cannot be accommodated in a single message frame

NOTE: The UDT procedure is invoked to transfer the address information using UDT signalling. UDT is also invoked to transport complementary and user data between dPMR entities.

network personalization: configuration parameters appropriate to network configuration programmed into an MS that may be set by an external agency but not by the user of an MS

payload: part of a data stream representing the user information

peer-to-peer mode: mode of operation where MS may communicate outside the control of a network

NOTE: This is communication technique where any MS may communicate with one or more other MS(s) without the need for any additional equipment (e.g. BS).

personalization: address and configuration information that characterizes a particular dPMR MS

NOTE: This information may be implanted by the installer before putting an MS into service.

physical channel: FDMA transmission

polite protocol: Listen Before Transmit (LBT) protocol

NOTE: This is a medium access protocol that implements a LBT function in order to ensure that the channel is free before transmitting.

prefix: most significant digit of an MS address in the user domain

radio frequency channel: radio frequency carrier (RF carrier)

NOTE: This is a specified portion of the RF spectrum. The RF carrier separation is 6,25 kHz.

Received Signal Strength Indication (RSSI): root mean squared value of the signal received at the receiver antenna

signalling: exchange of information specifically concerned with the establishment and control of connections, and with management, in a telecommunication network

simplex: mode of working by which information can be transferred in both directions but not at the same time

NOTE: Simplex is also known as half duplex.

superframe: four concatenated FDMA frames

NOTE: A superframe has a length of 320 ms.

supplementary service: supplementary service modifies or supplements a tele-service or bearer service

NOTE: Consequently, it cannot be offered to a user as a standalone service. It is offered together with or in association with a tele-service or bearer service. The same supplementary service may be common to a number of telecommunication services. Late entry is an example of supplementary service.

talkgroup: collection of MSs that have the same group address

traffic channel: channel in which control/payload frames are exchanged asynchronously

uplink: transmission from MS to BS

user numbering: decimal representation of dPMR air interface addresses, as seen by the user, i.e. user visible numbering

telecommunication service: offered by a dPMR entity in order to satisfy a specific telecommunication requirement

tele-service: type of telecommunication service that provides the complete capability, including terminal equipment functions, for communication between users

NOTE: Individual voice calls and talkgroup voice calls are examples of tele-services.

User-plane (U-plane): part of the protocol stack dedicated to user voice services

vocoder socket: 216 bits vocoder payload

wildcard: character in the user domain that represents all digits 0 to 9

3.2 Symbols

For the purposes of the present document, the following symbols apply:

B_2	algorithm that converts MS dialable talkgroup addresses between the User Interface and the Air Interface
dBm	absolute power level relative to 1 mW, expressed in dB
dBp	Power relative to the average power transmitted over a burst in decibel
Hz	frequency
Eb	Energy per bit
ms	milli-seconds
No	Noise per Hz
ppm	parts per million

3.3 Abbreviations

For the purposes of the present document, the following abbreviations apply:

4FSK	Four-level Frequency Shift Keying
ACK	ACKnowledgment
AI	Air Interface
BCD	Binary Coded Decimal
BS	Base Station
CC	Colour Code
CCH	Control CHannel
CLI	Caller Line Identity
COCHIn	CO-CHannel Identity n (n = 1 to 15)
C-plane	Control-plane
CRC	Cyclic Redundancy Checksum

NOTE: For data error detection.

CSF	Configured Services and Facilities
dPMR	digital Private Mobile Radio
e.r.p	effective radiated power
FDMA	Frequency Division Multiple Access
FEC	Forward Error Correction
FN	Frame Numbering
GPS	Global Positioning System
HI	Header Information
ID	Identifier
IP	Internet Protocol
IPV	Internet Protocol Version
ISF	Initial Services and Facilities
IT	Information Technology
LBT	Listen Before Transmit
MI	Message Information
MID	Manufacturers ID
MMI	Man Machine Interface
MS	Mobile Station
MSs	Multiplicity of mobile or handportable Stations
NACK	Negative ACKnowledgment
OACSU	Off Air Call Set Up
PABX	Private Automatic Branch eXchange
PC	Personal Computer

PDF	Packet Data Format
PDU	Protocol Data Unit
PL	Physical Layer
PMR	Private Mobile Radio
PSTN	Public Switched Telephone Network
PTT	Push-To-Talk
RF	Radio Frequency
RSSI	Received Signal Strength Indication
SLD	SLow Data
SYNC	SYNChronization
TCH	Traffic CHannel
UDT	Unified Data Transport
U-plane	User-plane

4 Overview of dPMR

The digital Private Mobile Radio (dPMR) protocol employs a Frequency Division Multiple Access (FDMA) technology in an RF carrier bandwidth of 6,25 kHz.

There are two types of dPMR equipment.

- TS 102 490 [i.1] describes low-cost peer to peer terminals in licence exempt spectrum.
- TS 102 658 [i.2] describes terminals and base station equipment for the professional market offering both peer-to-peer and repeater operation.

Business and industry users have a basic need for flexible, efficient and cost effective communications systems and this was the fundamental reason for the development of the dPMR standards.

In technical terms these requirements can be all met by using a low-latency, dPMR protocol employing a suitable quality vocoder. The coding gain is used to recover good quality audio at the coverage boundary rather than to extend the range to distances not achievable by analogue systems at the same transmit power.

dPMR is intended to be an enhancement that existing analogue users will most likely wish to take advantage of in the near term. It is assumed that the preferred approach will be to locate these new schemes on their existing frequency assignments wherever possible and in any event to be within the allocated land mobile service bands. Therefore, in preparation for this, every effort has been undertaken to ensure that the digital protocol complies with the harmonized spectrum regulation, the adjacent channel performance, and be carefully adjusted to not disturb the existing spectrum planning by excessive ranges being achieved in the field. Thus, the proposed protocol is designed to fit into the existing regulatory environment and spectrum planning assumptions with an absolute minimum of disruption.

The dPMR protocol is required to support a very wide variety of applications. Many users will continue to require customized solutions. However, it is recognized that in some instances, users will require units from a variety of suppliers, perhaps fulfilling different needs within the same overall operational environment. To assist this, the technical specifications on dPMR in TS 102 490 [i.1] and TS 102 658 [i.2] have been written to define an agreed list of specific features and facilities. These standards provide sufficient detail to allow them to be implemented in a consistent way and therefore ensure interoperability. To confirm the correct implementation of these features conformity testing documents (PICS) have been produced for both air interface standards, TS 102 587-1 [i.12] for licence-free equipment and TS 102 726-1 [i.9] for licensed equipment.

The extent to which interoperability can be applied is limited because the existing market has many different operational procedures that these units should comply to and not disrupt. For example, unlike some other communications schemes, it is not the case that a standardized numbering and dialling system can be universally employed. This is because some important customers already have methods of operating that include absolute requirements that have implications on dialling sequences. It would therefore not be possible to create a single dialling plan that would be acceptable to all users. For users such as these it will be necessary to address their requirements, perhaps on a case-by-case basis. An introduction to and addressing is presented in clauses 4.8 and the Standard User Interface in clause 4.9.

Equipment compliant to the dPMR protocol is able to realise a fully functional radio system that offers voice and data capability. dPMR terminals and base station can seamlessly integrate with IP networks and offering customisable application software that will have tailored solutions to business communication needs up and running in minimal time. Combined with the true low-cost approach that is the core of the dPMR protocol the overall result is a practical solution for users and network operators alike. Although the dPMR protocol is sub-divided into licence exempt and licensed (with three modes) much of the Air Interface is common, therefore manufacturers development costs are reduced.

4.1 Licence Exempt dPMR

This is a licence exempt version of dPMR. Because it is licence exempt, the transmit power is limited and only handheld MS with fixed antennae are permitted. The requirements for the licence exempt dPMR protocol are described in TS 102 490 [i.1].

Despite these limitations, dPMR446 is appropriate for many small commerce's and recreational users who can exploit both text messaging and voice calls using the voice and data functionality of dPMR.

There are two levels of functionality (services and facilities) that can be offered by the equipment. For the purposes of interoperability, a basic level of services and facilities (ISF) is defined along with a simplified mode of addressing such that all MS will be capable of interoperating without the need for any set-up or programming at the point of sale. An advanced level of services and facilities (CSF) is also defined for those equipments that can be re-programmed to offer a higher level of functionality.

Where manufacturers have declared compliance to the "Standard User Interface" for CSF MSs, TS 102 490 [i.1], annex A, provides the MMI.

4.2 Licensed dPMR

Not all business and industry users have the same needs. For this reason dPMR has been sub-divided into 3 separate functional modes. The licensed dPMR protocol is described in TS 102 658 [i.2].

Where manufacturers have declared compliance to the "Standard User Interface", TS 102 658 [i.2], annex A, provides the MMI.

4.2.1 Licensed dPMR Mode 1

There are many commercial users such as building sites, shops, hotels, sports centres etc where the use of repeaters is not required. dPMR Mode 1 offers the optimum radio functionality for such local area coverage.

Base station (non repeater type) MSs could still be interconnected to IT networks via IP and exploit both text messaging and voice calls using the voice and data functionality of dPMR.

4.2.2 Licensed dPMR Mode 2

This is the dPMR solution for normal business and industry users or local government services where a normal repeater coverage area is adequate. Where needed, dPMR Mode 2 can also provide wide area coverage using its special co-channel multi-repeater functionality. dPMR radio equipment can be seamlessly integrated into existing IT networks with both text messaging and voice calls using the voice and data functionality of dPMR. Solutions already exist for PC based remote control of dPMR base stations to give completely configurable dispatcher functionality.

What always remains is the advantage for dPMR that a single repeater failure results in the loss of one voice channel not two or more.

dPMR Mode 2 also offers users the possibility to operate efficiently in 'direct mode' separately from the network or beyond the coverage area of the network for special purposes such as on-scene activities.

4.2.3 Licensed dPMR Mode 3

This is the dPMR solution for either very large business and industry users or government services where wide area multi-site, multi-channel trunked repeaters can offer up to national or international coverage if required. dPMR radio equipment can be seamlessly integrated into existing IT networks with both text messaging and voice calls using the voice and data functionality of dPMR. Solutions already exist for PC based remote control of dPMR base stations to give completely configurable dispatcher functionality.

dPMR Mode 3 also offers users the possibility to operate efficiently in 'direct mode' separately from the network or beyond the coverage area of the network for special purposes such as on-scene activities.

4.3 Services and Facilities

More recently, the professional environment has undergone a change whereby old operational models are no longer-applicable in many cases. This has meant that the operational requirements placed on communication equipment have evolved, and the traditional analogue service is no longer able to meet the users' needs completely. It is therefore appropriate that more sophisticated services are made available which will meet this need. This raises the need for a technology enhancement that allows the PMR model (which remains very attractive in many regards) to support the basic and enhanced features and facilities existing and future users will require.

Industry research has indicated that in the event that certain key facilities can be provided, it may be expected that a significant improvement in the current market performance of this service can be expected. There are only a relatively small number of such features and facilities that are needed. However, these will dramatically change the value that the users can derive from the equipment and services.

The main user required features are:

Summary of Features:

a) Improved audio quality in weak signal conditions.

b) Improved battery endurance.

c) Better range performance (this is taken to mean a good quality of service out to the range boundary rather than much greater absolute range).

d) Security of communication.

dPMR is recognized as having specific advantages when used in applications relating to public services and similar environments. These are rarely quantified in economic terms due to the complexity of making such an analysis. However, due to the importance of these uses, it is important to recognize how the introduction will improve the operational efficiency of the service achieved. Here are a small number of examples by way of illustration.

1) Security Services

   The introduction of digital signalling greatly facilitates the inclusion of location and status services such as GPS. This could easily be integrated with automatic units providing details of status at particular locations under this security umbrella. The end impact to the security organization is greatly improved awareness of the location of all the security personnel and much faster response to incidents or other unusual situations. This in turn leads to improved levels of security and also improves the safety of the individuals involved.

2) Site Safety

   The introduction of significantly improved emergency facilities through reverse channel signalling means that an immediate notification can be sent to site personnel that an incident is in progress. This can be accompanied by data giving further details. It is equally possible to interrupt the current communication to pass the information by voice if so desired.

   This can have extremely important safety implications in very high noise or low-visibility environments because having a hands-free possibility may encourage the use of headsets and similar accessories.

3) Local Government and Social Services

Location information, coupled with status information can more easily be accumulated and sent back to other officers. This allows them a better ability to respond to incidents or perhaps aid co-workers who are in dangerous situations.

The superior signalling allows a very large degree of automation at the application level to be employed. This therefore offers the potential of having much improved operation with only small headcount implications.

4) Utilities

Maintenance workers in the field can be supported with much improved information through the signalling capability while maintaining the important closed user group structure. This information cannot currently be reliably provided through the analogue systems.

5) Specific Public Safety Applications

Whilst many public safety organizations are moving to sophisticated schemes, there remain some organizations whose needs are not so complex.

Typically, these users already have an analogue scheme and are seeking to upgrade to a scheme that meets their current and future needs. It may be that dPMR with this level of signalling may provide a suitable platform for their use.

4.4 Interoperability

The dPMR protocol is required to support a very wide variety of applications. Many users will continue to require customized solutions. However, it is recognized that in many instances, users will require units from a variety of different suppliers. To assist this, the technical specifications on dPMR in TS 102 490 [i.1] and TS 102 658 [i.2] for applications and interoperability have been created that defines an agreed list of specific features and facilities that are to be implemented and give sufficient detail to allow them to be implemented in a consistent way. This ensures that the necessary interoperability is achieved. To confirm the correct implementation of these features, conformity testing documents have been produced as the TS 102 726 multipart standard [i.9], [i.10] and [i.11].

4.5 Frequency Considerations

dPMR equipment is designed to be used in land mobile frequency bands in Europe. The ECC Decision (06) CC [i.8] addresses the use of the bands 68 MHz to 87,5 MHz, 146 MHz to 174 MHz, 406,1 MHz to 430 MHz, and 440 MHz to 470 MHz which are planned for narrow band applications within the land mobile service.

NOTE: dPMR tier II and tier III can be operated in all frequency ranges of as indicated in ERC Report 25 [i.6] wherever a dedicated frequency range is implemented by the national radio authorities. Other frequency ranges than identified in the ERC report can be dedicated to PMR in countries outside of Europe. The dPMR radio system is designed to operate in part of the RF frequency range of 30 MHz to 1 GHz.

4.5.1 dPMR systems compliant with TS 102 490

TS 102 490 [i.1] defines digital PMR446 applications operating in the frequency band 446,1 MHz to 446,2 MHz compliant with TS 102 490 [i.1] operating with limited functionality that offers only simplex, peer-to-peer voice and data communication. This mode is suitable for low cost licence exempt operation. The ECC Decision (05) 12 [i.7] addresses the use of this band. Channel spacing is 6,25 kHz.

4.5.2 dPMR systems compliant with TS 102 658

The ECC Decision (06) 06 [i.8] addresses the use of the bands 68 MHz to 87,5 MHz, 146 MHz to 174 MHz, 406,1 MHz to 430 MHz, and 440 MHz to 470 MHz which are planned for narrow band applications within the land mobile service. The term Narrow Band Digital Land Mobile PMR/PAMR is intended to cover dPMR digital systems.

The technical provisions for channelling are detailed in CEPT Recommendation T/R 25-08 [i.5] and both of the following implementations in clauses 4.5.2.1 and 4.5.2.2 comply with these requirements. Effectively this means that either users could be licensed for an individual 6,25 kHz channel or an existing 12,5 kHz licence could be split between two users of 6,25 kHz dPMR.

4.5.2.1 6,25 kHz raster

dPMR terminals operate within channel rasters compliant with EN 301 166-2 [i.4] operating with 6,25 kHz channel spacing as illustrated in figure 4.1.

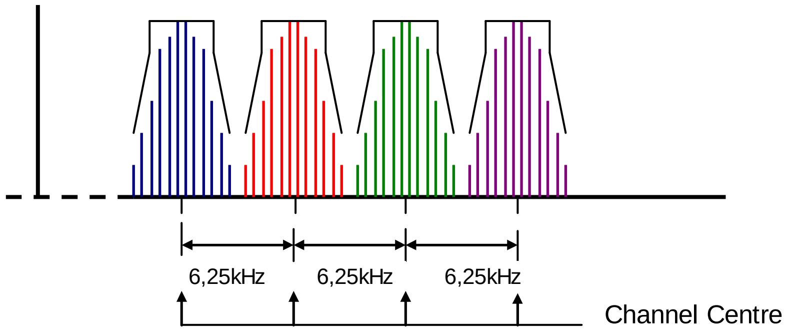


Figure 4.1

4.5.2.2 12,5 kHz raster

dPMR terminals operate in a 12,5 kHz channel raster, offset from the channel centre by offset by either +3,125 kHz or -3,125 kHz as illustrated in figure 4.2.

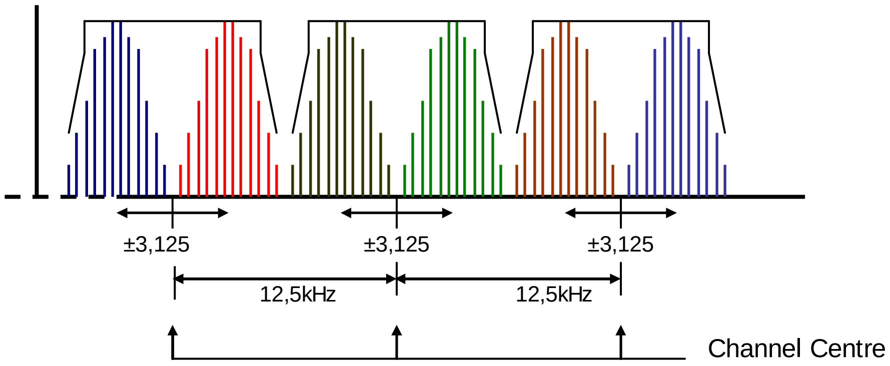


Figure 4.2

4.6 Protocol architecture

4.6.1 Architectural Configurations

The dPMR protocol has the flexibility to support from simple MS peer-to-peer operation to large trunked networks operating over a wide area. In order to separate the features and facilities of this wide ranging flexibility, a network of MS and/or BS are configured into one of three modes, Mode 1, Mode 2 or Mode 3. Within a network all entities are configured with the matching mode.

Entities employ a colour code. Colour Codes may be individually assigned by channel for spectrum management purposes or to differentiate different systems sharing a physical radio channel(s).

4.6.1.1 Peer-to-Peer (Licence exempt)

A Peer-to-Peer Direct Network illustrated in figure 4.3 is characterized by multiple MS communicating with each other directly on a single frequency channel (i.e. MS f_{tx} = MS f_{rx} = f_1) and compliant with TS 102 490 [i.1].

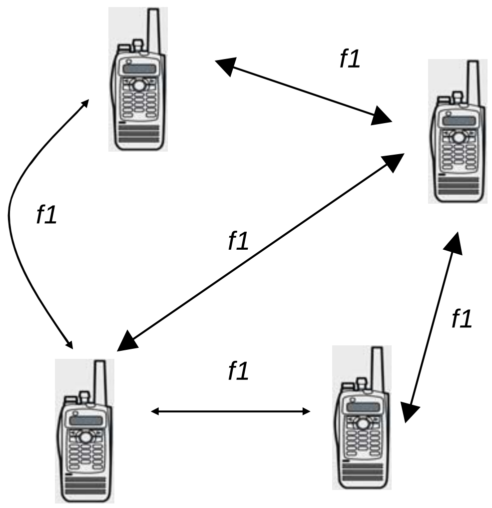


Figure 4.3: Peer-to-Peer Direct Network (Licence Exempt)

While a MS is partied to a voice call, it may transmit irrespective of whether the channel is "Idle" or "Busy" with 6,25 kHz FDMA activity pertaining to the same voice call but may not transmit if a Tx WAIT time has been invoked. However, for all other situations including data transmissions, MS are configurable to employ the following levels of "politeness" on a channel:

- Polite to own Colour Code: The MS refrains from transmitting on a channel while the channel is "Busy" with other 6,25 kHz FDMA activity from radios using the same Colour Code.
- Impolite: The MS transmits on a channel regardless of any other activity (either 6,25 kHz FDMA or otherwise) already present on the channel.

4.6.1.1A Peer-to-Peer Direct Network (Licensed Mode 1)

A Peer-to-Peer Direct Network illustrated in figure 4.4 is characterized by multiple MS communicating with each other directly on a single frequency channel (i.e. MS f_{tx} = MS f_{rx} = f_1) compliant with TS 102 658 [i.2].

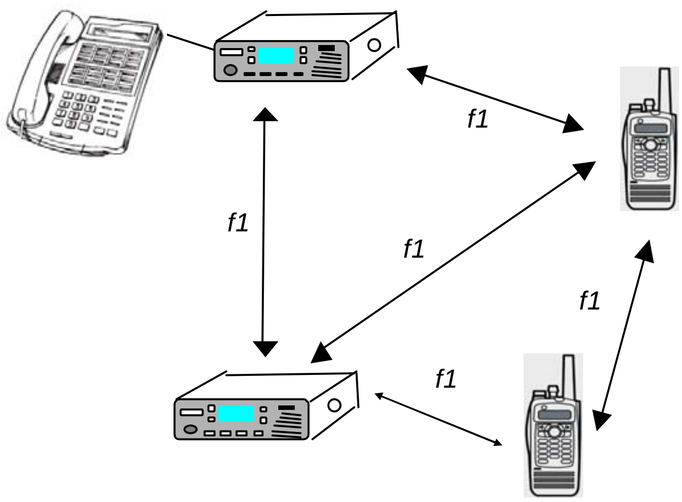


Figure 4.4: Peer-to-Peer Direct Network (Licensed)

Peer-to-Peer operation on a given channel is governed by the MS on that channel. There is no 'Master-Slave' relationship on such a channel and each MS is responsible for adhering to the channel access rules. Peer-to-Peer communication is directly between the MS.

Signalling between entities is asynchronous using a traffic channel.

4.6.1.2 Centralized Repeater Network (Licensed Mode 2)

A Centralized BS Network illustrated in figure 4.5 is characterized by multiple MS communicating with a BS on up-link and down-link channels (i.e. MS f_{tx} = BS f_{rx} = f_{uplink}, MS f_{rx} = BS f_{tx} = $f_{downlink}$) compliant with TS 102 658 [i.2]. All Centralized communication is via the BS. For polite operation, the BS is required to indicate on the down-link when the up-link is busy.

Signalling between entities is asynchronous using a traffic channel.

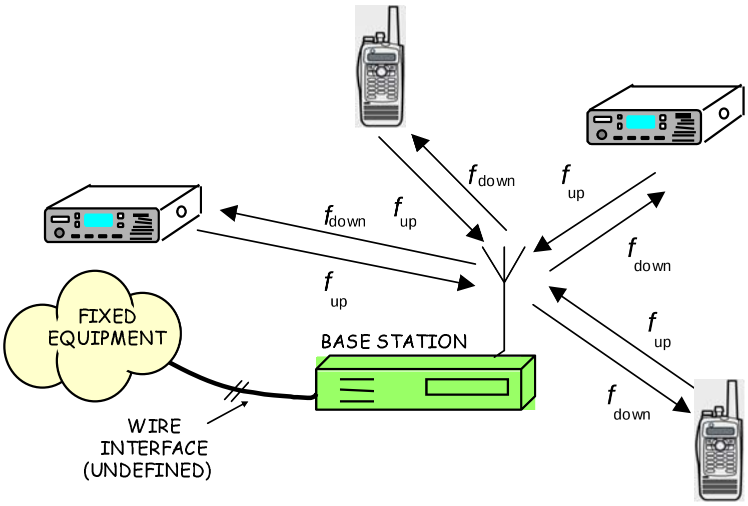


Figure 4.5: Centralized Repeater Network (Mode 2)

4.6.1.3 Managed Centralized Repeater Network (Licensed Mode 3)

A Managed Centralized BS Network illustrated in figure 4.6 is characterized by multiple MS communicating with a BS on up-link and down-link channels (i.e. MS f_{tx} = BS f_{rx} = f_{uplink}, MS f_{rx} = BS f_{tx} = $f_{downlink}$) compliant with TS 102 658 [i.2]. There is a 'Master-Slave' relationship on such a channel where the BS is considered the Master and the MS are considered the Slaves. All Centralized communication is via the BS.

A Mode 3 physical channel may be operating as a beacon channel or a traffic channel.

4.6.1.3.1 Beacon Channel

Signalling between entities is synchronous. Frames are transmitted by the BS to provide MS bit and slot timing. All call set-ups use a beacon channel.

By default, MS employ Random Access to access the channel, however the channel access rules may be modified at any time by the BS regulating channel access or implementing the role of a polling station. The BS is required to implement intelligent signalling functions such as indicating on the down-link when the up-link is busy.

4.6.1.3.2 Traffic Channel

For some services (such as voice) the BS and MS either switches to traffic channel operation or transfers to the call to an alternative BS that is activated as a traffic channel.

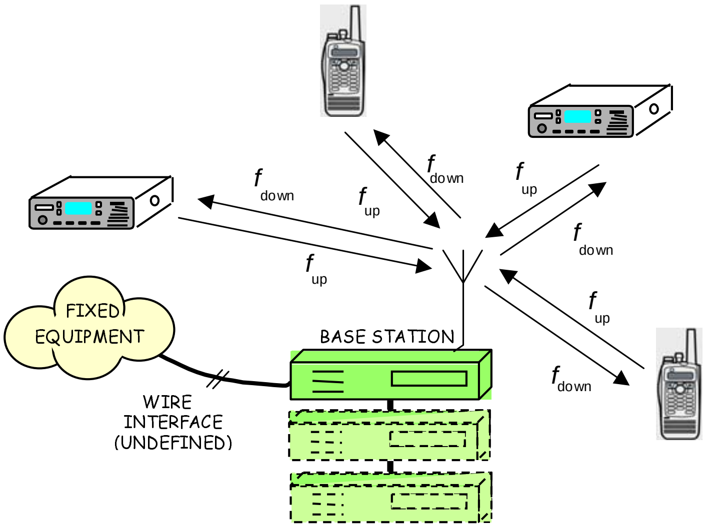


Figure 4.6: Managed Centralized Repeater Network (Mode 3)

4.6.1.4 Co-channel BS networks

Where geographical radio coverage is extended by multiple co-channel BSs, the system may operate by using a poll and vote call sequence. In all cases it is the MS that makes the assessment of the received signals to select the optimum BS.

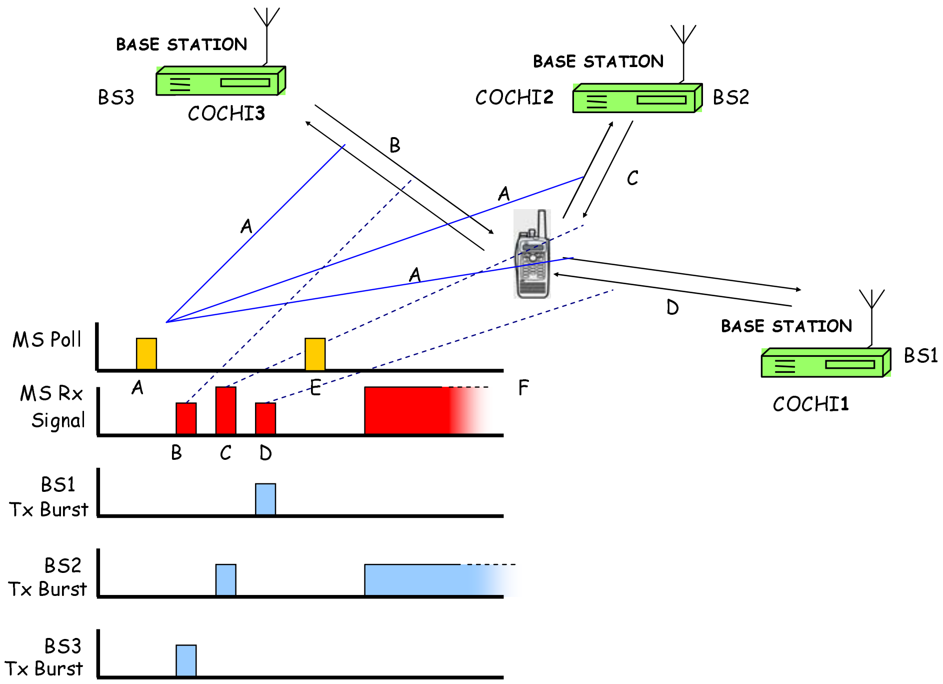


Figure 4.7: Co-channel Base Station networks

A network employing three co-channel BSs is illustrated in figure 4.7. An MS wishes to select the BS that will provide the best signal quality for the call.

Referring to the illustration in figure 4.7:

a) The MS makes an initial polling call to all BSs within range.

b) The BS with the highest assigned co-channel address (COCHI3 in this example) sends a response to the poll message. The timing of the poll message is determined by the particular COCHI index number.

c) The BS assigned as COCHI2 sends a response to the poll message.

d) The BS assigned as COCHI1 sends a response to the poll message.

e) The MS assess the signal quality of each of the poll responses. In this example, BS2 has the best signal quality. The MS then sends an acknowledgement to the gateway address COCHI2.

f) BS2 then asserts its carrier transmitting protection frames until the MS transmits its first call set-up or payload burst.

4.6.2 dPMR services overview

Table 4.1: Mode 1 Mode 2 Services

Bearer services	Tele-services	Supplementary services
Voice	Individual Call	Late Entry
		OACSU
		Cancel call set-up
		PTT call
		Slow user data
		Short Attached_Data
		Talking Party Identification
	Call to a talkgroup	Late Entry
		All Call
		PTT Call
		Slow user data
		Short Attached_Data
		Broadcast Call
		Talking Party Identification
Type 3 data	IP over dPMR	
	Individual Data Message	
Type 2 data	IP over dPMR	
	Individual Data Message	
	Data Message to a talkgroup	
Type 1 data	IP over dPMR	
	Individual Data Message	
	Data Message to a talkgroup	
Status Polling	Individual Status Polling	
Short Data	Short Data Delivery	

Table 4.2: Mode 3 Services

Bearer services	Tele-services	Supplementary services
Voice	Individual Call	Late Entry
		OACSU
		Cancel call set-up
		PTT call
		Slow user data
		Short Attached_Data
		Talking Party Identification
		Call Diversion
		Call Back
	Call to a talkgroup	Late Entry
		All Call
		PTT Call
		Slow user data
		Short Attached_Data
		Broadcast Call
		Talking Party Identification
Type 3 data	IP over dPMR	
	Individual Data Message	
Type 2 data	IP over dPMR	
	Individual Data Message	
	Data Message to a talkgroup	
Type 1 data	IP over dPMR	
	Individual Data Message	
	Data Message to a talkgroup	
Status Polling	Individual Status Polling	
Short Data	Individual Short Data Delivery	
	Short Data Delivery to a talkgroup	
Short Data Polling	Short Data Polling	

4.6.2.1 Call types

The dPMR protocol is able to offer the equipment designer a number of features that enhance the basic voice and data services.

a) Voice calls directed either to an individual MS, a group of MS, all MSs in a system, broadcasts to groups or all MS.

b) Voice with Slow Data (SLD) - when a PTT item is carrying voice payload, the superframe is also available to carry slow speed data.

c) Voice with Attached Data - If a MS releases the PTT before a superframe has completed the remaining traffic channel frames may carry attached data.

d) A choice of conventional and packet data calls.

4.6.2.1.1 Parties Involved in the Call

4.6.2.1.1.1 Individual call

The Individual Call service provides voice service between one individual user and another individual user. The individual call is made to a unique address that is not identified as a group address within an MS that is part of a system.

For equipment compliant with the Standard User Interface, an individual call is a call made to a dialable address that does not contain any "wildcard" characters.

4.6.2.1.1.2 Group call

The Group Call service provides voice call service between one individual user and a predetermined group of users. All parties in the group can hear each other. A group call is a call made to an address that is identified as a group address within one or more MSs that is part of a system.

For equipment compliant with the Standard User Interface, a group call is a call made to a dialable address using "wildcard" characters to define talkgroups.

A broadcast is a group call service whereby only the calling party is permitted to speak.

4.7 Colour Codes

The colour code is a means for dPMR entities to segregate an individual network of users from other networks of users that may be sharing a common physical radio channel. Sixty four colour codes are defined.

4.7.1 Colour Codes for TS 102 490

Colour Code are attributed directly to the RF operating channel and are not freely selectable. They are split into group 'A' and group 'B'.

For the purposes of interoperability and to differentiate the different modes of addressing used, radios employing Initial Services and Facilities (ISF) use the Group A colour codes only and radios employing Configured Services and Facilities (CSF) use the Group B colour codes only.

4.7.2 Colour Codes for TS 102 658

Colour Codes may be individually assigned by channel for spectrum management purposes or to differentiate different networks of users sharing a physical radio channel(s).

Where no specific Colour Code has been programmed for a physical channel, entities may determine the Colour Code applicable for the frequency by the following algorithm:

CC number = 64 x (f modulo 0,4) where f is the channel freq in MHz.

4.8 Addressing

All entities (MS, BS etc) within a particular network are assigned a unique individual ID. MSs may also be assigned one or more group identities to form a talkgroup. MSs and talkgroups use a 24 bit Identity.

Other entities connecting to MS and BS conforming to the present document may employ different addressing formats. As an example, PSTN destinations may be described by a string of numeric digits. An IP address may be defined by a 32 bit (IPV4) or a 128 bit address (IPV6). These destinations are defined as extended addresses.

When many different types of entity are linked in a particular system, a way of identifying these entities is essential The present document uses reserved addresses called Gateway Addresses that identity both destinations and certain intrinsic call services.

4.9 Standard User Interface

It is recognized that manufacturers of MSs may wish to exercise design independence in their products and, accordingly, the requirements of these annexes are only applicable to equipment where the manufacturer has declared compliance with the "Standard User Interface". The Standard User Interface defines:

a) the user visible numbering (User Interface domain); and

b) dialling in an MS for accessing other MS(s) over the AI; and

c) how the visible user numbering and dial strings may be mapped on to the AI.

As not to restrict manufacturer's independence, it is envisaged that dialling selection may be initiated in many ways. Some methods are:

a) direct number entry via a keypad;

b) mode selection buttons; and

c) soft key menu selection.

The dialling method may vary according to the MS terminal type. This annex is applicable to MSs with a basic CCITT number keypad, as illustrated in figure A.1 and/or with a display capable of displaying the decimal numbers "0" to "9" and the keys "*" and "#". However, manufacturers may employ other keypad layouts.

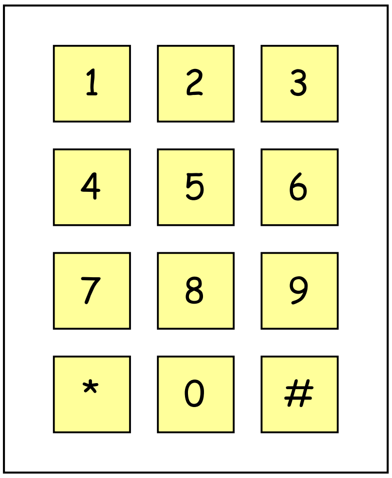


Figure 4.8: CCITT keypad layout

The primary use for the keypad is to enable the user to select the destination address, the type of service, and to initiate calls from the MS. Certain other services may be requested by dialling "call modifier" strings prior to entering the destination address.

1) the user dials digits; and

2) user initiates the call.

Dialled digits are represented in decimal notation and utilize the numbers "0" to "9" and the keys "*" and "#". For an MS fitted with a keypad, the "#" key may initiate a call (although other initiate methods may be implemented by a manufacturer). Dialled digits that represent a destination address are translated to a form for the Air Interface by a coding algorithm. This is illustrated in figure 4.9.

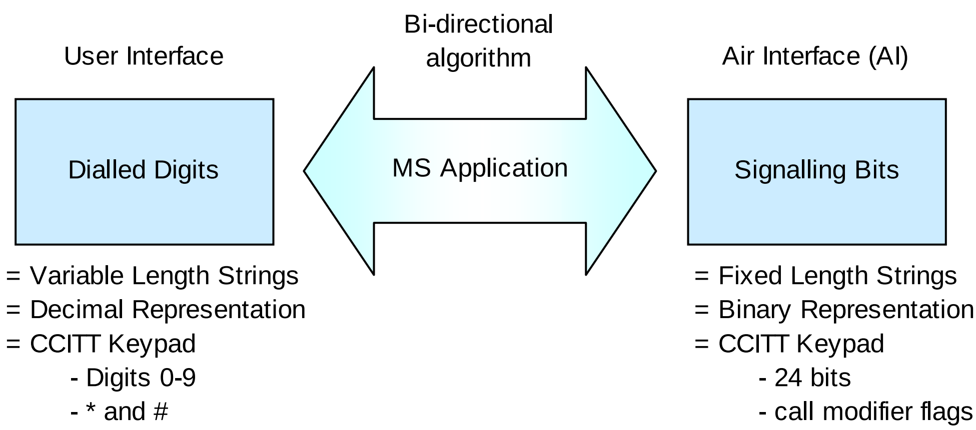


Figure 4.9: Number conversion

4.9.1 The concept of the wildcard character

The MS may discriminate a talkgroup call from an individual call by the use of the "wildcard".

In the User Interface domain structure, if the dialled string represents an MS address, and contains a "*" in any of the four least significant dialled characters, then that MS address represents a talkgroup of MSs. The "*" character is the "wildcard" and represents all numeric values in that digit position. For example, if the user dials "012345*", this means that the MS is addressing 10 separate MSs whose individual addresses are "0123450", "0123451", "0123452", "0123453", "0123454", "0123455", "0123456", "0123457", "0123458", and "0123459".

4.10 Unified Data Transport Mechanism

A dPMR system supports a wide range of facilities. To support these facilities, the transporting of data is a very common necessity. For example, although Short Data is a primary Mode 2 and Mode 3 data service, there are many instances where data needs to be transported to support other facilities. (For example when an MS dials a PABX or PSTN destination, the dialled digits are uploaded to the BS).

To reduce the dPMR complexity, all short data, extended addressing and complementary data transport between MS and BS share this common method - the Unified Data Transport mechanism.

In Mode 1 and Mode 2 systems Appended_Data Messages may be concatenated to Connection_Request messages. Mode 3 systems concatenate Appended_Data carrying short data, extended addresses and complementary data to a UDT Header message.

The data in these Appended_Data messages are coded in a uniform way and support dPMR addresses, binary, BCD, 7 bit text, 8 bit octets, a common GPS format (EN 61162-1 [i.3]) and IP addressing.

5 Channel Access Mechanisms

dPMR equipment may be employed in a diverse range of both licensed and licence exempt spectrum. To satisfy these differences, various channel access mechanisms are specified that impose certain levels of politeness. Channel access is either by random access or fully regulated.

5.1 Random Access (Mode 1, Mode 2)

By default, MS employ a Random Access method to access channels. This method provides a polite and organized protocol for MS to access the channel by ensuring that:

a) MS refrain from accessing a channel which is already in use. This uses a simple listen before transmit.

b) MS access a channel in a way which minimizes collisions (resulting from simultaneous transmissions).

c) Collisions are resolved in an orderly manner.

d) Emergency calls are given priority over non-emergency calls.

5.2 Regulated Random Access (Mode 3)

MS channel access on a given channel is regulated by a Managed Repeater (Mode 3). All MS not currently involved in a call or transaction listen to a control channel that manages all access. This Centralized control is a particularly useful mechanism for improving the throughput of heavily utilized channels.

5.3 Listen Before Transmit (LBT)

When accessing a channel to transmit, a MS takes account of the following types of activity which may already be present on the channel:

- 6,25 kHz FDMA activity;
- other digital protocol activity;
- analogue activity.

When determining whether activity is present on a channel, the radio monitors the RSSI level. If after a maximum period of time the RSSI level has not exceeded a configurable (within a predefined range) threshold, then the MS assumes that activity is not present on the channel.

If however the RSSI level does exceed this threshold, then the radio assumes that activity is present on the channel and it attempts to become frame synchronized to the activity.

If the radio is successful in becoming frame synchronized to the activity, then the MS assumes that 6,25 kHz FDMA activity is present on the channel. If the Colour Code is different then the MS assumes that the activity is interference. If however after a maximum period of time, the radio has not become frame synchronized to the activity, then the MS assumes that the activity is non-6,25 kHz FDMA activity.

5.4 Hang time messages and timers

5.4.1 Definition

A voice call consists of a series of speech items separated by gaps known as "call hang time periods".

As the protocol is inherently asynchronous, these gaps will be of random duration but it is possible for a radio involved in a group or talkgroup call to define a minimum call hang time period by transmitting a Tx_WAIT time in the end frame of the speech item.

5.4.2 Action by receiving stations

When a transmitting MS involved in a group or talkgroup call announces a non zero Tx_WAIT time then PTT activated transmissions are not be permitted to start during this Tx WAIT time irrespective of any polite or impolite criteria employed. This gap in transmission enables break in requests from MS not involved in the call.

Where a radio receives a break-in request during the announced Tx Wait time then there is an audible prompt to the user to leave the RF channel free for the station that has requested to speak.

Break-in requests are permitted for group and talkgroup calls. They are not permitted for individual calls or All Calls.

A user that wishes to break-in and use the RF channel pre-key a break-in request on their MS. That MS will not transmit the request until the start of the announced Tx WAIT time.

5.4.3 Call duration timers

dPMR MSs have a transmit TimeOut timer which limits the time of a single transmission item. In TS 102 490 [i.1] this is a fixed timer value of 180 seconds whenever the PTT key is pressed and counts down to zero.

If the transmit TimeOut timer expires, then all HSs will stop transmitting immediately and may not re-transmit until PTT has been released and pressed again.

TS 102 658 [i.2] permits this timer to be configurable.

5.5 Transmit admit criteria

5.5.1 General admit criteria

Where a radio has been solicited to transmit a response, it may transmit the response within response time [T_ack] irrespective of whether the channel is "Idle" or "Busy".

5.5.1.1 ISF admit criteria

While a radio is partied to a voice call, it may transmit irrespective of whether the channel is "Idle" or "Busy" with 6,25 kHz FDMA activity pertaining to the same voice call but may not transmit if a Tx WAIT time has been invoked. However, for all other situations including data transmissions, MS are configurable to employ the following levels of "politeness" on a channel:

- Polite to own Colour Code: The MS refrains from transmitting on a channel while the channel is "Busy" with other 6,25 kHz FDMA activity from radios using the same Colour Code.
- Impolite: The MS transmits on a channel regardless of any other activity (either 6,25 kHz FDMA or otherwise) already present on the channel.

5.5.1.2 CSF admit criteria

While a radio is partied to a voice call, it may transmit irrespective of whether the channel is "Idle" or "Busy" with 6,25 kHz FDMA activity pertaining to the same voice call but may not transmit if a Tx WAIT time has been invoked. However, for all other situations including data transmissions, MS are configurable to employ the following levels of "politeness" on a channel:

- Polite to own Group or Talkgroup: The MS refrains from transmitting on a channel while the channel is "Busy" with other 6,25 kHz FDMA activity from radios within its own group or talkgroup. For all other types of activity already present on the channel, the MS transmits regardless.
- Polite to own Colour Code: The MS refrains from transmitting on a channel while the channel is "Busy" with other 6,25 kHz FDMA activity from radios using the same Colour Code. For all other types of activity already present on the channel, the MS transmits regardless.
- Impolite: The MS transmits on a channel regardless of any other activity (either 6,25 kHz FDMA or otherwise) already present on the channel.

On a given channel, not all features may be supported the same level of politeness. So for example, voice transmissions may be configured to be "impolite" while packet data transmissions are configured to be "polite".

5.5.1.3 Random Access (Licence exempt, Mode 1, Mode 2)

By default, MS employ a Random Access method to access channels. This method provides a polite and organized protocol for MS to access the channel by ensuring that:

a) MS refrain from accessing a channel which is already in use.

b) MS access a channel in a way which minimizes collisions (resulting from simultaneous transmissions).

c) Collisions are resolved in an orderly manner.

d) Emergency calls are given priority over non-emergency calls.

5.5.1.4 Regulated Random Access (Mode 3)

MS channel access on a given channel is regulated by a Managed Repeater (Mode 3). Channel access is regulated while a payload transaction is not in progress in order to provide a Centralized control of the channel access. This Centralized control is a particularly useful mechanism for improving the throughput of heavily utilized channels.

5.5.1.5 Polling

For Polling applications, MS channel access is in response to transmissions generated by a Central entity (i.e. the Polling Station).

Polling is applicable both to Peer-to-Peer and Centralized operation, and where employed, the role of Polling Station is either implemented by an MS (Peer-to-Peer operation) or the BS (Centralized operation).

5.5.1.6 Beacon Signal

A Mode 3 BS transmits a Beacon Signal on a given channel in order to provide one or more of the following features:

a) Mark the presence of a system.

b) Radiate system parameters.

c) Provide timing information (common clock, timeslot timing, frame timing, etc.).

d) Provide signal strength information.

e) Invite MS to instigate a call service.

5.6 FDMA Structure

This clause describes the frame structure for dPMR. Not all structures are employed to build a dPMR system and only a sub-set are defined for TS 102 490 [i.1].

5.6.1 Overview of transmission and burst structure

dPMR is based on a FDMA structure.

The physical resource available to the radio system is an allocation of the radio spectrum.

A transmission or burst is a period of RF carrier that is modulated by a data stream. The physical channel of an FDMA transmission is required to support the logical channels.

A logical channel is defined as a logical communication pathway between two or more parties. The logical channels represent the interface between the protocol and the radio subsystem. The logical channels may be separated into two categories:

- traffic channels carrying control frames, speech or data payload (Mode 1, Mode 2, Mode 3); and
- beacon channels (Mode 3).

NOTE: A Mode 3 system employs a beacon channel for call set-up and beacon transactions. For some services (such as voice calls) the beacon channel may revert to a traffic channel or the beacon channel may transfer the call to a separate physical traffic channel for the duration of the call.

All traffic channel transmissions are asynchronous, since there is no entity to provide frame or slot timing.

All beacon channel transmissions are synchronous and rely on a BS to provide slot timing.

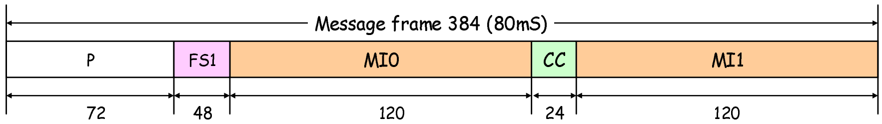
Peer-to-peer, uplink, and downlink messages are distinguished by a two bit Communications Format field that is carried in all message frames.

5.6.2 Transmission format

dPMR transmissions follow the formats in these clauses.

5.6.2.1 Traffic Channel Message Frame

The traffic channel message frame illustrated in figure 5.1 is of 80 ms (384 bits) in length.



P: Preamble, minimum of 72 bits
FS1: 48 bit Frame Sync 1 sequence
MI0: Message 0, 120 bits
CC: Colour Code, 24 bits
MI1: Message 1, 120 bits

Figure 5.1: Traffic Channel Message Frame

A beacon channel message frame has a very similar structure illustrated in figure 5.2.

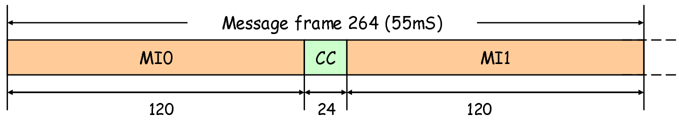
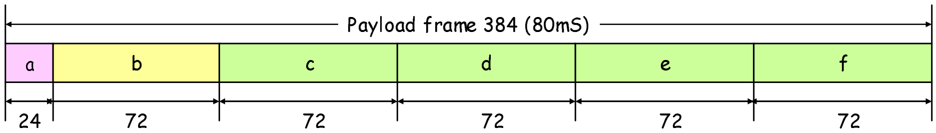


Figure 5.2: Beacon Channel Message Frame

5.6.2.2 Traffic Channel Payload Frame

An FDMA traffic channel payload transmission illustrated in figure 5.3 is made up of 80 ms payload frames, each comprising 384 bits.



a: 24 bits FrameSync2 (FS2) or Colour Code (CC) bits
b: 72 bits Control Channel (CCH) data
c: 72 bits Traffic channel (TCH)
d: 72 bits TCH
e: 72 bits TCH
f: 72 bits TCH

Figure 5.3: Payload Frame

5.6.2.2.1 Traffic Channel Superframe

Four 80 ms payload frames illustrated in figure 5.4 are concatenated to form a superframe of 320 ms.

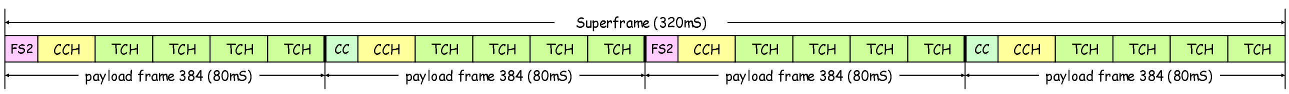
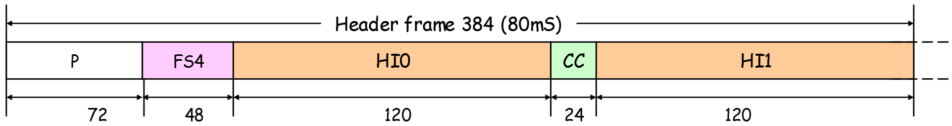


Figure 5.4: Superframe

5.6.2.2A Traffic Channel Packet Data Header Frame

The Header frame illustrated in figure 5.5 is of 80 ms (384 bits) in length.

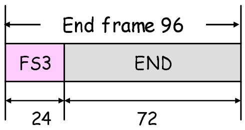


P: Preamble, minimum of 72 bits
FS4: 48 bit Frame Sync 4 sequence
HI0: Header Information 0, 120 bits
CC: Colour Code, 24 bits
HI1: Header Information 1, 120 bits

Figure 5.5: Packet Data Header Frame

5.6.2.3 Traffic Channel End Frame

The End frame illustrated in figure 5.6 is a shortened 96 bit frame.



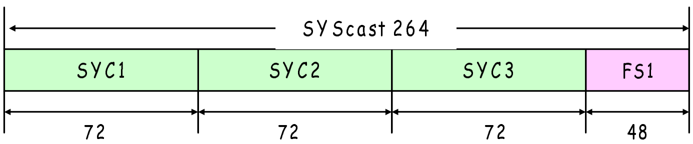
FS3: Frame sync, 24 bits
END: End data, 72 bits

NOTE: Type 3 data transmissions (packet data) use a different framing structure.

Figure 5.6: End Frame

5.6.2.4 Beacon SYScast Frame

The SYScast frame illustrated in figure 5.7 is transmitted by a Mode 3 beacon BS.



SYC1: SYScast1, 72 bits
SYC2: SYScast2, 72 bits
SYC3: SYScast3, 72 bits
FS1: Frame sync, 48 bits

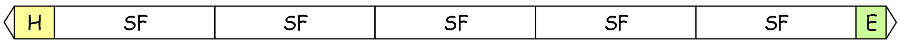
Figure 5.7: SYScast Frame

5.6.3 Transmission sequences

5.6.3.1 Traffic Channel Voice or data payload item transmission

The sequence is illustrated in figure 5.8. These transmissions are always started with a Header frame containing a preamble (for bit synchronization) and a frame synch (for frame synchronization). The Header is followed by a series of Superframes that contain both the payload (voice or data) and the information about the call such that receiving stations can implement late entry. A call always consists of an integral number of superframes and is terminated by an End frame.

For receiving stations, purpose and content of any transmission can be determined by the Message Information (MI0 and MI1).



H: Header frame
SF: Superframe
E: End frame

Figure 5.8: Voice or Data Payload continuous transmission

5.6.3.2 Traffic Channel Call set up, service request, etc

The transmission illustrated in figure 5.9 may be sent by Mode 1 and Mode 2 systems on a traffic channel at the start of a call. They are a concatenation of a Header frame and an End frame. Their purpose is to inform the receiving station of the call, type of call or information required.

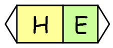


Figure 5.9: Call Set-up

The transmission may be sent for an individual call manually as a kind of 'polling call' to check if the called party is listening on the same channel.

These transmissions may be sent automatically by as the first part of an OACSU sequence or for initiating an individual data call.

5.6.3.3 Traffic Channel Acknowledgement:

Traffic channel acknowledgements are sent in response to applicable messages back to the originator. Acknowledgements are a type of Header that contains information such as confirmation of received data, errors in received data, etc.

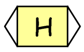


Figure 5.10: Acknowledgement

5.6.3.4 Traffic Channel Status request acknowledgements:

Traffic channel status request acknowledgements illustrated in figure 5.11 are sent by Mode 1 and Mode 2 systems. As the status information is contained within the End frame then the response of a receiving station to a status request call is a Header + End frame pair.

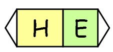


Figure 5.11: Status Request Acknowledgement

5.6.3.5 Traffic Channel Disconnection:

Sending stations can signal that all exchanges of a call have been completed by transmitting a disconnection request. This is a Header + End frame pair that is repeated illustrated in figure 5.12.

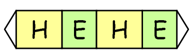


Figure 5.12: Disconnection

These transmissions may be sent manually as confirmation to the called party that the communication is complete.

These transmissions may also be sent automatically to the called party to indicate that an individual data call is completed.

5.6.3.6 Traffic Channel Preservation Message

These messages are transmitted by a Mode 2 or Mode 3 traffic channel BS to preserve the channel between MS items.

Figure 5.13: Preservation Frames

5.6.3.7 Mode 3 Beacon Channel

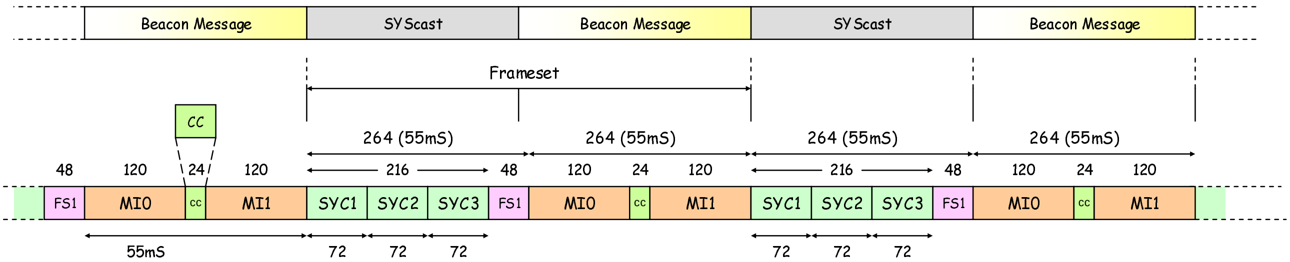


Figure 5.14: Beacon Channel

The Beacon Channel transmission is synchronous with a slot size of 264 bits. The slots alternate between beacon messages and SYScast broadcasts. One SYScast concatenated with a beacon message is a frameset.

6 Examples of Message Exchange for Calls

6.1 Parties Involved in the Call

6.1.1 Individual call

The Individual Call service provides voice service between one individual user and another individual user. The individual call is made to a unique address that is not identified as a group address within an MS that is part of a system.

For equipment compliant with the Standard User Interface, an individual call is a call made to a dialable address that does not contain any "wildcard" characters.

6.1.2 Group call

The Group Call service provides voice call service between one individual user and a predetermined group of users. All parties in the group can hear each other. A group call is a call made to an address that is identified as a group address within one or more MSs that is part of a system.

For equipment compliant with the Standard User Interface, a group call is a call made to a dialable address using "wildcard" characters to define talkgroups.

A broadcast is a group call service whereby only the calling party is permitted to speak.

6.2 Calls

6.2.1 Mode 1 Call Exchange

6.2.1.1 Mode 1 Voice Call

Figure 6.1 illustrates a Mode 1 voice call. This example shows the MS behaviour for a call to an MS talkgroup. The same behaviour may apply to an individual call where the calling party does not wish to first determine if the recipient of the call is in radio contact.

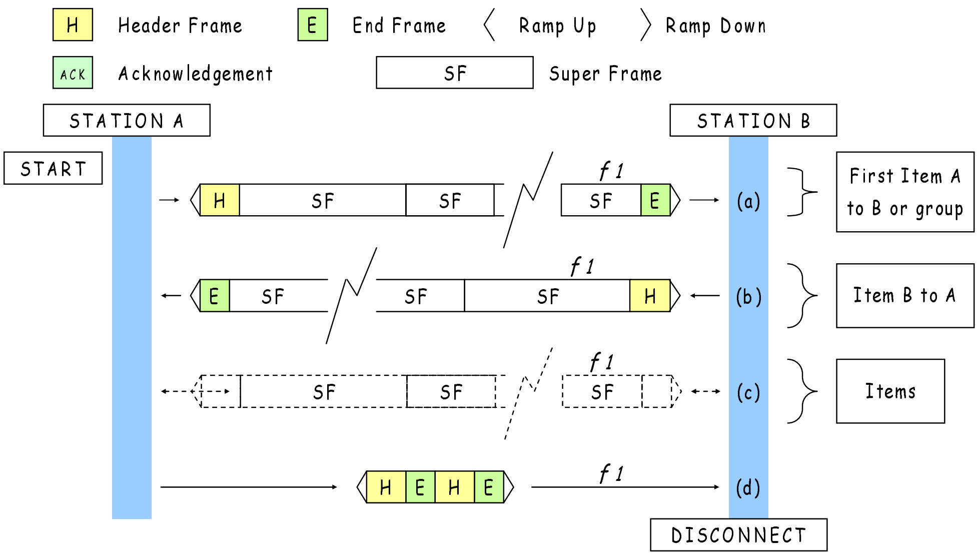


Figure 6.1: Mode 1 voice call message exchange

EXAMPLE 1:

The initial transmission from MS(A) is subject to polite access rules. If access is permitted then:

a) The sending station sends its first payload item to the talkgroup or individual recipient.

b) A payload item is returned to the sender.

c) Payload items continue.

d) When call is complete - if the call was to an individual MS either party may clear the call down; if the call was to a talkgroup only the initial calling party is permitted clear the call.

NOTE 1: The disconnect message at point (d) is optional.

Figure 6.2 illustrates an individual Mode 1 voice call with called party check. For this option, the calling party wishes to first determine if the recipient of the call is in radio contact before the call proceeds.

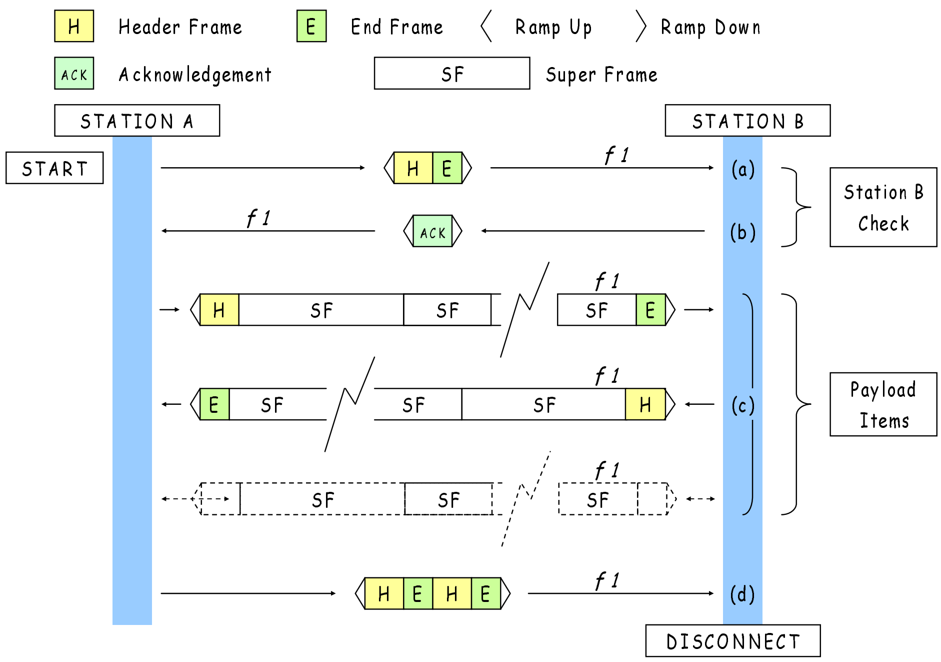


Figure 6.2: Mode 1 voice call exchanges with called party check

EXAMPLE 2:

The initial transmission from MS(A) is subject to polite access rules. If access is permitted then:

a) The sending station uses the call set-up (Header and End frames) to establish that the receiving station is within range and not busy.

b) When the receiving station has acknowledged with a T_ACK the sending station commences to send the first voice payload item.

c) Voice payload items continue.

d) When call is complete either party (but in this case the calling party) may end the call by sending a disconnect request to show that the transaction is complete.

NOTE 2: The disconnect message at point (d) is optional.

6.2.1.2 Mode 1 Data Call

Figure 6.3 shows an example of the exchanges involved in the call set-up and exchanges of an individual data call.

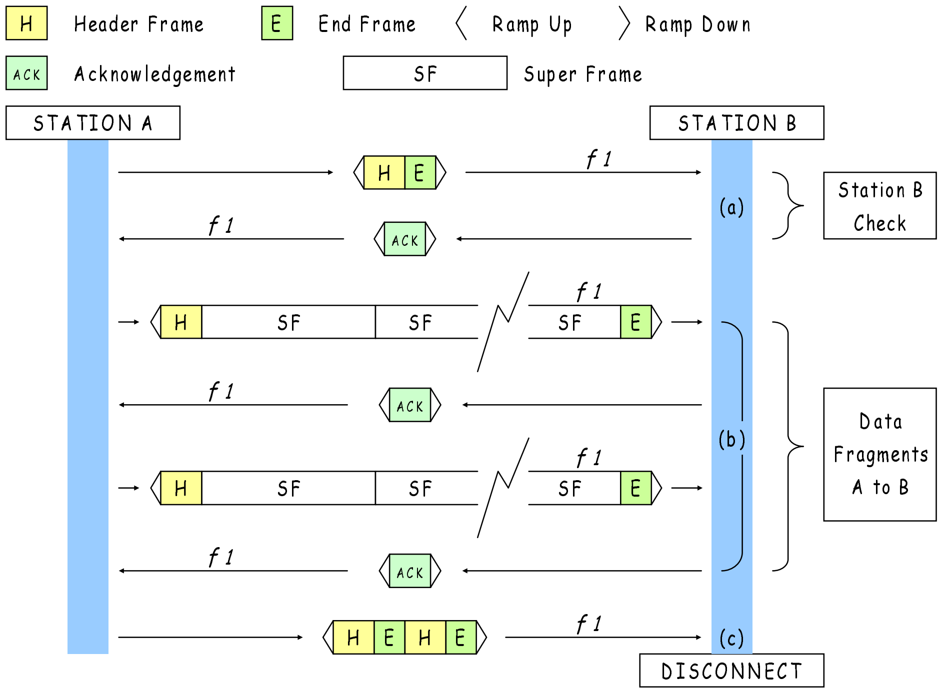


Figure 6.3: Mode 1 Individual data call exchanges

In this case:

The initial transmission from MS(A) is subject to polite access rules. If access is permitted then:

a) The sending station uses the call set-up (Header and End frames) to establish that the receiving station is within range and not busy. The receiving station acknowledges with a T_ACK.

b) The sending station commences to send the data in 4 superframe bursts. After each burst the receiving station decodes and error checks the data and if there are no uncorrectable errors a positive ACK is sent. If errors are detected then a negative ACK would be sent and the sending station would repeat that transmission.

c) When all the data has been transmitted and positively acknowledged the sending station sends a disconnect request to show that the transaction is complete.

6.2.2 Mode 2 Call Exchange

An example of a Mode 2 voice call message exchange is illustrated in figure 6.4. All Centralized communication is via the BS in Mode 2 systems.

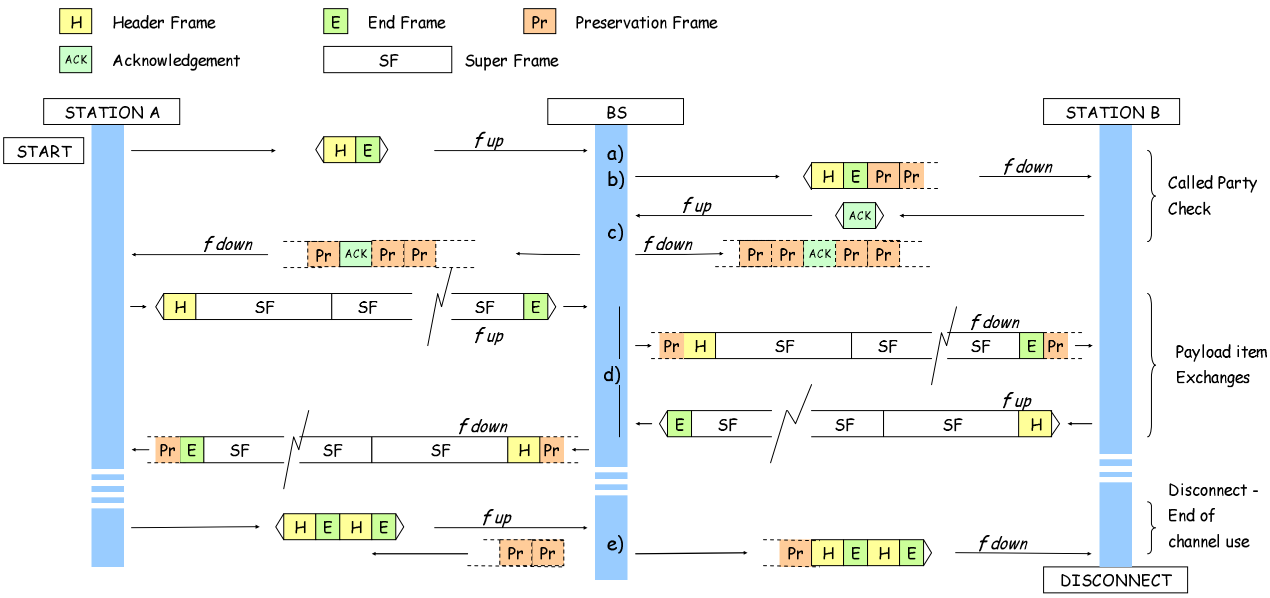


Figure 6.4: Mode 2 Voice Call Example

EXAMPLE:

The initial transmission from MS(A) is subject to polite access rules. If access is permitted then:

a) The sending station uses the call set-up (Header and End frames) to the BS on the uplink channel to establish that the receiving station is within range and not busy.

b) The BS retransmits the call set-up on the downlink channel to the receiving station. The BS then protects the traffic channel against access by MS not involved in the call by transmitting preservation frames.

c) When the receiving station has acknowledged with a T_ACK, the T_ACK is repeated by the BS to the sending station.

d) The MS exchange voice payload items.

e) When the call is ended BS(A) clears the call by transmitting Disconnect + END frame pairs. The message is retransmitted by the BS. The BS then returns to idle.

NOTE 1: There is an inherent delay between information received by the BS on the uplink channel and the BS retransmitting the information on the downlink channel.

NOTE 2: In the gap between transmission items, the Base Station transmits preservation frames to preserve the channel for the call.

NOTE 3: During the call, the retransmission from the BS is continuous. Preservation frames are transmitted when there are no MS originated messages to transmit. Unless an MS is transmitting, frames may be received that are directed to the other party. This is illustrated in figure 6.4 in the gaps between the payload items.

6.2.3 Mode 3 Operation

When idle, MSs listen to a beacon channel (see clause 5.6.3.7). All call services originate on this beacon with an exchange of call set-up messages. For some services such as voice, the MS participants in the call are transferred to a traffic channel for the transaction. When the call is complete the MSs return to the beacon channel.

An example of a Mode 3 call set-up is illustrated in figure 6.5. MS(A) and MS(B) is initially tuned to the Beacon Channel. This example illustrates a voice call set-up where the call is transferred to a traffic channel for the transaction.

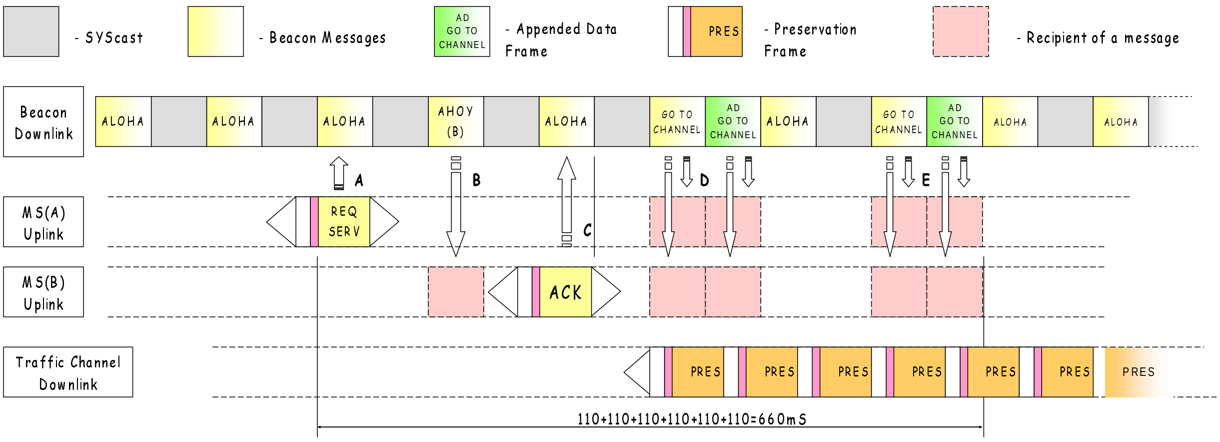


Figure 6.5: Mode 3 Beacon Channel Individual Voice Call Set-up

EXAMPLE 1:

The initial request for a transmission from MS(A) is permission subject to fully managed access rules. If access is permitted then:

a) The calling MS sends a service request to the beacon.

b) The beacon sends an ahoy message to MS(B) to determine if MS(B) is in radio contact.

c) MS(B) sends an acknowledgement to the beacon.

d) The beacon sends a Goto Channel message to MS(A) and MS(B) to direct the MSs to a traffic channel for the transaction. The system activates the traffic channel. The Goto Channel message contains the address of the called MS and the uplink and downlink frequencies of the traffic channel. The beacon may also concatenate an Appended Data message to the Goto Channel that contains the address of the calling MS.

e) Since the Goto Channel message is not acknowledged, the BS may repeat this message to the MSs.

If unacknowledged messages containing appended data are repeated there is at least one SYScast frame inserted between the two messages.

An example of a group call set-up is illustrated in figure 6.6.

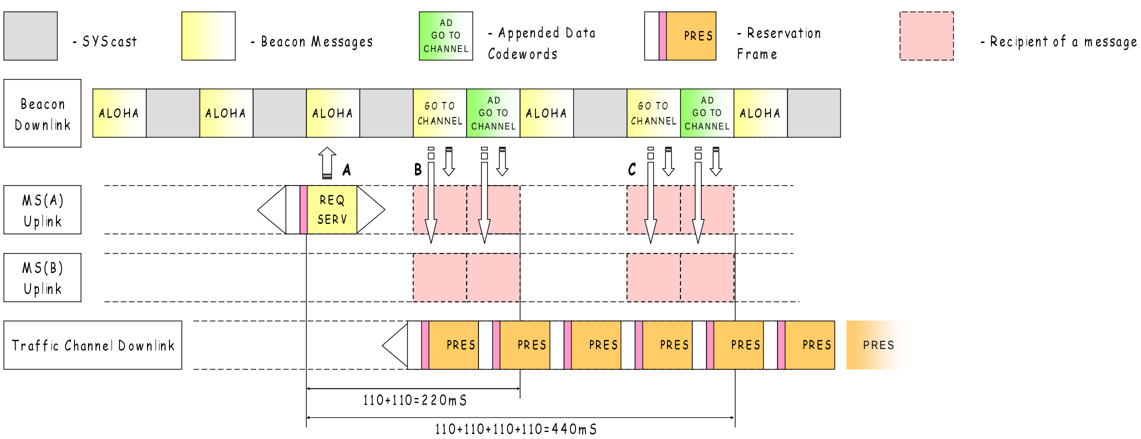


Figure 6.6: Talkgroup Voice Call-Setup

EXAMPLE 2:

The initial request for a transmission from MS(A) is permission subject to fully managed access rules. If access is permitted then:

a) The calling MS sends a service request to the beacon.

b) The beacon sends a Goto Channel message to MS(A) and MS(B) to direct the MSs to a traffic channel for the transaction. The system activates the traffic channel. The Goto Channel message contains the address of the called MS and the uplink and downlink frequencies of the traffic channel. The beacon may also concatenate an Appended Data message to the Goto Channel that contains the address of the calling MS.

c) Since the Goto Channel message is not acknowledged, the BS may repeat the message to the MSs.

If unacknowledged messages containing appended data are repeated there is at least one SYScast frame inserted between the two messages.

6.2.4 Packet data

6.2.4.1 Format

Packet data uses a different format to the normal communications frame format. The use of frame sync 4 (FS4) indicates that the frames following are in the PDF format.

The basic PDF format is illustrated in figure 6.7.

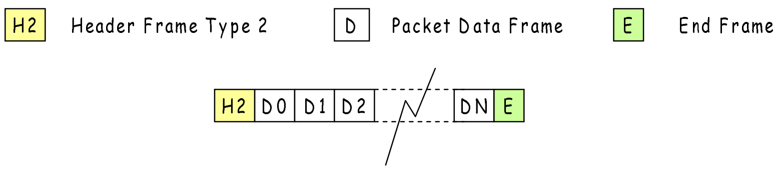


Figure 6.7: PDF format

Total length of a data frame D(N) = 80 x (pdS + 1) ms.

The value of pdS transmitted indicates the number of 80 ms frames.

Figure 6.8 illustrates concatenated PDF frames.

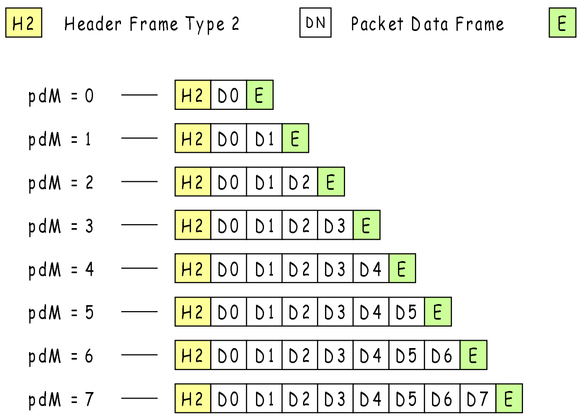


Figure 6.8: PDF frames

The value of pdM transmitted indicates the number of 320 ms frames.

The maximum transmission time of a single packet occurs when pdS = 3 and pdM = 7.

i.e. Header2 + (PDF max x pdM max) + END.

= 80 + (320 x 8) + 20 ms.

= 2 660 ms.

6.2.4.2 Standard Packet Exchange Format

A packet data call is illustrated in figure 6.9.

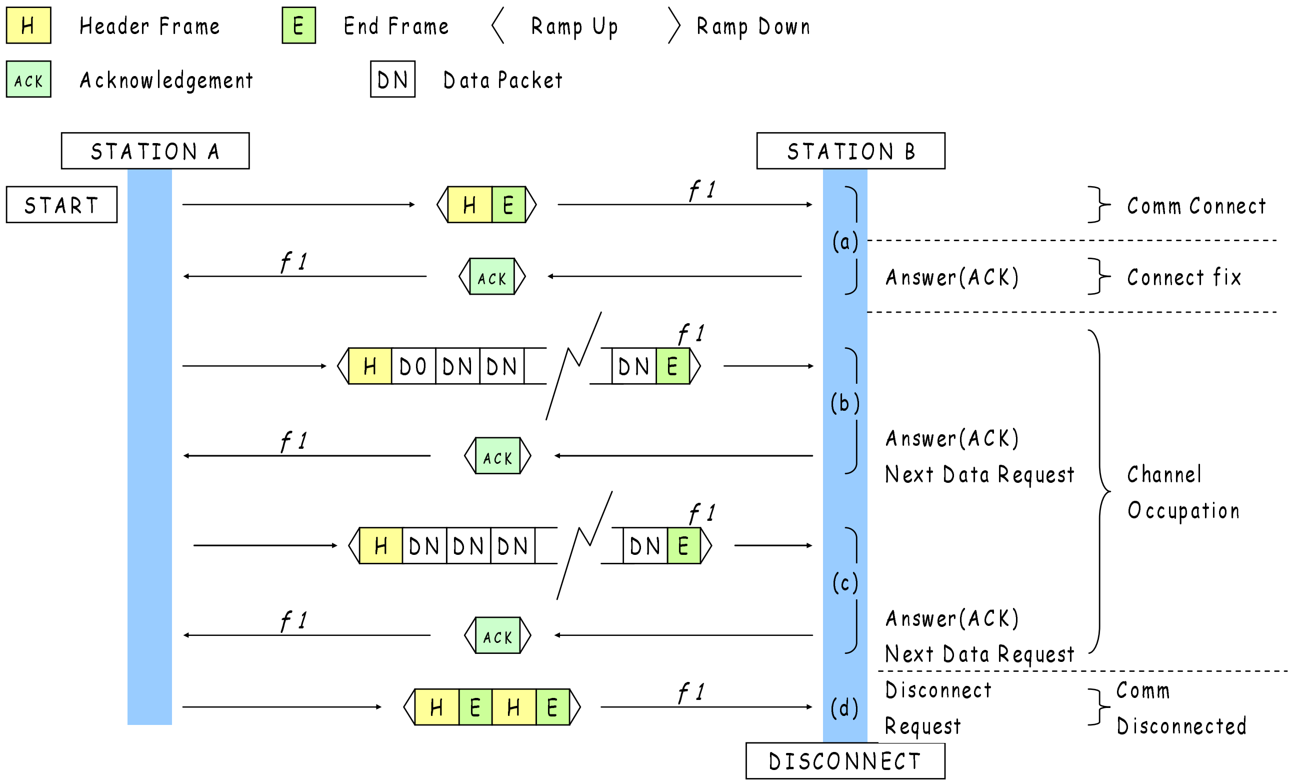


Figure 6.9: Packet exchanges

Station 'A' is conducting a packet data transaction with station 'B'. Station 'A' fragments its data message into suitable packets and chooses the most suitable value for pdS (packet size) and pdM (data a packets per transmission item).

Referring to figure 6.9:

a) Station 'A' attempts to establish a connection by transmitting a type 3 header frame/END. Station 'B' responds with a positive acknowledgement. If 'A' receives the acknowledgement the connection is established.

b) Station 'A' transmits the item and appends pdM packets to the header. Station 'B' acknowledges that the packets were received without any uncorrectable errors.

c) Assuming that station 'A' received the positive acknowledgement, station 'A' transmits the next item. Again the item is acknowledged.

d) When that data has been completely transmitted, station A send the disconnect request. Since the disconnect is not acknowledged, the header/end is repeated.

If a transmission item from station 'A' contains errors in some packets, the whole item does not need to be retransmitted. If station 'B' receives a transmitted item containing an error in one of the data packets, 'B' will send NACK to 'A' in response to the item. The NACK from station 'B' contains a field which indicates the packet that contains the first error detected.

Figure 6.10 illustrates a packet data call with an error.

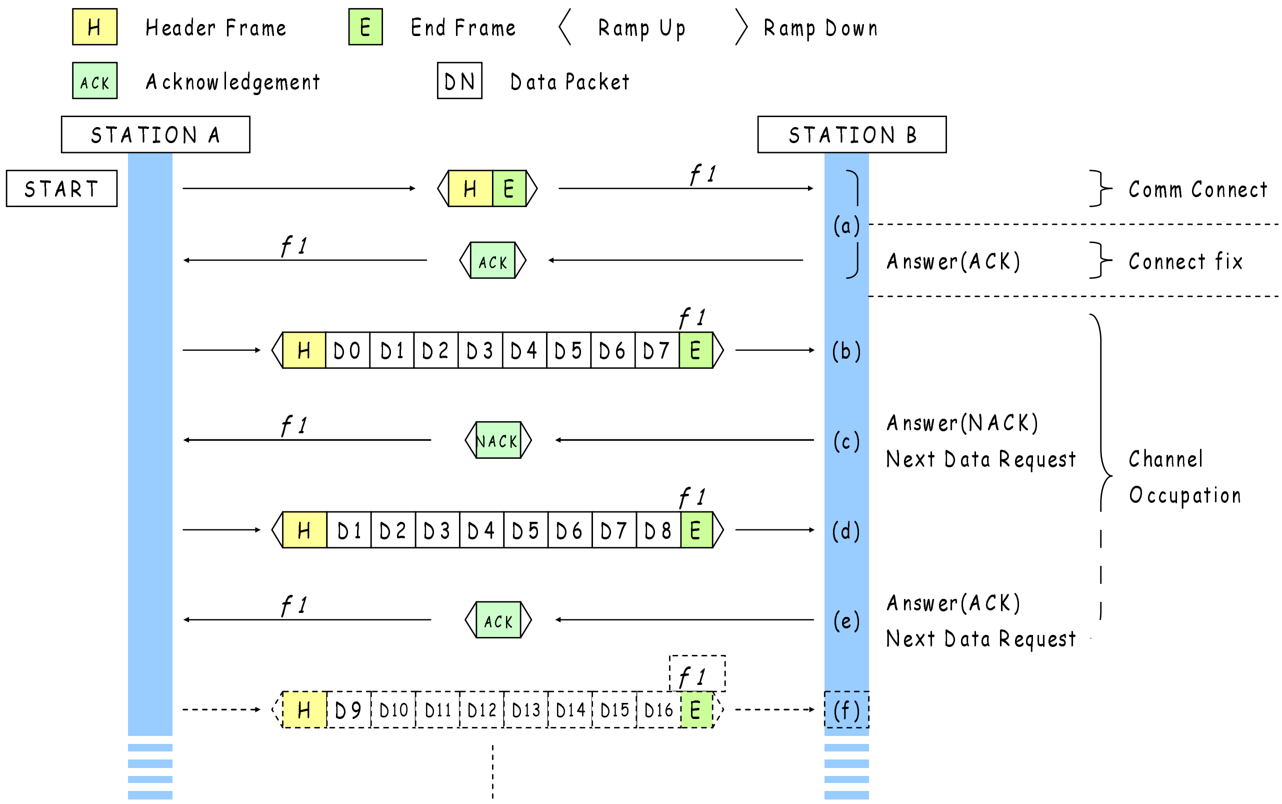


Figure 6.10: Packet retransmissions

Referring to figure 6.10:

a) Station 'A' attempts to establish a connection by transmitting a type 3 header frame/END. Station 'B' responds with a positive acknowledgement. Station 'A' receives the acknowledgement and the connection was established.

b) Station 'A' transmits the item and appends 8 (pdM=0111_2) packets to the header.

c) Station 'B' received the header and D0 correctly but D1 was received with errors. Station 'B' therefore transmitted a NACK (Type=010_2, Information = 0 asking for a retransmission from data packet #1.

d) Station 'A' transmits the item from data packet #1.

e) When that data has been completely transmitted, station 'A' send the disconnect request. Since the disconnect is not acknowledged, the header/end is repeated.

If Station 'A' has sent a packet and does not receive any acknowledgement, station 'A' may send a Repeat_Last_Ack + END message instead of repeating the packet data item. If station B receives a Repeat_Last_Ack + END message, it sends verbatim the acknowledgement that was previously sent.

7 Synchronization

The dPMR protocol relies on four separate frame synchronisation sequences to recognise dPMR PDUs and to distinguish between differing frame types.

7.1 Frame synchronization

7.1.1 FS1

The Frame Sync 1 sequence transmitted by MS and contained in the non packet data header frame (Header 1) is a 48 bit sequence that has the following value:

Binary: $0101\ 0111\ 1111\ 1111\ 0101\ 1111\ 0111\ 0101\ 1101\ 0101\ 0111\ 0111_2$.

Hex: $57\ FF\ 5F\ 75\ D5\ 77_{16}$.

7.1.2 FS2

The Frame Sync 2 sequence transmitted by MS and contained in the superframe (frames 1 and 3) is a 24 bit sequence that has the following value:

Binary: $0101\ 1111\ 1111\ 0111\ 0111\ 1101_2$.

Hex: $5F\ F7\ 7D_{16}$.

7.1.3 FS3

The Frame Sync 3 sequence transmitted by MS and contained in the End frame is a 24 bit sequence that has the following value:

Binary: $0111\ 1101\ 1101\ 1111\ 1111\ 0101_2$.

Hex: $7D\ DF\ F5_{16}$.

7.1.4 FS4

The Frame Sync 4 sequence transmitted by MS and contained in the Packet Data header frame (Header 2) is a 48 bit sequence that has the following value:

Binary: $1111\ 1101\ 0101\ 0101\ 1111\ 0101\ 1101\ 1111\ 0111\ 1111\ 1101\ 1101_2$.

Hex: $FD\ 55\ F5\ DF\ 7F\ DD_{16}$.

NOTE: FS4 is a symbol-wise complement of FS1. The frame sync correlator will find a positive result for FS1 and an equal but negative result for FS4 when running a single correlator.

8 Interleaving and FEC coding

Frames are protected by interleaving and FEC. Long strings of 0s or 1s are mitigated by scrambling.

8.1 CRC addition

Table 8.1: CRC coding

Use	CRC	Polynomial
Frame (CCH)	CRC7	$X^7 + X^3 + 1$
Message (MI and Header (HI)	CRC8	$X^8 + X^2 + X^1 + 1$

8.2 Hamming code

A shortened Hamming code (12,8) is employed and the generator matrix is illustrated in table 8.2.

$X^7,X^6,X^5,X^4,X^3,X^2,X^1,1$ are Identity bits (8 bit): C3,C2,C1,C0 are Parity bits (4 bit).

Table 8.2: Generator matrix

	12	11	10	9	8	7	6	5	4	3	2	1
	X^7	X^6	X^5	X^4	X^3	X^2	X^1	1	C3	C2	C1	C0
1	1	0	0	0	0	0	0	0	1	1	1	0
2	0	1	0	0	0	0	0	0	0	1	1	1
3	0	0	1	0	0	0	0	0	1	0	1	0
4	0	0	0	1	0	0	0	0	0	1	0	1
5	0	0	0	0	1	0	0	0	1	0	1	1
6	0	0	0	0	0	1	0	0	1	1	0	0
7	0	0	0	0	0	0	1	0	0	1	1	0
8	0	0	0	0	0	0	0	1	0	0	1	1

Shortened Hamming code (12,8) Polynomial: $X^4 + X + 1$.

8.3 Scrambling

The scrambling polynomial is $X^9 + X^5 + 1$ with an initial preset value of all "1"s.

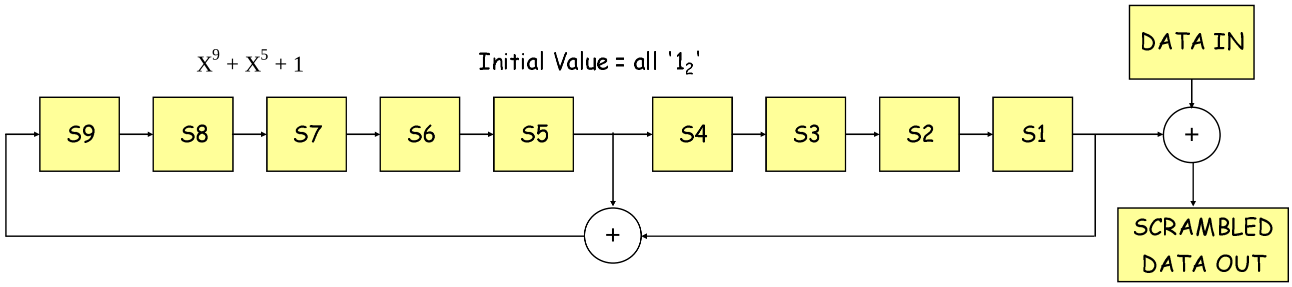


Figure 8.1: Scrambling format

8.4 Interleaving

There are two interleaving matrices, one for the TCH and one for the MI/HI field.

TCH interleave structure matrix:

Table 8.3: TCH Interleaving matrix

	1	2	3	4	5	6
1	1	13	25	37	49	61
2	2	14	26	38	50	62
3	3	15	27	39	51	63
4	4	16	28	40	52	64
5	5	17	29	41	53	65
6	6	18	30	42	54	66
7	7	19	31	43	55	67
8	8	20	32	44	56	68
9	9	21	33	45	57	69
10	10	22	34	46	58	70
11	11	23	35	47	59	71
12	12	24	36	48	60	72

The Interleave Structure Matrix Map (Tx side: 12 bit x 10).

Table 8.4: MI and HI field Interleaving matrix

	1	2	3	4	5	6	7	8	9	10
1	1	13	25	37	49	61	73	85	97	109
2	2	14	26	38	50	62	74	86	98	110
3	3	15	27	39	51	63	75	87	99	111
4	4	16	28	40	52	64	76	88	100	112
5	5	17	29	41	53	65	77	89	101	113
6	6	18	30	42	54	66	78	90	102	114
7	7	19	31	43	55	67	79	91	103	115
8	8	20	32	44	56	68	80	92	104	116
9	9	21	33	45	57	69	81	93	105	117
10	10	22	34	46	58	70	82	94	106	118
11	11	23	35	47	59	71	83	95	107	119
12	12	24	36	48	60	72	84	96	108	120
NOTE: Applied in the Header MI0/MI1 and HI0/HI1.										

Use of interleaving matrices:

- Transmit data is input to the matrix in vertical columns from top left to lower right. Data is output from the matrix in horizontal rows from top left to lower right.
- Receive data is input to the matrix in horizontal rows from top left to lower right. Data is output from the matrix in vertical columns from top left to lower right.

8.5 FEC coding of CCH (superframe)

There are a total of 41 bits of CCH data.

The 7 bit CRC checksum is added using the polynomial given in clause 8.1 giving a total of 48 bits.

These 48 bits are now separated into 6 bytes. Each byte is now coded by a shortened 12,8 Hamming Code (see clause 8.2) giving 6 x 12 bit blocks.

To protect against burst interference, these 6 x 12 bit blocks are now interleaved using the 12 x 6 TCH interleaving matrix given in table 8.4.

Then the interleaved CCH data is scrambled using the polynomial given in clause 8.3.

8.6 FEC coding of MI (message info') and HI (header info')

There are a total of 72 bits of MI/HI data.

The 8 bit CRC checksum is added using the polynomial given in clause 8.1 giving a total of 80 bits.

These 80 bits are now separated into 10 bytes. Each byte is now coded by a shortened 12,8 Hamming Code (see clause 8.2) giving 10 x 12 bit blocks.

To protect against burst interference, these 10 x 12 bit blocks are now interleaved using the 12 x 10 HI interleaving matrix given in clause 8.4.

Then the interleaved MI/HI data is scrambled using the polynomial given in clause 8.3.

8.7 FEC coding of END information

There are a total of 17 bits of END information.

The 7 bit CRC checksum is added using the polynomial given in clause 8.1 giving a total of 24 bits.

These 24 bits are now separated into 3 bytes. Each byte is now coded by a shortened 12,8 Hamming Code (see clause 8.2) giving 3 x 12 bit blocks. These 36 bits are now repeated and the total 72 bits are scrambled using the polynomial described in clause 8.3.

8.8 Channel Coding Process - Example

Separate coding processes are employed for voice, type 1 data, type 2 data, packet data, message coding, end frames and SYScast frames.

The example illustrated in this clause illustrates the channel coding process for a voice superframe and a voice superframe + attached data. All channel coding is described in TS 102 490 [i.1] and TS 102 658 [i.2].

8.8.1 Voice superframe

Construction of the voice superframe starts with CCH control channel data.

Frame Numbering (FN) is from 00_2 to 11_2 (1 to 4).

FN is followed by 12 bits of the called station address or own ID as follows:

The called station ID and own ID make a total of 48 bits. These bits are split into 12 bit blocks and one block is included in each of the 4 frames of the superframe:

- FN 00_2 includes the upper 12 bits of the called station ID.
- FN 01_2 includes the lower 12 bits of the called station ID.
- FN 10_2 includes the upper 12 bits of the own ID.
- FN 11_2 includes the lower 12 bits of the own ID.

The Communications Mode value is added that determines if slow data is being included within the voice superframe.

Two version bits are added.

The communications format bits are now added. Occasionally they may be set to 00_2 (all call) but this is a special case, similar to a broadcast.

The next bit is the Emergency Priority bit.

The next bit is the Preservation message bit. This bit will be used by BS downlinks only and MS sets this to 0.

This gives the total of 41 bits of CCH data.

The 7 bit CRC checksum is added using the dPMR CRC7 polynomial giving a total of 48 bits.

These 48 bits are now separated into 6 bytes. Each byte is now coded by a shortened 12,8 Hamming Code giving 6 x 12 bit blocks.

To protect against burst interference, these 6 x 12 bit blocks are now interleaved using the dPMR 12 x 6 TCH interleaving matrix.

Then the interleaved CCH data is scrambled using the dPMR standard scrambling polynomial.

The frame is completed by prefixing with either the 24 bits of FS2 (for frame numbers 00_2 or 10_2) or the 24 bits of Colour Code (frame numbers 01_2 or 11_2).

Finally the 4 x 72 bit blocks of Forward Error corrected vocoder data (TCH) are appended.

8.8.2 Voice + Attached data call

In each transmitted item the format is always that of a series of complete superframes (SF) with Header and End frames as illustrated in figure 8.2.

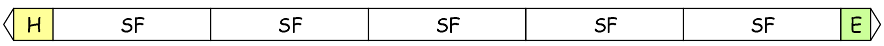


Figure 8.2: Transmitted Item

Within each superframe, there are 4 payload frames.

For this example illustrated in figure 8.3 it is assumed that the PTT is released in frame 2 and the voice codec data stops. 36 bytes of data with FEC (type 2) is attached. As each frame has a capacity of 20 bytes of type 2 data, both frames 3 and 4 are required.

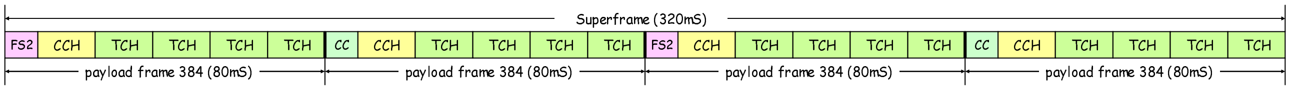


Figure 8.3: Transmitted Item Example

A diagram with the complete voice frame coding process is illustrated in figure 8.4.

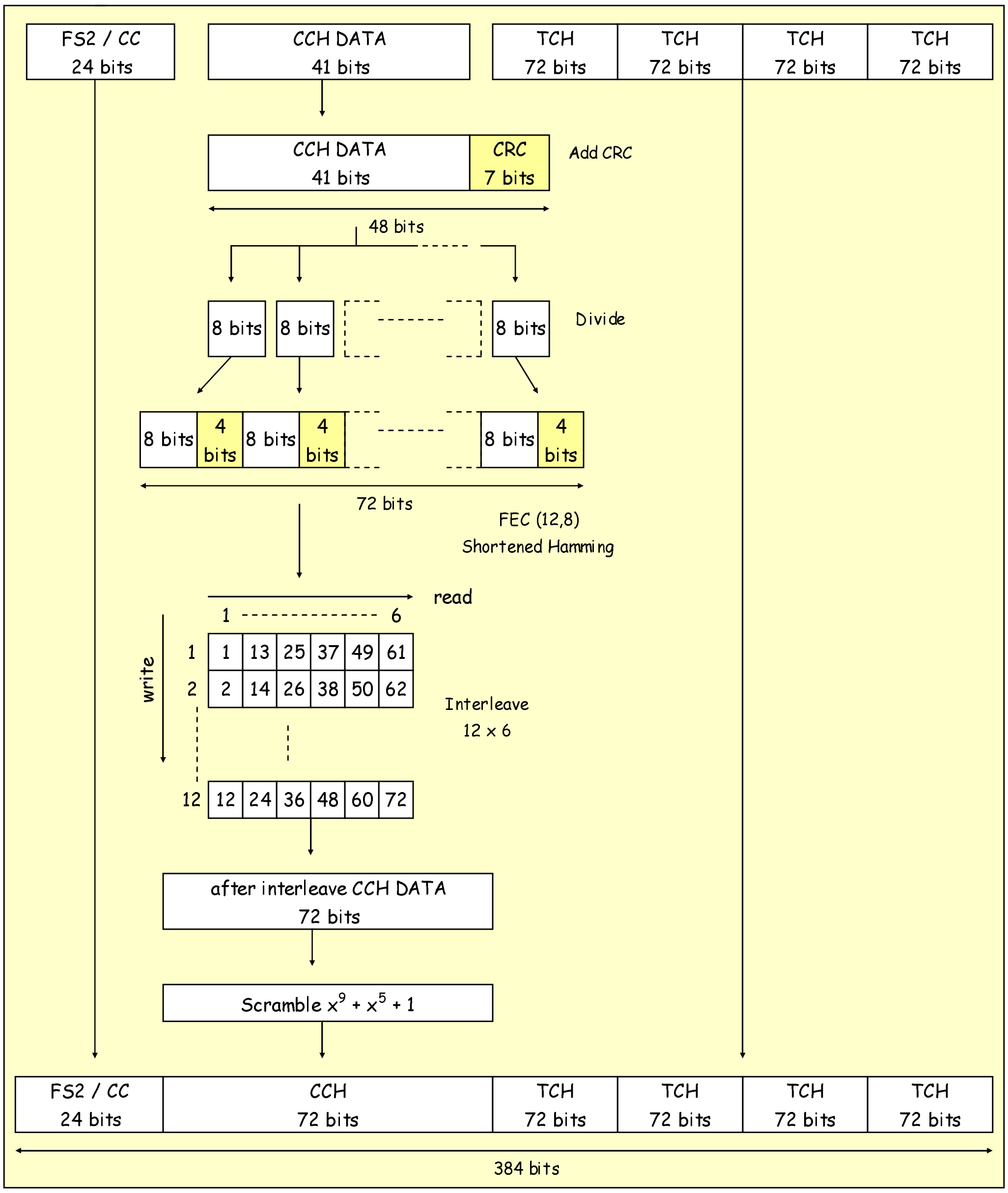


Figure 8.4: Voice frame coding

9 Physical Layer

9.1 General parameters

The MS complies with the essential requirements as stated in EN 301 166-2 [i.4].

9.1.1 Frequency range

9.1.2 RF carrier bandwidth

The radio system operates within a 6,25 kHz RF carrier bandwidth.

9.1.3 Transmit frequency error

The maximum transmit frequency error from the assigned RF carrier centre is within ±625 Hz as stated in EN 301 166-2 [i.4].

9.1.4 Time base clock drift error

The maximum time base clock drift error is be ±2 ppm. This error is the amount of clock drift that is acceptable during a transmission.

9.2 Modulation

9.2.1 Symbols

The modulation sends 2 400 symbols/sec with each symbol conveying 2 bits of information. The maximum deviation, D, of the symbol is defined as:

$$D = 3h/2T$$

Where:

- h is the deviation index defined for the particular modulation; and
- T is the symbol time (1 / 2 400) in seconds.

9.2.2 4FSK generation

This clause describes the characteristics of the constant-envelope modulation, entitled 4FSK.

9.2.2.1 Deviation index

The deviation index, h, for 4FSK is defined to be 0,29. This yields a symbol deviation of 1 050 Hz at the symbol centre. The mapping between symbols and bits is given in table 9.1.

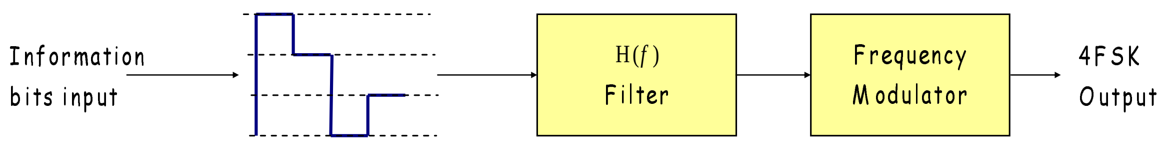
Information Bits Symbol Mapping to 4FSK Deviation.

Table 9.1: FSK symbol mapping

Information Bits		Symbol	4FSK Deviation
Bit 1	Bit 0		
0_2	1_2	+3	+1 050 Hz
0_2	0_2	+1	+350 Hz
1_2	0_2	-1	-350 Hz
1_2	1_2	-3	-1 050 Hz

9.2.2.2 Square root raised cosine filter

*Tx Baseband Filter

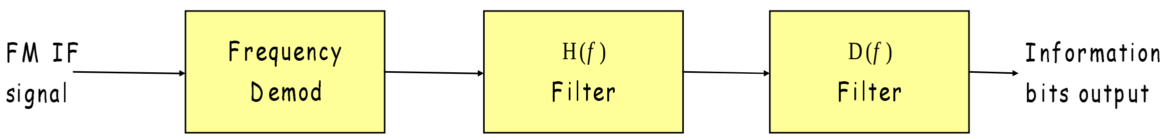


$$|H(f)| = \begin{cases} 1 & , \quad 0 \le |f| < (1-\alpha)/2T \\ \cos[(T/4\alpha)(2\pi|f| - \pi(1-\alpha)/T] & , \quad (1-\alpha)/2T \le |f| < (1+\alpha)/2T \\ 0 & , \quad (1+\alpha)/2T \le |f| \end{cases}$$

$\alpha = 0.2 \qquad T = 1/2400$

4 FSK Deviation		
Di-bit	Symbol	Deviation
01_2	+3	+1 050Hz
00_2	+1	+350Hz
10_2	-1	-350Hz
11_2	-3	-1050Hz

*Rx Baseband Filter



$$|H(f)| = \begin{cases} 1 & , \quad 0 \le |f| < (1-\alpha)/2T \\ \cos[(T/4\alpha)(2\pi|f| - \pi(1-\alpha)/T] & , \quad (1-\alpha)/2T \le |f| < (1+\alpha)/2T \\ 0 & , \quad (1+\alpha)/2T \le |f| \end{cases}$$

$$|D(f)| = \frac{\pi f T}{\sin(\pi f T)}$$

$\alpha = 0.2 \qquad T = 1/2400$

Figure 9.1

9.2.2.3 4FSK Modulator

The 4FSK modulator consists of a Square Root Raised Cosine Filter, cascaded with a frequency modulator as illustrated in figure 9.2.

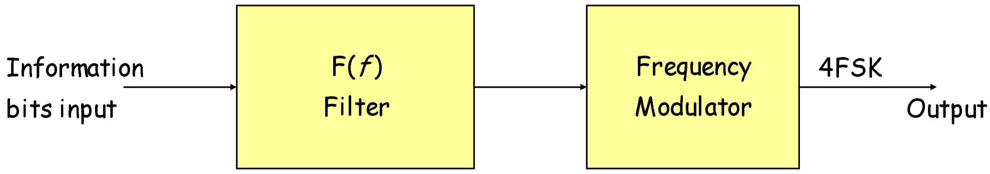


Figure 9.2: 4FSK Modulator

9.3 Channel access transmitter ramp timing

dPMR entities are obliged to conform to transmitter ramp timing. In order that entities may use low cost PLLs, the transmitter ramp timing is very relaxed.

Where an MS has been solicited to transmit a response, the preamble at the start of the transmission is timed to conform with figure 9.3. Figure 9.3 illustrates the case where MS(A) (or BS) has transmitted a message that solicits a response from MS(B). The MS transmitting the response sends its first bit of preamble 30 ms from the last bit of the message that solicited the response. The diagram does not imply any limitation on the start of the MS Tx RF power ramp which does not need to have attained full power for the first 24 bits of the preamble.

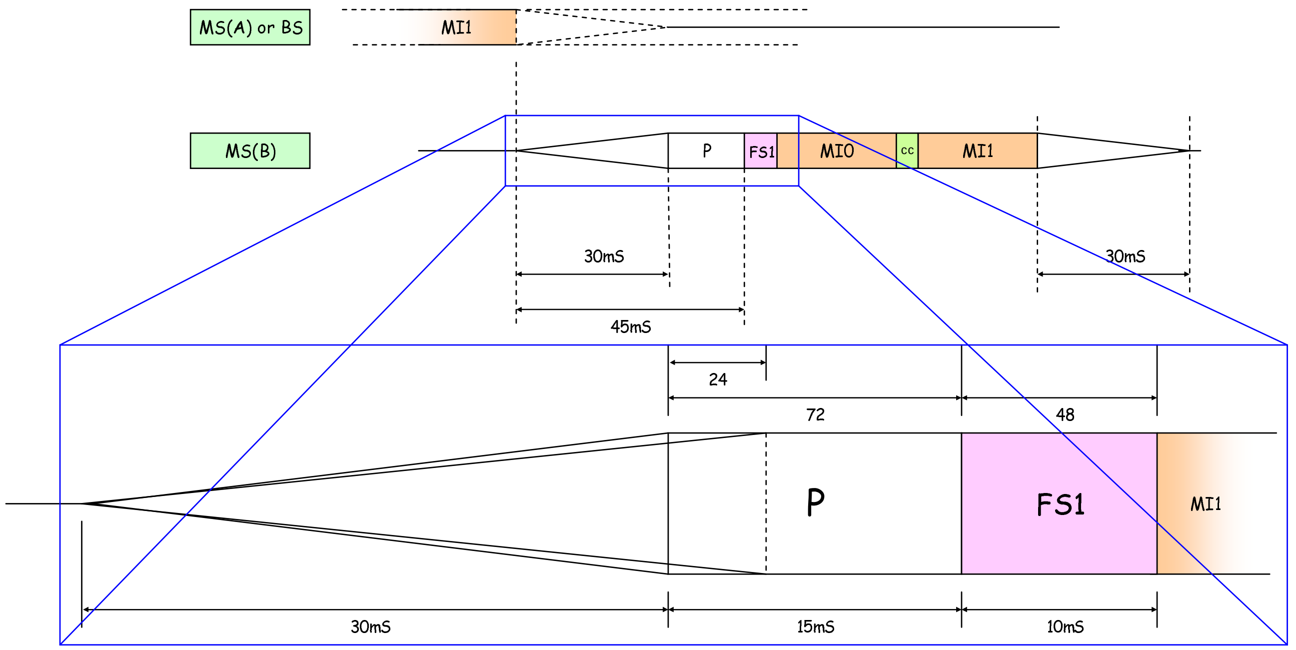


Figure 9.3: Preamble Timing

When the MS has transmitted its response, the MS ramps down the transmitter and in time to be able to decode a new message within 30 ms of the last transmitted message bit.

Annex A (informative): Bibliography

ISO/IEC 646 (1991): "Information technology - ISO 7-bit coded character set for information interchange".

ISO/IEC 8859 series (1998 - 2001): "Information technology - 8-bit single-byte coded graphic character sets".

MPT 1327 (June 1997): "A Signalling Standard for Trunked Private Land Mobile Radio Systems".

MPT1318: "Engineering Memorandum, Trunked Systems in the Land Mobile Service". February 1986, United Kingdom Department of Trade and Industry.

History

Document history		
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