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TECHNICAL REPORT

Digital Enhanced Cordless Telecommunications (DECT);
DECT-2020 New Radio (NR) interface;
Study on MAC and higher layers

Reference

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Keywords

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ETSI

650 Route des Lucioles
F-06921 Sophia Antipolis Cedex - FRANCE

Tel.: +33 4 92 94 42 00 Fax: +33 4 93 65 47 16

Siret N° 348 623 562 00017 - NAF 742 C
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Foreword

This Technical Report (TR) has been produced by ETSI Technical Committee Digital Enhanced Cordless Telecommunications (DECT).

The present document presents a study of a new radio interface named DECT-2020. DECT-2020 is a state of the art radio interface based on OFDM with options for MIMO and is intended as long-term evolution of DECT technology.

The present document is focused on the MAC and higher layers.

The technical content in the present document has been compiled from numerous contributions by members of TC DECT and ad-hoc working groups. The structure of the document sometimes reflects this ad-hoc nature.

Modal verbs terminology

In the present document "**should**", "**should not**", "**may**", "**need not**", "**will**", "**will not**", "**can**" and "**cannot**" are to be interpreted as described in clause 3.2 of the ETSI Drafting Rules (Verbal forms for the expression of provisions).

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Introduction

The current DECT radio interface was designed in the early 1990's and is based on TDMA/TDD with Gaussian Frequency Shift Keying (GFSK) modulation. Although this interface is able to provide a cost-effective solution for cordless telephony applications with an appropriate reuse of the spectrum, it cannot provide the high data rates and bandwidth efficiency required by most modern evolution scenarios. In addition, promising applications such as Audio-Streaming and Wireless Industrial Automation in Internet of Things (IoT) domain introduces Ultra Reliability and Low Latency requirements that have to be taken into account in any technology evolution.

IMT-2000 is the term used by the International Telecommunications Union (ITU) for a set of globally harmonised standards for third generation (3G) mobile telecoms services and equipment. 3G services are designed to offer broadband cellular access at speeds of 2 Mbps, which will allow mobile multimedia services to become possible.

DECT is, and will continue to be, one of the IMT-2000 technologies. However, the ITU work continued, first with IMT-Advanced, and it is now going further with IMT-2020. The term IMT-2020 was coined in 2012 by the ITU and means International Mobile Telecommunication system with a target date set for 2020, with the intention of addressing fifth generation (5G) mobile telecoms services and equipment.

The ETSI DECT Technical Committee and the industry body DECT Forum are currently supporting activities to develop DECT to meet the IMT-2020 requirements. This will require major changes to the existing DECT standards, and specifically to the MAC and PHL layers.

The present document contains the outcome of a series of initial technical studies focused on the MAC and higher layers of DECT-2020: New Radio Interface (NR). DECT-2020 NR is a state of the art radio interface based on OFDM and supporting MIMO and is able to offer the required data rates, spectrum efficiency and other characteristics to become an IMT-2020 radio interface as defined by ITU-R.

The PHY layer study of DECT-2020 is described in ETSI TR 103 514 [i.26].

The present document does not attempt to close the topic and subsequent, more detailed studies, on the different layers are expected in further project stages.

The material described in the present document contains the outcome of STF 564, an ETSI task force created to perform the initial studies on the field, along with other contributions from the DECT industry.

1 Scope

The present document contains the outcome of a series of initial technical studies focused on the MAC and higher layers of DECT-2020: New Radio Interface (NR). DECT-2020 NR is a state of the art radio interface based on OFDM and supporting MIMO and is able to offer the required data rates, spectrum efficiency and other characteristics to become an IMT-2020 radio interface as defined by ITU-R.

The PHY layer study of DECT-2020 is described in ETSI TR 103 514 [i.26].

The present document does not attempt to close the topic and subsequently, more detailed studies, on the different layers are expected in further project stages.

The material described in the present document contains the outcome of STF 564, an ETSI task force created to perform the initial studies on the field, along with other contributions from the DECT industry.

For the purpose of the present document the terms "DECT-2020", "DECT-2020 New Radio" or "DECT-2020 NR" all have the same meaning, and all of them refer to DECT utilizing the new radio interface based on OFDM as described in ETSI TR 103 514 [i.26] (PHY layer) and in the present document (MAC and higher layers). This new radio interface is targeted to meet the IMT-2020 requirements.

The terms FP-2020 or PP-2020 refer to FP and PP (respectively) devices supporting DECT-2020.

The present document is motivated by recent efforts to identify new ways of utilizing efficiently DECT frequency bands and potentially additional bands. New modes of operation are defined to target a more diverse set of use cases, while addressing 5G requirements for low latency, high spectral efficiency and large numbers of client nodes.

2 References

2.1 Normative references

Normative references are not applicable in the present document.

2.2 Informative 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 referenced document (including any amendments) applies.

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

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 EN 300 175-1: "Digital Enhanced Cordless Telecommunications (DECT); Common Interface (CI); Part 1: Overview".

[i.2] ETSI EN 300 175-2: "Digital Enhanced Cordless Telecommunications (DECT); Common Interface (CI); Part 2: Physical Layer (PHL)".

[i.3] ETSI EN 300 175-3: "Digital Enhanced Cordless Telecommunications (DECT); Common Interface (CI); Part 3: Medium Access Control (MAC) layer".

[i.4] ETSI EN 300 175-4: "Digital Enhanced Cordless Telecommunications (DECT); Common Interface (CI); Part 4: Data Link Control (DLC) layer".

[i.5] ETSI EN 300 175-5: "Digital Enhanced Cordless Telecommunications (DECT); Common Interface (CI); Part 5: Network (NWK) layer".

[i.6] ETSI EN 300 175-6: "Digital Enhanced Cordless Telecommunications (DECT); Common Interface (CI); Part 6: Identities and addressing".

[i.7] ETSI EN 300 175-7: "Digital Enhanced Cordless Telecommunications (DECT); Common Interface (CI); Part 7: Security features".

[i.8] ETSI EN 300 175-8: "Digital Enhanced Cordless Telecommunications (DECT); Common Interface (CI); Part 8: Speech and audio coding and transmission".

[i.9] ETSI TS 102 939-1: "Digital Enhanced Cordless Telecommunications (DECT); Ultra Low Energy (ULE); Machine to Machine Communications; Part 1: Home Automation Network (phase 1)".

[i.10] ETSI TS 102 939-2: "Digital Enhanced Cordless Telecommunications (DECT); Ultra Low Energy (ULE); Machine to Machine Communications; Part 2: Home Automation Network (phase 2)".

[i.11] ITU-R Recommendation M.2410-0: "Minimum requirements related to technical performance for IMT-2020 radio interface(s)".

[i.12] ETSI TR 103 515: "Digital Enhanced Cordless Telecommunications (DECT); Study on URLLC use cases of vertical industries for DECT evolution and DECT-2020".

[i.13] ITU-R Recommendation M.2412-0: "Guidelines for evaluation of radio interface technologies for IMT-2020".

[i.14] ITU Radiocommunication Study Groups; Working Party 5D; Attachment 7.4 to Document 5D/758; Liaison Statement to External Organizations; Further information related to draft new Report for IMT-2020 evaluation.

[i.15] Guidelines for evaluation of radio interface technologies for IMT-2020, ITU, Revision 2 to Document 5D/TEMP/347-E, 20 June 2017.

[i.16] IEEE Transactions on Communications: "Robust Frequency and Timing Synchronization for OFDM"; Timothy M. Schmidl and Donald C. Cox, Vol. 45, No. 12, December 1997, pp 1613-1621.

[i.17] ETSI TS 136 211 (V10.7.0): "LTE; Evolved Universal Terrestrial Radio Access (E-UTRA); Physical channels and modulation (3GPP TS 36.211 version 10.7.0 Release 10)".

[i.18] 3GPP TS 38.211 (V1.0.0) (2017-09): "NR; Physical channels and modulation".

[i.19] IEEE P802.11ah™/D10.0, Part 11: "Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specifications, Amendment 2: Sub 1 GHz License Exempt Operation", September 2016.

[i.20] IEEE 802.11ac™-2013, Part 11: "Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specifications, Amendment 4: Enhancements for Very High Throughput for Operation in Bands below 6 GHz".

[i.21] IEEE P802.11ax™/D1.4, Part 11: "Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specifications, Amendment 6: Enhancements for High Efficiency WLAN", August 2017.

[i.22] IEEE 802.11-03™/940r4: "TGn Channel Models", May 2004.

[i.23] ETSI TS 136 212 (V10.9.0): "LTE; Evolved Universal Terrestrial Radio Access (E-UTRA); Multiplexing and channel coding (3GPP TS 36.212 version 10.9.0 Release 10)".

[i.24] 3GPP TS 38.212 (V1.0.0) (2017-09): "NR; Multiplexing and channel coding".

[i.25] IEEE 802.15-04-0585-00-004b: "Multipath Simulation Models for Sub-GHz PHY Evaluation", October 2004.

[i.26] ETSI TR 103 514: "Digital Enhanced Cordless Telecommunications (DECT); DECT-2020 New Radio (NR) interface; Study on Physical (PHY) layer".

[i.27] ETSI TR 103 669: "Digital Enhanced Cordless Telecommunications (DECT); DECT-2020 New Radio (NR) interface; Study of self evaluation towards IMT-2020 submission".

[i.28] ETSI TR 103 637: "Digital Enhanced Cordless Telecommunications (DECT); DECT-2020 New Radio (NR) interface; Study on Security Architecture".

[i.29] METIS Channel Models, Deliverable D1.4, Document Number: ICT-317669-METIS/D1.4.

[i.30] ETSI EN 301 908-10: "Electromagnetic compatibility and Radio spectrum Matters (ERM); Base Stations (BS), Repeaters and User Equipment (UE) for IMT-2000 Third-Generation cellular networks; Part 10: Harmonised Standard for IMT-2000, FDMA/TDMA (DECT) covering the essential requirements of article 3.2 of the Directive 2014/53/EU".

[i.31] ETSI TS 102 527-1: "Digital Enhanced Cordless Telecommunications (DECT); New Generation DECT; part 1: Wideband Speech".

[i.32] ETSI TS 102 527-3: "Digital Enhanced Cordless Telecommunications (DECT); New Generation DECT; part 3: Extended Wideband Speech Services".

[i.33] ETSI TS 102 527-5: "Digital Enhanced Cordless Telecommunications (DECT); New Generation DECT; Part 5: Additional feature set nr. 1 for extended wideband speech services".

[i.34] IEEE 802.11: "Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specifications".

3 Definition of terms, symbols and abbreviations

3.1 Terms

For the purposes of the present document, the terms given in ETSI EN 300 175-1 [i.1] and the following apply:

beacon bearer packet types: packet formats intended for use in beacon bearers and C/L downlink bearers

NOTE: They include synchronization fields and do not need to support MIMO.

burst: concatenation of an I or O packet immediately followed by one or several C packets or, alternatively, an L or S packet

burst train: concatenation of several bursts transmitted over the same carrier or carriers separated by blank spaces of duration no longer than a given value (N_{MAXO}) and usually introduced for listening for responses from the opposite peer or to allow the transmission of other traffics

"HE" packet types: packet formats intended for continuous data transmission over several frames

NOTE: They may support circuit-mode traffic, URLLC traffic as well as packet mode traffic, and may implement MIMO.

"Legacy" DECT: current DECT technology as defined by ETSI EN 300 175 parts 1 [i.1] to 8 [i.8]

packet-mode: asynchronous unscheduled data transmission

RAC packet types: packet types formats intended for use in Random Access Channels (RAC)

NOTE: They may be used for initially accessing a channel, carry only C-plane traffic, and do not need to support MIMO.

RAC traffic: asynchronous unscheduled data traffic consisting on signalling only

slotxcarrier: basic resource block consisting on a single carrier (1,728 MHz) over a full slot

"Standard" packet types: packets intended for IP data packet-mode transmissions

NOTE: They are self-detectable packets usable in either synchronous or asynchronous way and may implement MIMO. The design of these packets is closer to the designs used in other WLAN technologies.

ULE packet types: packet formats intended for use in ULE (Ultra Low Energy) packet data transmissions

NOTE: They may be used for initially accessing a channel, are able to carry both U-plane and C-plane traffic, and do not need to support MIMO.

ULE traffic: asynchronous unscheduled data traffic consisting on a small amount of data combined with signalling

Ultra-Low Energy (ULE): ultra-low power consumption packet data technology based on DECT intended for M2M communications and defined by ETSI TS 102 939 part 1 [i.9] and part 2 [i.10]

WLAN traffic: asynchronous unscheduled data traffic consisting on a on a variable and potentially huge amount of data potentially combined with some signalling

NOTE: "WLAN" refers to Wireless LAN in generic sense, and is not intended to imply a particular technology.

3.2 Symbols

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

N_{BPSC}	Number of Bits Per SubCarrier
N_{CBPS}	Number of Coded Bits Per Symbol
N_{CTF}	Number of channel training symbols
N_{DBPS}	Number of data bits per symbol
N_{DC}	Number of null subcarriers at or surrounding DC
N_{DFT}	Discrete Fourier transform size
N_{SD}	Number of data subcarriers per OFDM symbol
N_{MAXB}	Maximum length of a burst (measured in full slots)
N_{MAXO}	Maximum length of the separation between the end of a burst and an open continuation packet (measured in full slots)
$N_{SERVICE}$	Number of bits in the SERVICE subfield of the Data field
N_{SN}	Number of null subcarriers
N_{SP}	Number of pilot subcarriers per OFDM symbol
N_{SR}	Highest data subcarrier index per OFDM symbol
N_{SS}	Number of Spatial Streams
N_{ST}	Total number of used subcarriers per OFDM symbol,
N_{SYM}	Number of data SYMbols
N_{TAIL}	Number of TAIL bits for BCC encoder
RX	Receiver
T_{CTF}	Channel Training Field Time
T_{DFT}	DFT period
T_{FRAME}	Frame Time
T_{GT}	Guard field Time
T_{HF}	Header Field Time
T_{HFS}	Short Header Field Time
T_{SLOT}	Slot Time
T_{STF}	Synchronization Training Field Time
T_{STFS}	Short Synchronization Training Field Time
T_{SYM}	Symbol Time
TX	Transmitter
W_{BC}	Basic Channel Bandwidth/Spacing

3.3 Abbreviations

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

ACK	ACKnowledgement
AES	Advanced Encryption Standard

AGC	Automatic Gain Control
AP	Access Point
ARI	Access Rights Identity
ARQ	Automatic Repeat-reQuest
ARQ	Automatic Retransmission Query
AWGN	Additive White Gaussian Noise
BBC	Beacon Bearer Controller
BCC	Binary Convolutional Codes
BMC	Broadcast Message Control
BPSK	Binary Phase Shift Keying
BS	Base Station (a.k.a FP, AP)
BSSID	Basic Service Set Identifier
BTS	Base Transceiver Station
BW	BandWidth
C	immediate Continuation full slot packet
CBC	Connection-less Bearer Control
CCF	Cluster Control Function
CCM	Counter with CBC-MAC
C_F	higher layer signalling Channel (fast)
CFO	Carrier Frequency Offset
CH	Immediate continuation Half slot packet
CISS	Call Independent Supplementary Services
CL	ConnectionLess
CL_F	higher layer ConnectionLess channel (fast)
CLMS	ConnectionLess Message Service
CMC	Connection-less Message Controller
COMS	Connection Oriented Message Service
CP	Cyclic Prefix
CRC	Cyclic Redundancy Check
C_S	higher layer signalling Channel (slow)
CSF	Cell Site Function
CTF	Channel Training Field
CTFM	Channel Training Field for MIMO
D	Downlink
DA	Destination Address
DBC	Dummy Bearer Control
DC	Direct Current
DECT	Digital Enhanced Cordless Telecommunications
DECT-2020	Physical Layer for DECT-2020
DF	Data Field
DFT	Discrete Fourier Transform
DL	DownLink
DLC	Data Link Control
DoS	Denial of Service
ED	Error-Detecting
eMBB	enhanced Mobile BroadBand
FDMA	Frequency Division Multiple Access
FEC	Forward Error Correction
FFS	For Further Study
FP	Fixed Part (a.k.a BS, AP)
FP-2020	PP implementing DECT-2020
FS	Full Slot
FT	Fixed radio Termination
GF	inter-slot Guard Field
GFSK	Gaussian Frequency Shift Keying
HARQ	Hybrid Automatic Repeat-reQuest
HE	High Efficiency
HE-FS	High Efficiency-Full Slot
HE-HS	High Efficiency-Half Slot
HF	Header Field
HFS	Header Field Symbol
I	Initial packet full slot

IE	Information Element
IH	Initial Half slot packet
IMEI	International Mobile Equipment Identity
IMSI	International Mobile Subscriber Identity
IMT	International Mobile Telecomunications
IP	Internet Protocol
I_{PF}	higher layer Information channel (protected) transported multiplexed with signalling in the E+U type slots
IPUI	International Portable User Identity
IRC	Idle Receiver Control
ITU-R	International Telecommunication Union, Radiocommunication sector
JIT	Just-in-time
LA	Location Area
LAN	Local Area Network
LAPC	DLC layer C-plane protocol entity
LDPC	Low Density Parity Check (code)
LLME	Lower Layer Management Entity
LMS	Last Minute Scan
LP	Long Preamble
LSB	Least Significant Bit
MAC	Medium Access Control
MBC	Multi-Bearer Control
MCC	Mobile Country Code (3GPP)
MCS	Modulation and Coding Scheme
MIB	Master Information Block
MIC	Message Integrity Check
MIMO	Multiple Input/Multiple Output
MM	Mobility Management
mMTC	massive Machine Type Communications
MNC	Mobile Network Code (3GPP)
MSB	Most Significant Bit
MU	Multi-User
N/A	Not Applicable
NACK	Negative ACKnowledgement
NR	New Radio

NOTE: Refers to DECT-2020 radio interface as described in the present document.

NWK	NetWorK
O	Open continuation full slot packet
OFDM	Orthogonal Frequency-Division Multiplexing
OH	Open continuation Half slot packet
OSI	Open Systems Interconnection
OTAP	Over-the-air programming
PARK	Portable Access Rights Key
PBR	Prioritized Bit Rate
PBX	Private Branch eXchange
PD	Protocol Discriminator
PDCCH	Physical Downlink Control Channel
PDCP	Packet Data Convergence Protocol
PDF	Probability Density Function
PDU	Protocol Data Unit
PEI	International Portable Equipment Identity
PER	Packet Error Rate
PHL	PHysical Layer
PHY	PHYsical
PLMN	Public Land Mobile Network (3GPP)
PMSE	Programme-Making and Special Events
PP	Portable Part (equivalent to the UE in 3GPP terminology)
PP-2020	PP implementing DECT-2020
PRB	Physical Resource Block
PSS	Primary Synchronization Signal

PT	Portable radio Termination
QAM	Quadrature Amplitude Modulation
QoS	Quality of Service
QPSK	Quadrature Phase Shift Keying
Q_T	system information and Multiframe marker
R	code Rate
RA	Receiving STA Address
RABC	Random-Access Bearer Control
RAC	Random Access Channel
RAN	Radio Access Network
RF	Radio Frequency
RFP	Radio Fixed Part
RFPI	Radio Fixed Part Identity
RL	Routing Layer
RMS	Root Mean Square
RNTI	Radio Network Temporary Identifier
RPF	Reference Pilot Field
RPN	Radio fixed Part Number
RRC	Radio Resource Control
RSSI	Radio Signal Strength Indicator
RTT	Round trip time
SA	Source Address
SAP	Service Access Point
SAPI	Service Access Point Identifier
SAW	Stop-And-Wait
SCS	Sub Carrier Spacing
SDU	Service Data Unit
SIB	system information broadcast
SISO	Single Input/Single Output
SNIR	Signal-to-Noise-plus-Interference Ratio
SSID	Service Set Identifier
STA	STAtion (IEEE 802.11)
STF	Special Task Force
STF	Synchronization Training Field
STFS	Synchronization Training Field (Short)
SU	Single User
TA	Transmitting STA Address
TBC	Traffic Bearer Control
TCP	Cyclic Prefix Time
TCP	Transmission Control Protocol
TDD	Time Division Duplex
TDMA	Time Division Multiple Access
TI	Transaction Identifier
TMSI	Temporary Mobile Subscriber Identity
TPUI	Temporary Portable User Identity
TS	Technical Specification
U	Uplink
UDP	User Datagram Protocol
UE	User Equipment (3GPP terminology for the PP)
UL	UpLink
ULE	Ultra-Low Energy
UPCS	Unlicensed Personal Communications Services (band)
URLLC	Ultra-Reliable and Low Latency Communications
WLAN	Wireless LAN

4 Introduction to DECT-2020 Use Cases and their Requirements

4.1 Introduction

A separate study on DECT evolution and DECT-2020 use cases and requirements has been conducted and published as ETSI TR 103 515 [i.12]. According to ETSI TR 103 515 [i.12], the following three major application areas have been identified as target for DECT-2020 radio technologies. These are:

- Home and Building Automation, including Smart Living.
- Industry automation - Factories of the Future, Industry 4.0.
- Media and entertainment industry - Programme Making and Special Events (PMSE).

Nevertheless, DECT-2020 application areas will not be restricted to these three major domains and additional applications and use cases may be supported.

In particular, massive Machine Type Communication (mMTC) and Ultra Reliable Low Latency Communications (URLLC) have been selected as initial primary design targets of DECT-2020 and are the basis for the initial contribution to IMT-2020 and associated self-evaluation [i.13], [i.14], [i.15]. The IMT-2020 compliance part of the project is published as ETSI TR 103 669 [i.27].

In addition to that, the support of enhanced Mobile Broadband (eMBB) is seen as a natural evolution in the project and several features for future support of eMBB are considered in the MAC and higher layers study.

4.2 Other Design Targets for DECT-2020

In addition to the use cases related to URLLC identified by ETSI TR 103 515 [i.12], the support of efficient transmission of IP data and the support of voice communications are also considered basic requirements.

Regarding bandwidth efficiency, the technology should be efficient as any other state of the art (5G) radio technology.

Regarding radio propagation characteristics, the new technology should provide an advantage over existing DECT that may be used to either, extend the cell range or decrease the power.

Regarding transmission power, the working assumption is that the maximum transmission power over the existing DECT band will be the same as DECT (250 mW). In case of using space multiplexing, this power will be split between the different antennas.

Based on existing IMT-2000 specifications, similar maximum transmission power can be assumed for bands adjacent to current DECT, such as the IMT-2000 FT allocation (see ETSI EN 301 908-10 [i.30]).The maximum transmission power for other bands that could be allocated to the service in the future is an open topic and cannot be anticipated at time of the present document.

4.3 IMT-2020 scenarios and performance requirements

IMT-2020 defines 3 usage scenarios:

- Enhanced Mobile Broadband (eMBB).
- Massive Machine-Type Communications (mMTC).
- Ultra-Reliable Low Latency Communications (URLLC).

IMT-2020 defines 13 technical performance requirements for these usage scenarios (see [i.11]):

- Peak data rate.
- Peak spectral efficiency.

- User experienced data rate.
- 5th percentile user spectral efficiency.
- Average spectral efficiency.
- Area traffic capacity.
- Latency:
    - User plane latency.
    - Control plane latency.
- Connection density.
- Energy efficiency.
- Reliability.
- Mobility.
- Mobility interruption time.
- Bandwidth.

In addition to the IMT-2020 requirements, the following general requirements are also design goals:

- Improved range compared to legacy DECT.
- Improved voice quality compared to legacy DECT.
- Improved data rates compared to legacy DECT.
- Improved number of simultaneous connections compared to legacy DECT.
- DECT-2020 should be able to coexist in the same area with legacy DECT systems operating over the same spectrum and should implement the proper channel selection rules to mitigate any interference to/from legacy DECT systems.
- It should be possible the implementation of compatible devices, either FP or PP, implementing both, DECT-2020 and legacy DECT, radio interfaces.

DECT-2020 devices need to coexist with legacy devices. Specifically, the operation of a DECT-2020 device should not interfere with, or significantly degrade the performance of nearby legacy DECT systems. Likewise, the design of DECT-2020 should ensure that legacy DECT systems will not interfere with nearby DECT-2020 systems, or reduce its performance beyond the unavoidable limitation by the available spectrum.

5 Methodology, initial sources, simulation tools, models and material from the PHY layer

5.1 Initial sources

Different OFDM 5G or pre-5G technologies have been studied and have had an influence in the design of DECT-2020. In particular, the following technologies should be noted:

- IEEE 802.11ah [i.19].
- LTE (4G) [i.17] and [i.23].
- LTE NR (new radio) [i.18] and [i.24].

Other technologies leveraged in DECT-2020 development have been the following:

- IEEE 802.11ac [i.20].
- IEEE 802.11ax [i.21].

Nevertheless, DECT-2020 is an original technology with its own design choices, OFDM parameters, overall concepts and PHY layer architecture.

5.2 Simulation tools

A simulation environment combining MATLAB® and C++ code has been developed and has been used for assessing important performance metrics concerning detection, synchronization, channel estimation and forward error correction. This is done under various types of channel impairments.

5.3 Channel models

The radio channel has been modelled primarily by Additive White Gaussian Noise (AWGN) model and Exponential Power Profile model, for both SISO and MIMO configurations. These models are very commonly used in the literature.

The current simulation environment contains an implementation of IEEE 802.11-03 [i.22]. These models include support for channel variation over time caused by motion and fluorescent lighting, and will be used in the future for simulations of complex in-door scenarios.

Suitable out-door models are still being studied.

5.4 Channel measurements

No specific channel measurements have been done by this study. However, information from external sources has been used where applicable. The following sources have been used:

- IEEE 802.11-03 [i.22] "TGn" channel models.
- IEEE 802.15-04 [i.25] "Multipath Simulation Models for Sub-GHz PHY Evaluation".
- Guidelines for evaluation of radio interface technologies for IMT-2020, ITU, Revision 2 [i.15].

5.5 Review of material from the PHY layer

5.5.1 "Standard" frames (long and short variants)

These were the first frames proposed and were used in all initial simulation studies. They were also the basis for the numbers in the initial IMT-2020 submission template (see ETSI TR 103 669 [i.27]).

Standard frames are self-contained frames that can operate with and without a beacon. These frames have a robust synchronization preamble (making possible asynchronous synchronization without prior observation of a beacon); duplicated training symbol allowing S&C synchronization; two symbols for HF (they do not need necessarily to be 1+1 redundancy). They allow MIMO operation and training.

One symbol is used as inter-slot guard space (not optimal, possibly too much). But no guard space is needed when they are used to transmit continuously over consecutive slots (i.e. multi-slot transmission).

Comment: They are efficient only if long bursts of multi-slot on the same carrier can be found. They are inefficient if used as single slots. The short format allows some improvement but only relatively.

These frames are ok if a WLAN operation over clean spectrum is planned. Not optimal design if they have to coexist with several slot-like bearers (i.e. HF slots) on the same carriers.

Synchronization is ok for WLAN designs.

5.5.2 Beacon frames

A special formal for downlink bearers. Intended for single bearers with no MIMO.

5.5.3 RAC and ULE frames

A special formal for uplink short transmissions such as RAC and ULE. Design is (for the while) identical to beacons, but can be further optimized.

5.5.4 HE frames

Optimized design for slot/resource block oriented design. Suitable for operation over single slots (even half slots are proposed). Suitable for circuit mode channels and for scheduled transmission.

In practice, they require a more complex MAC design. They are not suitable in simple WLAN operation. In practice a scheduled architecture similar to LTE or NR is intended.

5.5.5 Items requiring further study

At the end of the Physical Layer study (published as ETSI TR 103 514 [i.26]), the following PHY layer items were identified as requiring further study:

- The feasibility of half slots was unclear due to the tight inter-slot guar space and reduced CP.
- It was also challenged the feasibility of the most efficient full-slot format (named HE1) over single carriers. Some experts pointed that at least double carriers may be required.

6 Channel Access Concepts for DECT-2020

6.1 Review of previous design choices

A primary assumption from earlier work, and a review of other technologies (e.g. WiFi, LTE, etc.), is that the PHL should be based on OFDM. However, there are a number of other assumptions and design choices to be considered:

- Basic radio technology is OFDM and channel access is TDMA/FDMA based.
- DECT basic frame time of 10 ms:
  - This is the same as legacy DECT.
  - Basic frame split into 24 time-slots (i.e. same number of slots as legacy DECT).
  - Time-slots can be aggregated (e.g. double slots, quadruple slots, etc.).
  - Half-slots for some packet types are also supported.
- DECT basic channel width of 1,728 MHz:
  - This is the same as legacy DECT.
  - Multiple contiguous channels can be aggregated (i.e. bonded).
- Data rates require higher order modulation (up to 1024-QAM).
- Improved reliability requires protection by FEC and CRCs, with ARQ mechanism (i.e. HARQ).
- Use of MIMO for better data rates, increased reliability and efficiency.
- Security will be state of the art security (to be covered by a separate study).

6.2 Possible approaches for the channel access

6.2.1 General

The following approaches were considered at the beginning of the MAC study (described in the present document).

6.2.2 Concept 1: improved WLAN approach

In this concept, a channel access based on modified CSMA/CA will be used (i.e. 802.11 principles).

A concept such as Restricted Access Window may be used to improve (and get differentiation) over mainstream 802.11 technologies. A possible model would be IEEE 802.11ah [i.19].

Standard frames long and short format may be used for this operation.

This paradigm is seen as suitable for efficient transmission of general IP packet traffic over "green" spectrum.

The PHY layer work (see ETSI TR 103 514 [i.26]) is able to support this MAC paradigm.

The design complexity of the approach is seen as the simpler than concept 2 and 3 (however it is not trivial).

However, this paradigm has the following drawbacks:

- It cannot provide proper support of URLLC scenarios, specifically scenarios combining Ultra Reliability with Low latency and high data rates. These scenarios were identified in ETSI TR 103 515 [i.12].
- It restricts the options for differentiation with IEEE 802.11 [i.34] solutions (however, some differentiation is still possible).
- It cannot provide optimal compatibility with legacy DECT and DECT evolution (compatible but not optimal design).
- It is unclear if IMT-2020 requirements can be meet. Specifically URLLC requirements.

6.2.3 Concept 2: scheduled operation design with complete MAC design

The design will use a complex MAC design with downlink channels (providing synchronization), RAC, and traffic channels. The basic idea will be following LTE or NR solutions (but not necessarily the same).

This design is suitable for URLLC periodic traffic. It can provide good coverage of URLLC requirements. This includes both IMT-2020 requirements and the requirements identified in ETSI TR 103 515 [i.12].

The PHY layer study (ETSI TR 103 514 [i.26]) considered this case and provided the basic elements for this type of design (beacon, RAC and HE frames).

This paradigm is considered far more complex that concept 1.

6.2.4 Concept 3: the best of 1 + 2

The concept 3 is defined as the ideal solution. It should provide both operation modes and should combine the advantages of both concepts 1 and 2.

In short, it should combine WLAN frames (with MIMO) for asynchronous packet mode traffic (with quick access) combined with scheduled operation for other traffic.

Concept 3 is more complex than either concept 1 and concept 2.

6.2.5 Decision and working assumption

6.2.5.1 Working decision on overall channel access concept

After extensive discussion within TC DECT, it was decided that the working assumption will be CONCEPT 3, which is the most complex and challenging solution.

The following additional notes are taken from the TC discussion:

- All traffic types should be supported: WLAN, URLLC, ULE and circuit mode.
- It does not need to be as LTE/NR.
- Due to the inherent complexity the project can be structured in phases or releases:
  - Phase 1, scheduled for 2020, should be simpler to meet the schedule.
- Resources and resource allocation will be FP controlled (or more FP controlled) than legacy DECT.
- The problem of uncoordinated systems need to be taken into account.
- There should not be the need for any radio network planning (i.e. as it happens in DECT today).
- The PHY layer work documented in ETSI TR 103 514 [i.26] and its repertory of PHY layer frames will be used as base material.

6.2.6 Expected spectrum bands

The technology will assume DECT spectrum and additional new spectrum are available. The design will be usable in legacy DECT spectrum, however features related to very high performance may require new "green" spectrum.

In particular the following additional spectrum allocations are seen as candidates and MAC design should take them into account:

- The frequency bands currently allocated to DECT service (1 880 MHz - 1 900 MHz).
- The frequency bands currently allocated to IMT-2000 FT (see ETSI EN 301 908-10 [i.30]) service (1 900 MHz - 1 980 MHz and 2 010 MHz - 2 025 MHz).
- New license exempt frequencies at the 5 GHz band.
- New local area licensed frequencies at the 3,7 GHz - 3,8 GHz band.

The following additional frequency allocations are identified as possible in the future, however full MAC support may be left to further releases:

- Bands above 24,25 GHz.

NOTE: Operation over higher frequency bands (over 24,25 GHz) is assumed to require subcarrier spacing scaling.

6.2.7 Specifications for channel access and service coexistence

The following principles are proposed:

- There is a beacon providing synchronization and time references.
- There may exist both scheduled services and non-scheduled services in the same system.
- There will be RAC (Random Access Channels) used for signalling operations.
- The RAC should be used for requesting scheduled access services.
- There will be ULE data channels intended for low traffic operations and operating in un-scheduled mode.

- There will be WLAN channels operating in un-scheduled mode and able to support high data rates. They should be able to support MIMO.
- There is a resource allocation process done by FP for any scheduled service.
- Resources used by scheduled services may be designated as protected and this will prevent collisions from un-scheduled access in the same system.
- Scheduled services will use High Efficiency packet formats (HE) and may support MIMO.
- FP will broadcast using the beacon bearer information for assisting PPs in channel selection and TDD coordination processes.
- FP will broadcast using the beacon bearer information for informing PPs of FP blind slots and for protecting resources used by some scheduled services, such as URLLC services.
- Algorithms for prevention of access collisions, containing random components, will be used to reduce the probability of access collisions in any random access service.
- Algorithms for resolution of access collisions, containing random components and exponential back-off times will be used to resolve access collisions in any random access service, when they happens.

6.3 Void

6.4 Void

6.5 Contribution 1: Channel access for DECT-2020

6.5.1 Background

The optimal type of channel access mechanism used for DECT-2020 depends very much on the spectrum regulations in which it is applied:

- DECT "core band" (1 880 MHz - 1 900 MHz):
  - The primary concern here is that the new channel access rules have to be compatible with the legacy devices, otherwise coexistence in this band may be compromised.
  - It might be acceptable to do things differently, but the onus should be on the new DECT-2020 devices to work with the legacy devices.
  - In time, as legacy use declines, it might be possible to relax the rules in favour of the DECT-2020 devices.
- (Proposed) DECT "extension band" (1 900 MHz - 1 920 MHz):
  - There are no legacy devices currently operating in this band.
  - If this band is shared with other industries (e.g. drones, rail, etc.) then this will influence the adopted channel access rules for operation in this band.
  - It would also be "desirable" to move "regular" DECT devices into this band (such devices are using the legacy radio, but since it is new band they could be different/relaxed channel access rules).
- Other IMT bands:
  - Rules applicable to the particular band may apply.

As discussed elsewhere, there are 3 main approaches to channel access for DECT-2020:

- Scheduled access:
    - Similar to legacy DECT, LTE, etc.
- Unscheduled access:
    - More like IEEE 802.11 series [i.34]
- Hybrid system:
    - Supports both depending on system requirements or function

6.5.2 DECT "core band" (1 880 MHz - 1 900 MHz)

The primary assumption is that DECT-2020 is being developed to operate primarily in the DECT core band (1 880 MHz - 1 900 MHz).

There is no extension band yet; and operating in other allowed IMT bands is not necessarily right for "DECT" since one of the key strengths and motivations for using DECT is the use of its own unlicensed, technology-dedicated, spectrum, rather than using public network operators, etc.

The result of using the DECT "core band" is that DECT-2020 has to coexist with legacy DECT systems, and much of the early discussion in the design of DECT-2020 was centred around the question of the slot/channel raster, which is aligned with legacy DECT in order to facilitate interop/coexistence with legacy DECT (i.e. 24 slots and 10 channels). Of course, in bands other than the DECT "core band" it would be possible to use a different slot/channel raster. However, this would also add some complexity to the design of the MAC.

So, the working assumption is the adoption of the legacy DECT channel raster (24 slots and 10 channels), but with some flexibility retained in the design (e.g. trying to avoiding any hard restrictions such as fixed bit-maps relating to slots/channels in MAC messages).

Operation on other bands, is left for further study at this stage, primarily due to time/effort restrictions needed to study them fully.

6.5.3 MAC Channel Access Background

As already mentioned our working assumption is that DECT-2020 system is TDMA based, using a slot/channel raster of legacy DECT (but with flexibility retained for other options in future).

A non-TDMA access mechanism like 802.11 does not fit well with the goal of coexisting with legacy DECT.

The other classification to consider is scheduled vs unscheduled.

A technology like 802.11 is unscheduled, and devices wanting to transmit use CSMA/CA techniques for each transmission. Features such as Restricted Access Windows and Target Wake-Time can be used to limit the number of devices that are able to access the base, which reduces the chance of collisions, and this could be regarded as a form of semi-scheduled access (i.e. a group of devices is allowed/scheduled to access at a particular time window) but ultimately CSMA/CA is still required.

On the other hand, in a scheduled system, there has to be a deterministic method to allow a particular device to communicate with the base at a specific time. There is a number of ways to do this: for example, particular slots/channels are allocated to devices, either on a permanent or semi-permanent basis; or more dynamically, as in the case of LTE, where the Physical Downlink Control Channel dynamically indicates the contents of the following Physical Downlink Shared Channel (and there is another similar mechanism for uplink access).

NOTE 1: An unscheduled system, can be regarded as "random access" or "contention-based access"; whereas a scheduled system can be regarded as "guaranteed access" or "non-contention based".

Even in scheduled systems, there will often be some cases where unscheduled (or random) access is required. This could be to "join" a system, to request to have resources allocated to it, or to perform some low-latency ad-hoc communication.

In an unscheduled system, the CSMA/CA is fundamental to the operation. This involves scanning the air prior to every transmission (often referred to as "listen before talk"), detecting collisions (usually done by not receiving an ACK for the transmitted message), and having some kind of random back-off/retry mechanism in the event of a failure.

In a scheduled system, there is an assumption that the availability of the allocated resources is guaranteed, i.e. there is no contention, and therefore CSMA/CA is not required. However, this assumption is not always valid. For example, in the DECT core band, there may be other users (i.e. unrelated neighbouring systems) which are not centrally coordinated (i.e. no frequency allocation/planning between neighbours) sharing the same resources.

To overcome this problem, legacy DECT uses two techniques: "background scanning", and "last minute scanning".

The "background scan" uses RSSI measurements taken over the entire slot/channel matrix in an attempt to identify free/busy resources. When selecting resources for transmission, free resources are used preferentially (or the quietest resources in the absence of perfectly free ones).

The background scanning will not solve the issue of the "hidden node problem", and so in addition to that DECT devices also perform a "last minute scan" (see CSMA/CA) prior to starting transmission on an allocated channel. This last minute scan is used to determine if the allocated resource really is free to use.

However, unlike unscheduled access's CSMA/CA, in legacy DECT, the last minute scan is only used when *initially* attempting to access the resources. It does not have to be repeated for every transmission (on those specific resources) thereafter.

There are a number of issues with the legacy DECT background scan and last minute scan:

- Background scanning is only suitable for detecting free/busy resources that are periodic and semi-static in nature, e.g. "circuit mode connections". It is not well-suited to detecting, short/random packet mode transfers since these are usually unpredictable.
- Last minute scanning is performed in the frame preceding the intended transmission time. This leads to increased latency. Also, the scan can only detect periodic transmissions, e.g. if the last minute scan detected another user on a particular slot/channel, the assumption is that the same user will be present on that slot/channel in the following frame too. This assumption is only valid for periodic transmissions, and specifically only when the period is 10 ms (or 5 ms or 2,5 ms, etc.)
- The PP is responsible for background scanning for normal traffic bearer establishment process. This is not suitable for low power applications. (For ULE operation, the FP was responsible for background scanning, and suggestion available channels to use; the PP was still responsible for last minute scan).

Looking towards DECT-2020, there are a few considerations:

- Resource allocation is much more complicated, and ultimately the FP is in the best position to allocate resources. Therefore, the FP needs to be performing the majority of background scanning.

NOTE 2: The FP can augment its own background scanning by using information provided by PP's, either voluntarily or by request of the FP.

- Reduced latency is important, and so removal or change of last minute scan should be considered.
- Whilst "circuit mode" applications will continue to exist, there will also be a move towards more "packet mode" operation, which is harder to detect by background scanning.

6.5.4 Notes on Last Minute Scanning

In legacy DECT, the last minute scan is in the frame prior to the intended frame of transmission. As already mentioned this adds to the latency of the system. It is also only useful for detecting periodic transmissions.

Other options:

- No last minute scan at all:
    - The assumption is that the selected channel is a good one (i.e. not in use by any other user), and so access is guaranteed. If this really is the case, then all is well, and the transmission occurs successfully, and there is no impact on another user.

- However, if there was another user using that channel, then a) the transmission might be unsuccessful, and b) the existing user will suffer from interference. The attempting device will have to try again (increased latency), and the existing device will be affected by loss of data (requiring retransmission or missing packets for streamed audio).

NOTE 1: Some existing cases for double-simplex bearer operation already use no last minute scan.

- "Just-in-time" (JIT) last minute scan:
  - Rather than scan the frame before, the scan is performed right at the beginning of the actual allocated time-slot itself.
  - The resulting transmission would necessarily be shorter in order to fit in the slot width (e.g. only possible to transmit a partial slot/half-slot, etc.) and so the payload capacity would be smaller.
  - Also, it could only detect a transmission in the part of the slot being scanned. This might detect transmissions from systems that were slot-synchronized, but it would likely miss transmissions from unsynchronized systems.

NOTE 2: Perhaps neighbouring systems can self-synchronize (at slot level) so that JIT scanning can be effective.

- Hybrid last minute scan:
  - Devices/applications that are latency/power sensitive may omit the last minute scan; they are instead relying only the quality of the channel selection to find a free/quiet channel.
  - Devices/applications that are not latency/power sensitive will use last minute scan.

6.5.5 Summary of Working Assumptions

The following working assumptions are used as input for the analysis:

- DECT "core band" as primary spectrum.
- Coexistence with legacy DECT devices is essential.
- General channel access methodology is TDMA based, using legacy DECT raster (24 slots/10 channels).
- Majority of background scanning is performed by FP.

6.5.6 Channel Access - Idea 1

An initial idea was to use very similar channel access methodology to legacy DECT/DECT ULE, with a few important changes:

- FP is performing background scanning to identify other users (interference) and find good (free/quiet) resources.
- FP advertises a selection of channels allocated for "random access" (i.e. a Random Access Channel or RAC).
- A PP wanting to establish a "traffic bearer" does so by sending a request message using one of the advertised RACs.
- The FP responds to the request on a pre-designated channel.
- In some cases, e.g. where only a single message exchange was required then the process ends there. However, sometimes further communication is required, and the initial message exchange is usually to request for resources to be allocated to it, and in that case the FP's response message will provide details (location/timing) of the allocated resources.

NOTE 1: The RAC/response process can be used to send small amounts of data, similar to ULE.

NOTE 2: When resources are allocated, these are similar to traffic bearers in legacy DECT. However, there is more flexibility in terms of periodicity, symmetry, etc.

Last minute scan strategy:

- The PP should perform LMS on the RAC channel to ensure it is free prior to use.
- The PP should perform LMS on any other allocated resources prior to use.
- There is an assumption that the FP is scanning more or less constantly and has selected good (free/quiet) channels for the RAC, designated response channel and allocated resources; therefore LMS (by the FP) on those channels is not required.

This initial exchange of messages can be seen as a "pilot bearer", where the PP requests a service, and the FP allocates and grants resources. The pilot bearer is released after the resources have been granted. On the next frame, the PP and FP can then switch to using the allocated resources.

The "access request" and "access grant" should use a low MCS for robustness and reliability.

An example of such an initial channel access procedure is shown in Figure 1, Figure 2 and Figure 3. The yellow "B" is the FP's beacon; the green "C" represents the positions for contention-based access; the red "1" is an existing user. Note, the green "C" positions are not actual transmissions, they are the potential positions that the PP's may use.

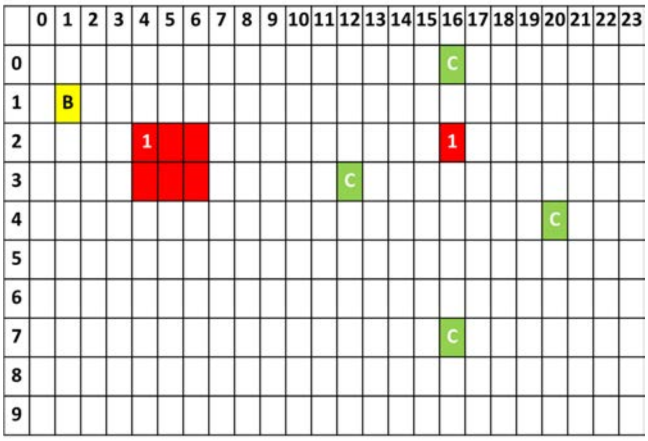


Figure 1: Frame N

On frame N+1, the PP transmits an "access request" on slot 16 channel 0, and the FP responds with an "access grant" two slots later on slot 18 channel 0.

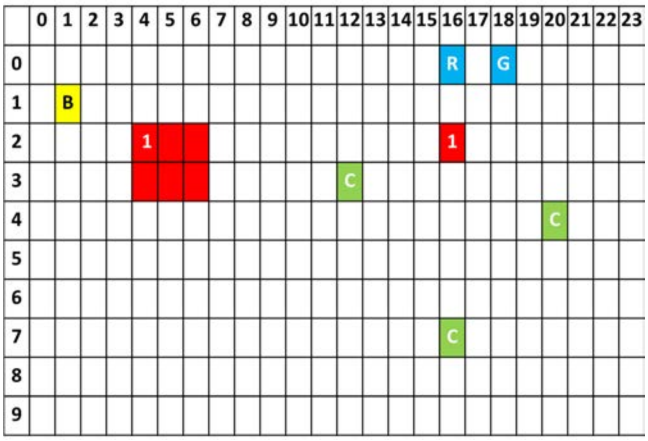


Figure 2: Frame N+1

On frame N+2, the FP allocated the requested resources, and the communication can now shift to slot 0-1 channel 6-7 and slot 12-13 channel 6-7. The FP also allocated a new "C" position on slot 18 channel 2.

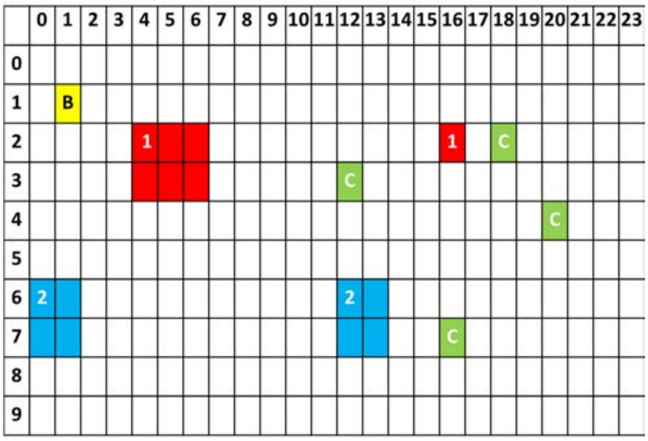


Figure 3: Frame N+2

In the above example, the "access grant" message was sent two slots after the "access request". This is the minimum possible time (since it will not be possible to transmit immediately after receiving a packet because of there is no time to process the message). However, it is not decided, exactly how the timing of the "access grant" message is determined; it could be deterministic, e.g. always 2 slots later, or 5 ms later, or it could be specified on the beacon, or maybe even the PP could specify the position when sending the "access request".

Open Issues:

- Latency due to LMS. As mentioned previously, LMS could be omitted for devices/applications that are sensitive to latency/power.
- Location/timing of the "pre-designated" response channel is TBD.

6.5.7 Channel Access - Idea 2

The concepts presented in "Idea 1" demonstrates the Random Access procedure, which can be used to request more permanent resources (e.g. "circuit mode" connections) or for very small data transfers (e.g. "ULE like" connections). However, for "packet mode" data services, there are other possibilities which can be used in addition.

For example, imagine an application requiring a connection to send/receive IP packets on a semi-regular basis, e.g. web browsing, messaging service, etc. The data content is generally too large to be using the "ULE like" connection mode (which is only really suited for single packets of data), but equally does not need the permanent high bandwidth or fixed latency of a "circuit mode" connection. Instead the system would like to be able to send/receive packets more or less when they arise, but not too frequently, and probably without any tight latency requirement.

NOTE 1: In legacy DECT DPRS "packet mode" operating is supported by creating a circuit mode connection and then using suspend/resume during inactive periods. However, this still requires establishing new bearers (new resources) each time the link is resumed.

There are shared resources like traffic bearers, but rather than being between the FP and a single PP, it is shared between different PPs. The sharing could be achieved by a number of methods. For example:

- Fully scheduled shared access (i.e. non contention based access) to different PPs based on frame/multi-frame, etc. For example PP1 has access to the resource on frame 6 of every multi-frame; PP2 has access to the resource on frame 8 and 12 of every multi-frame, etc.
- Contention-based shared access to different PPs (or groups of PPs). For example, PPs 1-10 may access the FP during frame 1 of every multi-frame. (Of course, this assumes that PPs are not accessing every frame of course, and the fewer PP's allocated to a particular "group" there less chance of collision.)
- A mixture of the two above methods.

In both cases, use of the shared resource is optional, i.e. a PP does not have to transmit on its allowed frame if it has no data to send - it is more like an "opportunity to use the resource". Similarly, in the downlink direction, the FP has an opportunity to transmit to a PP, but it does not have to if there is no data.

NOTE 2: For contention based access, if the FP has data to send to multiple PP's in the same contention window, them it will have to use some priority strategy as it can only send data to one PP at a time.

NOTE 3: The use of source/destination addresses in the packet is probably essential to ensure that they are delivered correctly.

NOTE 4: If the FP has no downlink data to send any PP, it may still decide to send some broadcast data (possibly indicating buffer status, broadcast ACK's, changes to contention window timing, handover to different slot/channel, etc.). This broadcast also has the effect of utilizing the resource, and so making it less likely to be used (stolen) by another system.

This type of shared resource, removes the need to continually resume/suspend connections (or start new connections), and seems to be well suited if latency or high data rate is not important. The contention based mode is similar to the operation of the RAC, but with a more managed approach (i.e. allowing windows for specific groups).

Table 1 shows typical message flow over several frames for such a shared traffic bearer, on slot 0/12 (i.e. typical 10 ms framing, of course, shorter framing could also be arranged). In this example, all access is non-contention based, i.e. the PPs have been allocated specific frames where they can access the FP.

Table 1: Example message flow for channel access messages

Frame	Slot	Action
0	0 (downlink)	FP broadcasts some status info
0	12 (uplink)	No PP has data to send. Empty
1	0 (downlink)	FP broadcasts some status info
1	12 (uplink)	PP 1 sends data to FP
2	0 (downlink)	FP broadcasts some status info, including ACK for PP1
2	12 (uplink)	PP 2 sends data to FP
3	0 (downlink)	FP sends data to PP3
3	12 (uplink)	PP3 sends data + ACK to FP
4	0 (downlink)	FP broadcasts some status info, including ACK for PP3, and (delayed) ACK for PP2
4	12 (uplink)	No PP has data to send. Empty
…	…	Etc.

In the above example, 3 PPs are accessing the FP and can send/receive data utilizing 1 physical resource.

Last minute scan strategy:

- Resources are allocated by the FP. FP can perform LMS prior to first transmission. The downlink side is utilized with a high % (maybe 100 %), and so LMS is not required by the FP again (until handover?).
- The up-link utilization depends on how many PPs are using this mode of channel access, but it is probably significantly less than 100 %. A PP may decide to perform LMS prior to utilizing the resource.

Open Issues:

- Since FP is allocating the resources, there may be issue of interference at one or more of the PPs (because of near/far problem), resulting in the inability of the PP to receive messages properly.
- ACKing of previous packets could be sent with the data, e.g. using send/receive sequence numbers to indicate if a previous packet was received, or using some common/broadcast mechanism. The former will result in some delay, since a packet is not ACK'd immediately. In practice, both methods could be used.

6.6 Contribution 2: Channel access for DECT-2020

6.6.1 Aspects to be considered

The following aspects needs to be considered when defining access scheme for DECT-2020:

- Traffic is not symmetric between uplink and downlink in modern wireless radio systems:
  - Several studies show that statistically traffic is 1:10 between uplink and downlink but there is significant variations in time, based on location and between networks.
- Even in simple mMTC sensor networks, which are limited by the use cases, there is occasionally need for significant amount of DL traffic:
  - Authentication, Device configuration, and OTAP procedures for example.
- Round trip time (RTT) of the system directly impacts the achievable application data rate over TCP/IP:
  - Well-designed networks the delay caused by radio interface can be the dominant factor.
- To support URLLC communications or professional audio, UL and DL switching points needs to be more frequent than every 5 ms.

Due to above, it is considered essential that future DECT-2020 systems are capable of:

- Dynamically adjust the number of DL and UL slots in a frame.
- Perform adjustments locally i.e. each FP can make own local decisions, including mesh devices that operating as FP mode.
- FPs can perform adjustments in fast manner, even in every 10ms frame. The adjustment rate should only depend on FPs signalling frequency. In order to avoid frequent signalling, when there is no need to adjust frame content, a single signalling message from the FP could indicate certain pattern which is in use for certain period.
- FP can create empty slots or even empty frames into the system. This can be achieved by defining that PPs are not allowed to make any implicit assumption on slot type (DL, UL), if the type is not indicated by FP:
  - Empty slots or even empty frames are found essential in 3GPP NR for base station power saving as well as introducing forward compatibility for future evolution in NR system. (Empty slots can be used for other purposes unknown by the initial release of the standard.)
- Combining multiple ACK transmissions into single slot. Considering the assumption that the traffic is roughly 1:10 there could be for e.g. 20 DL slots and 4 UL slots in 10ms frame. In this example, each UL slots could contain (H)ARQ feedback of 5 slots, or 2-3 (H)ARQ respectively, if double slots transmissions are used.

6.6.2 Proposal

DECT-2020 system is capable of:

- Dynamically adjust number of DL and UL slots in a frame.
- Performing adjustments locally i.e. each FP can make own local decisions, including mesh devices that operates as FP.
- The maximum rate of performing adjustments is in every 10ms frame. Adjustment rate should only depend on FPs signalling frequency.
- Signalling also allows FP to indicate certain pattern (semi-persistent configuration) that is in use for certain period.
- Signalling allows FP to create empty slots or even empty frames into system to introduce power savings and/or enable future evolution of standard.

It is possible to combine multiple ACK transmissions into single slot.

6.7 Contribution 3: Channel access considerations

When considering today's and future data communication, it is apparent that broadband data is dominated by TCP/IP connections serving different application level protocols such as HTML5. In massive Machine Type Communication (mMTC) the selection is perhaps less clear but UDP/IP is widely used and it is expected that IPv6 will be dominant in near future.

From TCP/IP one needs to recognize the impact of slow start and congestion avoidance protocol. Several different algorithms exist but basic principle is that both are sensitive to end to end latency and packet loss. Typically packet loss is well handled in radio protocols by using ARQ protocol providing error free delivery of the data. However, the impact of latency has drawn less attention before 5G era. For slow start, the connection setup delay, i.e. the first packet delay has significant impact how fast the data rate can ramp up and thus needs to be recognized. Then during the data transfer the end to end delay impacts directly to the bandwidth-delay product and achievable data rate with available transmitter window size. Finally, the TCP/IP sessions are typically quite short data bursts, even in case of video streaming sessions, where high data rate is used to download data to playout buffer and then the connection is "silent" for several tens of seconds.

For mMTC, the communication is generally driven by event-based application, where events are generally random processes. Therefore, deriving any device-based patterns for resource optimization and allocation is difficult. The driver for mMTC data transfer should be:

a) Energy needed to transmit the data - due to battery constrained systems.

b) Overall resource efficiency - due to high number of devices.

c) Infrastructure cost to support high number of devices.

d) Reliability.

In all above aspects the evaluation should also include all necessary UL/DL signaling, authentication/authorization and security and connection setup procedures to determinate overall efficiency.

To meet above requirements and to obtain competitive DECT-2020 design, the layer 2 protocol design should support:

1) Avoid performing full channel scan especially when load of the used carriers is low.

2) Allow carrier RSSI measurement just before actual transmission - not in previous frame.

3) Allow operation where authentication/authorization and security procedures are performed once for multiple data sessions & data packet transfer procedures.

4) Allow TX side to select modulation and coding scheme (MCS) for individual transmission.

5) Allow operating only single or multiple carrier frequencies.

6) Support ARQ, segmentation and concatenation.

7) TX Power control with open loop and closed loop link adaptation.

8) Avoid constant downlink control signaling (i.e. having a beacon bearer in every frame).

The points (1), (2) and (3) are essential for establishing the communication and sending randomly arriving first data packet with minimum delay to the indented receiver as well as minimize any higher layer signaling before able to transfer user plane data.

To support rapid changes in transmitted data amount provided by application or due to TCP slow start, the points (4) and (5) are essential. A carefully considered system design combining points (1) - (5) will ensure the rapid data transfer, fast release of used resources. This is also the most energy efficient way of operating as it minimizes the RF and Baseband activity time in the devices.

Point (5) is essential to provide robust and error free data delivery for applications and TCP as well as have high frame utilization as application data can be optimally matched with available physical layer frame sizes.

Finally points (7) and (8) are essential for minimizing interference caused by the system as well as minimizing both PPs' and FPs' energy consumption.

Therefore, it is proposed that the requirements outlined in points (1) to (8) above, are adopted as design principles for DECT-2020 Layer 2 (MAC, and above radio protocols).

7 MAC Protocol Function and PHY Services

7.1 A flexible lower MAC model for DECT-2020

7.1.1 Overview

The overall MAC design of DECT-2020 is currently ongoing. One of the several fundamental topics is the design of the frame slot structure and TDD operation.

Some important initial design decisions are already taken:

- One of them, is the decision to support efficiently both, scheduled services and immediate packet data access.
- In addition to that, current design ideas and (also IMT submission material) have considered a wide range of potential systems, from systems implementing 1 ms ultra-low latency, to systems optimized for LAN-like traffic requiring long LAN-like frames combined with MIMO for state-of-the-art bitrate figures.
- Frame structure will be based on 24 slots, with potential use of half-slots and fundamentally no restriction in direction (guard intervals foreseen in all slots). However a management strategy is needed to make possible a flexible operation, and it has to take into account the fundamental limitations in any TDD system. Possible expansion to 48 slots is taken into account.
- Several packet formats have been designed by the PHY layer team and are available for use. For the purpose of this discussion the packets can be classified in two groups: packets allowing random access (RAC, ULE, long formats) and packets for scheduled services (HE formats).
- Packets for scheduled services are allocated by the FP and are only usable after a MAC signalling process (with confirmed setup and release). They are intended i.e. for circuit mode or long burst traffics. On the other hand, random access may be usable for asynchronous traffic and are allowed to send U-plane traffic (in certain variants, considerable) before the first feedback from the other side.
- A broadcast mechanism over a pilot bearer with enough capacity is available and is able to provide update information of allocated, usable and "blind" resource.
- MIMO should be supported in both, random access and HE formats.

7.1.2 Types of possible systems

From previous work the following two extreme cases of systems have been identified:

- Systems intended to achieve the lower latency limit in URLLC systems. These systems operate with a frame structure up/down/up/down (in order to get the lowest latency) and can achieve sub ms latency for circuit mode traffic. They are part of the technology possibilities and are identified in the IMT-2020 submission material.
- Systems intended for LAN-like operation over relatively clean spectrum, intended to handle efficiently asynchronous IP traffic, with state-of-the-art data rates and supporting MIMO. These systems tend to operate with relatively long continuous frames, potentially up to 23 consecutive slots (or maybe 24) and use the long slot format combined with MIMO.

Operation of extreme case systems has several flaws that may require further study. The most obvious is that systems configured for one type of traffic may be very with other traffics that do not conform to the optimized paradigm. They may also be inefficient or even aggressive regarding spectrum use.

Also, the merits of the "long-LAN approach" under real spectrum situation and not entirely convincing, even operating with the expected traffic and with optimal conditions. The approach is very aggressive, especially if wideband channels are used (potentially a huge amount of data may be sent before receiving any feedback contaminating the complete carrier/slot raster giving to any other system attempting to access a near 100 % probability of collision).

In any case, it is assumed that multiple system configurations and FP resource management strategies will exist in the technology. Potentially a FP may decide dynamically how to behave according to traffic needs and such system operation modes or "models" may be dynamically announced to its registered PPs by means of the beacon broadcast channel.

7.1.3 The model "A", an intermediate system operation model suitable for many systems and types of traffic

7.1.3.1 Design targets

The exercise done by this contribution consists on designing an intermediate model, considered suitable for model many systems and types traffic. This intermediate model "A" may be considered a paradigm of the technology flexibility and may be used as a reference model for the calculation and optimization of the higher layer signalling.

In short the model A" should be able to provide:

- Both, scheduled and non-scheduled services, in both cases with efficiency and without major disturbance between them.
- Low latency circuit mode traffic, but not the lowest limit.
- Low traffic circuit mode services, like voice service.
- Efficient IP traffic operation, with MIMO activation, but without the aggressiveness towards other traffics and users of a "long-packet" model.
- Efficient ULE traffic.
- Efficient Low latency ULE, with a delay target in the range of 2,5 ms roundtrip.

7.1.3.2 The design

The model is oriented towards the consecution of 2,5 ms roundtrip delay in basic signalling exchanges. However it is still able to provide effective WLAN operation.

2,5 ms basic signalling exchanges refer to simple MAC exchanges (like ULE signalling). But it will also have impact in the higher layer signalling. Furthermore, what is also important, is the ARQ and HARQ mechanism that will also operate with the 2,5 ms roundtrip delay (an important figure of the technology).

7.1.3.3 Remarks

NOTE 1: The 2,5 ms basic signalling round time appears in some of the contributions to the IMT template.

NOTE 2: The 2,5 ms is a system configuration. The principles given can easily be applied to a 10 ms, to a 5 ms model or to a 1 ms model. However it is felt that 2,5 ms is a good compromise for high performance systems.

7.1.3.4 Elements taken from other MAC study areas

In order to build the design the following elements provided by other MAC study groups are assumed. Some of them are still open ideas proposed in the discussions and a few ones are original ideas proposed here by first time:

- There are HF and data fields in all random access packets transmitted with robust/efficient MCS respectively.
- HF has enough bits to include, in addition to PHY layer header, some MAC information.

- A MAC information that should be in the protected field (HF) is the info needed for the ARQ/HARQ operation, including quality feedback.
- Some MAC signalling (ULE) may also go in the protected HE field.
- To serve all uses, the following is proposed:
    - A variable size HF design where the size of the HF field (number of symbols) can also be dynamically changed and is part of the slot construction information.
    - The HF symbols will contain an initial "header" with slot construction information and a "tail" of variable size where several MAC channels may be carried. One of them will be the quality feedback channel. The quality feedback channel will be able to provide feedback to several forward bearers (not only to one) according to a schema that is part of the system "model" (announced by the FP). It may carry feedback selectively to both HF and data sections in forward bearers.
- The FP broadcast mechanism (on the beacon) is able to efficiently announce at least the following:
    - The "model" in use (model "A" in this case).
    - The position of any allocated "scheduled resource" (therefore, protecting it).
    - Any other "blind slot" information due to any other limitation.
    - Potentially, channel selection info from the FP to held the PP (can be combined with previous items).
    - The "listening for setup" capability of the FP (since, by power or other considerations) the base does not necessarily need to listen in all slots and in all carriers.

7.1.3.5 Basic design

Assuming the model targeted to 2,5 ms:

- One of each 6 slots is configured as "preferred-downlink" slot. Positions are assumed to be 1, 7, 13, 19.

Note that there is a 2,5 ms difference between slots:

- One of each 6 slots is configured as "preferred-uplink" (with 3 slot difference with downlink). Positions are 4, 10, 16, 22.
- All other slots are "flexible pool" and dynamically usable in any direction (assuming it is free and no conflict with the blind slot info).

7.1.3.6 Broadcast information

It is assumed that the FP broadcasts the following information over the beacon:

- The position of the given slots is broadcasted in some way by the beacon. It may be probably implicit in the model:
    - For example, a broadcast indicating system model = X may indicate position of slots as given in the example.
- The willingness for accepting setups, i.e. in which slots/carriers the FP will be listening for potential setups:
    - This may be implicit in the model or not.
- The message "sequence", as explained below:
    - Again, this may be implicit in the model or not.

7.1.3.7 The message "sequence"

The message sequence is an important element of this design. It consist on a pre-arrangement of slot sequences (not allocated yet), defined in the "model" or explicitly indicated by the FP that indicates when a PP should expect a feedback message when attempting any type of random access.

EXAMPLE: In the design sketch of this exercise (as shown in the Figure 4, see clause 7.1.5) it is assumed that any access attempt on slots 2, 3, 4, 5 or 6 will be replied over slot 7.

7.1.3.8 Configuration of scheduled services

This design sketch proposes that the allocation of scheduled services may be done with full flexibility and using as elementary unit the "simplex bearer".

A scheduled service may contain any combination of simplex bearers with any direction and without restriction. This is a new approach compared to legacy DECT.

EXAMPLE: A circuit mode two-way single bearer service may be allocated to bearers 1 (down) and 4 (up).

7.1.4 Slot allocation and message/response sequences

7.1.4.1 Slot allocation

The proposed strategy for slot allocation will be as follows.

7.1.4.2 Beacon bearers

- The FP allocates the beacon (or dummies) to any of the "preferred-downlink" slots.

7.1.4.3 Scheduled services

- The FP has flexibility to allocate resources. However, the expected strategy is that it allocates traffic to the "preferred up/down bearers" or to adjacent bearers (by this order) depending of bearer usage and service needs (e.g. delay):
    - The reason is that it is a good compromise strategy for the balance between non-scheduled/scheduled.

7.1.4.4 Random Access channels

- RAC access by PPs using RAC packed format may be done in either 1) "preferred-uplink" or 2) flexible slots. However, this depends on the "listening for setup" configuration set by (and announced) by the FP.
- Confirmation should be received at the agreed "downlink" slot. If this is not the case, collision handling should follow.
- If confirmation is received, but bad quality (e.g. HF received, data field not received) ARQ/HARQ can be attempted.
- The response command may contain instructions (slot position) for executing the ARQ/HARQ.

7.1.4.5 Packed mode traffic - single channel

- Packet mode (LAN-like) traffic access by PPs will use long format packets starting in either 1) "preferred-uplink" or 2) flexible slots. However, the packet burst should end no later than the last flexible slot before a preferred-downlink" slot.
- Access should also take into account the "listening for setup" configuration set by (and announced) by the FP.

In the Example, a burst of 5 slots (long format) is initiated at slot 2 ending at slot 6:

- At the first "preferred-downlink" slot position the PP will listed for feedback over this bearer.

- The feedback will potentially contain a MAC message and will contain quality bits for the up to 5 forward slots with selective feedback on HF and data fields. In case of MIMO selective feedback for space streams is also considered as an option:
  - Obviously this restricts the length of long format bursts, but also restricts the traffic send to the air without any reception assurance. It is a compromise.
- After receiving the feedback, the transmitter may chose:
  - If good feedback has been received:
    - to continue the transmission after the "preferred-downlink" if there is traffic. (A short format initial slot will be used);
    - shown in slots 8, 9, 10 in Figure 4 (see clause 7.15).
  - If no feedback from the expected slot has been received:
    - to stop transmission and to execute a collision handling strategy.
  - If feedback is received but not perfect:
    - to ARQ/HARQ badly received packets at MAC layer; and/or
    - to "slow down" transmission by changing MCS;
    - to change MIMO configuration;
    - to pass error information to higher layers (DLC) in order to trigger DLC retransmission of selected channels only (e.g. signalling).

7.1.4.6 ULE channels

- ULE channels are similar to RAC with the difference that 1) they contain some U-plane data and that 2) the transmission may continue.
- The strategy may be the same as per RAC channels.
- The response command may contain instructions (e.g. slot position) for continuation.
- RAC access by PPs using RAC packet format may be done in either 1) "preferred-uplink" or 2) flexible slots. However, this depends on the "listening for setup" configuration set by (and announced) by the FP.
- Confirmation should be received at the agreed "downlink" slot. If this is not the case, collision handling should follow.
- If confirmation is received, but with bad quality (e.g. HF received, data field not received) ARQ/HARQ can be attempted.

7.1.4.7 Packet mode traffic - wideband

Our current PHY layer design allows direct wideband "LAN-like" access over multicarrier channels. However at MAC layer there are two possibilities:

- To allow that in the initial access:
  - This is the most "LAN-like" approach and the optimal for data rate and response speed, assuming that there are no other traffics, other PP access and other services. Otherwise it may not be the optimal. It is the most spectrum aggressive strategy.
  - It also imposes to the other peer the requirement to be listening for setups in all carriers/all slots. This may be problematic or inconvenient.
  - Doing the same for downlink traffic may even be more problematic due to power considerations.

- Not to allow wideband initial accesses. This would mean that all initial packet accesses should be single-channel:
    - The access may change to wideband only after first received feedback.
    - Potentially the request to change may be inserted in the initial message sent be the PP.
    - Pros:
        - Much less spectrum aggressive and less collision probability.
        - The FP does not need to be in permanent "listening for setup" on all slots and carriers.
    - Cons:
        - Requires new long format and MIMO configuration after this point.
        - Reduces initial speed and adds access time.
- Configure it depending on the system and expected product use.

7.1.4.8 Packet mode traffic - downlink packets - Requires further study

The topic of downlink packet access requires further study. There is a trade-off between capacity, response speed and power consumption at the PP. Possible strategies are:

- Triggering it by over the pilot - power efficient but very slow.
- The PP are listening for setup on all slot/carriers to allow wideband access.
- The PP are listening for setup on a selection of slot/carriers allowing narrowband access only, changing to wideband after the first feedback exchange.

7.1.5 Example figure

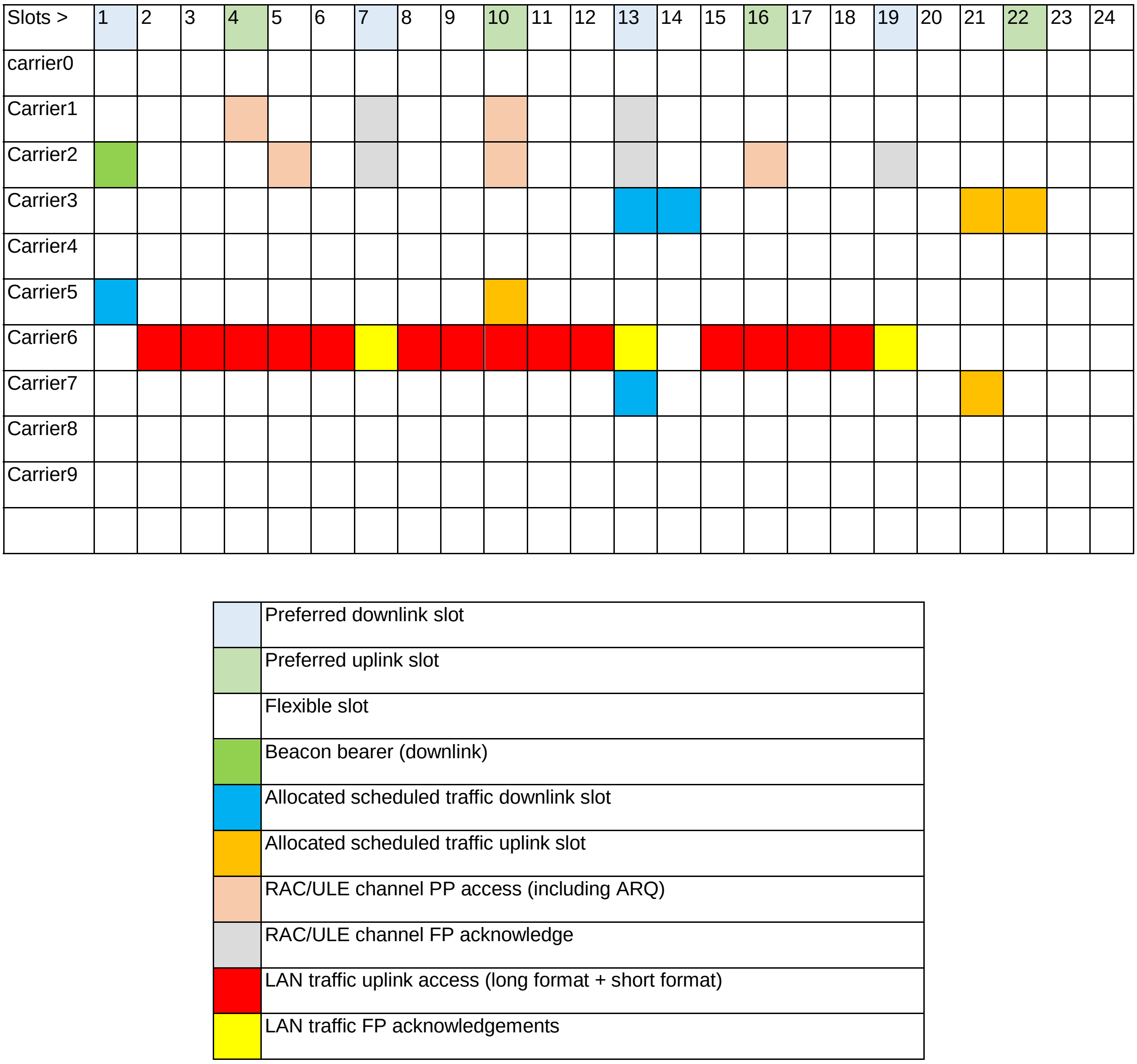


Figure 4: Example slot/traffic allocation

7.2 Void

This clause is intentionally left blank.

7.3 Dual mode solution for the beacon bearer

7.3.1 Summary

The current DECT-2020 MAC and PHY designs take as working assumption that there exist a beacon bearer (also known as "beacon" or "dummy bearer") which is broadcasted regularly by the FP. A proper PHY layer configuration is provided by the PHY layer (ETSI TR 103 514 [i.26]). Other MAC contributions foresee considerable amount information to be carried in this bearer if all technology planned features are deployed.

Assuming that there is a beacon bearer, the next decision is the choice of which MCS configuration should be used. This is a trade-off between the services provided by the bearer and the desired link budget. If the bearer has to provide complex information to aid in channel selection and slot selection processes, then a significant bitrate may be needed.

The needed bitrate may be further increased if ULE support mechanisms have to be provided. Proper Low Energy operation requires fast synchronization and paging response observing as few occurrences of the beacon bearer as possible. This also increases the required payload over the "beacon".

The possibility to use the "beacon" to also send some C/O information is also theoretically possible and interesting from architecture perspective. However it would further increase bitrate needs. In general, such operation would require a higher MCS configuration.

On the other hand, operation at maximum range and/or difficult radio conditions require lower MCS settings. As the beacon bearer has to be ready for potential PPs placed at an unknown range, the worst case should be assumed in "normal" operation. However, this "worst case" may in fact happen only in specific situations, so optimizing the whole design for range may compromise overall technology efficiency.

The proposed compromise solution attempts to address both parts of the problem in a balanced (and also future proof) way.

7.3.2 Assumptions

The current PHY layer design of the dummy/beacon bearer is shown in the next figure (from ETSI TS 103 514 [i.26]).

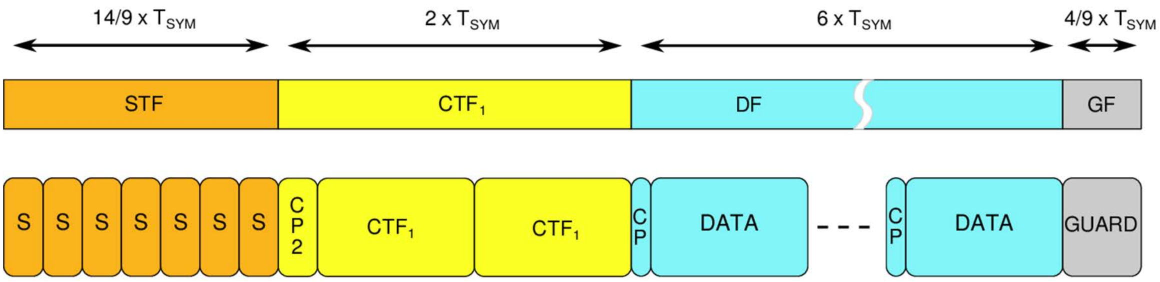


Figure 5: Beacon C/L downlink packet type

Table 2: Packet fields

Field	Description
STF	Synchronization training field. Same as in long packet format. Used to perform detection, AGC, coarse CFO estimation and frame synchronization.
CTF_1	First channel training field. Same as in long packet format Used to perform fine CFO estimation, frame synchronization and initial channel estimation. CTF is assumed to be transmitted duplicated in two symbols to allow fast synchronization (see [i.16]).
DF	Data field. Contains typically several MAC control channels. Transmitted with a very robust MCS. In ULE transmissions, at least the first data symbols are assumed to be transmitted with a very robust MCS, while a more efficient MCS may be used in the remaining symbols.
GF	Inter-slot guard interval.

The Physical layer aspects (such the need - or not- of duplication of CTF field) are not in the scope of this contribution. In short, the packet provides 6 usable symbols (named "DATA" in the figure). The figure does not preclude a fixed structure of HF/Data fields but leaves this to the MAC design.

7.3.3 Proposal

7.3.3.1 Idea

To better address all scenarios it is proposed to split the payload part of the bearer ("DATA") in two parts: one transmitted with a robust MCS and o the other with a higher MCS. The low MCS part would not be a mere PHY layer header, but it would have capacity to transmit several MAC channels (the most critical for basic system operation). The higher MCS part would transmit additional beacon channels with extended information required to implement all technology potential or to optimize operations and/or reduce response time.

7.3.3.2 Terminology and tentative construction

To align terminology and avoid confusion, the initial part transmitted with low MCS will be called "A part" and the corresponding symbols "A symbols". The final part transmitted with higher MCS will be called "B" part and "B symbols".

The "A" part will be transmitted with a fixed MCS that is assumed to be MCS = 1 for general systems, with option to use MCS = 0 by configuration, in special application:

- MCS = 1 provides 52 bits per symbol (see note).

The "B" part will be transmitted with a higher MCS, indicated by a header field at the beginning of the "A" part. In principle, this MCS may be fixed or dynamically variable. An intermediate MCS mode is assumed as most probable in most systems:

- As example for calculations, MCS = 6 will be assumed. MCS = 6 provides 234 bits per symbol (see note).

NOTE: The quoted value assumes single spatial stream (i.e. non-MIMO) and 1,728 MHz bandwidth.

7.3.3.3 Structure of the bearer

The "A" part will start with an initial physical header part providing information of the construction of the rest of the slot (such as the symbol split between A and B parts and the MCS mode of B part). Then, the remaining capacity would be used for transmitting several MAC channels that may be multiplexed based on a header coding with a similar structure to the general structure proposed for the "Data" (now "B") field.

Only a limited number of channels would be transmittable over the "A" part. The coding of the identifying header and payload structure can be optimized accordingly to use efficiently the scare bits in the "A" part.

The "B" part will be able to transmit any channel and may follow the general multiplexing structure proposed for the "Data" (now "B") field.

The "B" part will contain more detailed MAC channels (or transmitted more often) providing additional flexibility and/or fast response to the PPs.

Neither MIMO (nor MIMO training symbols) are assumed to be used in the beacon bearer (however it may be the subject to a separate proposal and discussion).

Additional "free" capacity may exist in the "B" part that can be used for additional C/L services (or perhaps C/O) when needed.

7.3.3.4 Example of capacity calculation

With the MCS values used as example, the "A" part will be able to carry 156 bits and the "B" part 702 bits (assuming single spatial stream (i.e. non-MIMO) and 1,728 MHz bandwidth).

7.3.3.5 Operation of the bearer and content of the fields

7.3.3.5.1 General

The "A" part should be able to provide basic operation to any device able to recognize the bearer. However, operation may require observation of several occurrences of the bearer due to multiplexing cycles used to administer the capacity.

Note that this is not very different to current DECT design.

The "B" part, will provide further information required for fast operation of the PP (or may transmit the same information more often). The "B" part information will be designed in a way that many PP actions may be performed with the observation of a single (or maybe a few) occurrence(s) of the bearer. This is similar to some principles used in ULE.

7.3.3.5.2 MAC Information in the "A" field

A device only able to observe the "A" part should be able to:

- Performing initial synchronization, FP selection and locking, however observation of several occurrences of the bearer will be needed.
- Receive "slow paging" commands. However observation of several occurrences of the bearer will be needed. (this is not very different to current DECT).
- Reception of blind slot and reserved channels information. However, observation of several occurrences of the bearer before setup will be needed.
- Receive system information channels. Observation of several occurrences of the bearer will be needed.
- Receive some channel selection aid information. However, more autonomy of the PP in channel selection is assumed in this case.
- Use (i.e. performing setups on) packet-mode, RAC and ULE bearers. However, some delay may happen due to the need of observation of several occurrences of the beacon and/or to run RSSI scans.
- Use scheduled channels.
- MIMO is assumed to be usable. However packet access over wideband channels (multi carrier) may be restricted due to limited information on channel selection.

In short, the "A" field should allow to perform all operations, but not with ultra-fast response time. A PP able to receive the "A" part only will be able to operate, but not with the maximum speed, response time and energy efficiency.

7.3.3.5.3 MAC Information in the "B" field

If the device is also to observe the "B" part of the beacon, then it is expected that, in addition to previous list, it will be able to:

- Synchronization and FP locking with only one observation of the bearer. This means that a proper identifier is transmitted in each frame. A shortened TPUI-like identifier is assumed.

NOTE: This optimized process refers mostly to re-locking of PPs already registered in the FP. For the initial registration, fast response is not as critical and "A" field mechanism may be used.

- Reception of paging commands at every frame (or more often if multiple beacons are used). Therefore the "B" part will include an extended paging channel.
- Complete information of blind slots, protected channels and channel selection information. This should allow a PP with a single observation of the beacon to start system access (even for ULE-like devices that may have not performed RSSI scans for a long time).
- Further options for setup slotxcarrier positions compared to observation of A part only.
- Information and commands instructing the PP to be ready to accept FP direct setups in specific channels (slotxcarrier) in order to be ready for downlink traffic. Depending on the system, this information may include the capability to accept wideband direct setups (multi-carrier) when needed.

In short, the "B" field should allow performing all normal operations plus high performance operations, with fast response times, maximum throughput and ULE-like energy efficiency.

7.3.3.5.4 Further ideas on beacon bearer content

The "B" part may contain spare capacity that may be used by other C/L (or even C/O) services when needed.

A PP able to receive the "A" part but not the "B" part may report it to the FP (by means of an specific MAC operation). The FP may consider the option to reduce MCS mode in "B" part according to the services configured in the system and to the needs of other PPs.

7.3.3.6 For further study

- The use and real need of a duplicated CTF symbol in the beacon bearer should be analysed and re-evaluated since it is expensive in terms of bitrate. It is believed that other mechanisms may exist to get synchronization (e.g. observation and correlation of multiple occurrences of the bearer).
- The possibility of using half-slots for beacon bearers is interesting from requirement perspective and should be considered in future evolutions. However, it is considered that with current PHY layer design this is not possible: a half slot would have 5 symbol intervals and 4 of them are used by different overheats leaving only one usable "data" symbol (that may be extended to two if CTF is not duplicated).
- For certain systems operation with beacons transmitted less often than once per frame may be required. This does not change the general principles given in this contribution (A+B design may still be the most convenient), but increases all reaction times.
- Operation with no continuous broadcast of beacon (no- emission-like) may be considered (however this is not trivial).

7.4 Void

7.5 Latency Considerations

7.5.1 Overview

The requirements from ITU-R WP5D (see [i.11]) define the requirements for latency as 20ms for control-plane (C-plane) latency, and 1ms for user-plane (U-plane) latency.

And for C-plane latency, it is defined as follows:

- Control plane latency refers to the transition time from a most "battery efficient" state (e.g. Idle state) to the start of continuous data transfer (e.g. Active state).

Specifically, for U-plane latency, it is defined as follows:

- User plane latency is the contribution of the radio network to the time from when the source sends a packet to when the destination receives it (in ms). It is defined as the one-way time it takes to successfully deliver an application layer packet/message from the radio protocol layer 2/3 SDU ingress point to the radio protocol layer 2/3 SDU egress point of the radio interface in either uplink or downlink in the network for a given service in unloaded conditions, assuming the mobile station is in the active state.

7.5.2 Control Plane Latency

The definition of the requirement from ITU-R WP5D is actually somewhat ambiguous.

EXAMPLE: What does the "most battery efficient state" mean? Should this also included deep sleep/hibernation states?

For the purpose of this study, it is assumed that the PP is "locked" to the FP's beacon and is receiving beacon broadcast information on a regular basis (i.e. it knows the slot/frame timing, it knows position and availability status of contention-based access resources).

The basic steps required to start/re-start communication (i.e. transition from "Idle" to "Active") are:

- PP raises bearer on Random Access Channel (RAC) (i.e. a contention-based channel access), and sends request for resources:
  - Include the delay/latency waiting for RAC availability (depends how many such channels are free).
- FP receives the RAC message, allocates resources, and respond on the next available response slot.

- PP receives the RAC response, and on next opportunity on the allocated resources the "continuous data transfer" can start.

Scenario 1

Assume that initial random "waiting" latency for the RAC slot availability is 5 ms on average.

Assume no contention (no other users competing for the RAC).

Assume the RAC "response slot" is 2 slots after the "request slot", i.e. leave 1 slot for processing message, and then respond on next available slot.

Assume that resources can be allocated and used immediately, in practice this will mean that the "continuous data transfer" can start 2 slots after the "response slot".

NOTE 1: Pre-transmit/post-receive signal processing time is essentially "absorbed" into the inter-slot waiting times, and so does not have be additionally accounted for.

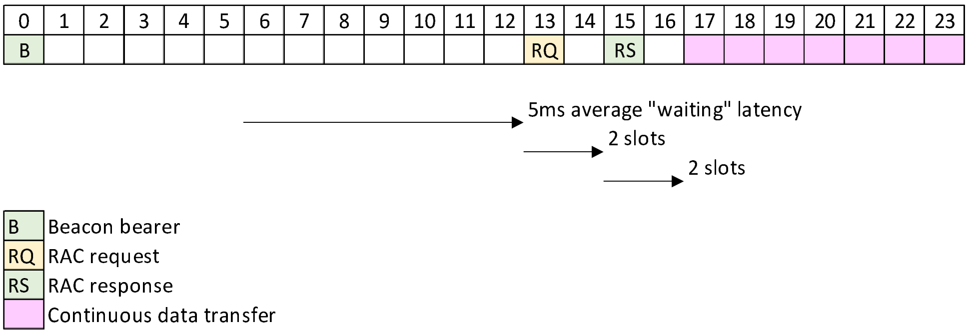


Figure 6: C-plane latency diagram

$$\text{C-plane latency} = 5 + 2 \times 0{,}416 + 2 \times 0{,}416 = 6{,}664 \text{ ms}$$

This scenario easily meets the 20 ms target requirement.

Scenario 2

Assume that initial random "waiting" latency for the RAC slot availability is 5 ms on average.

Assume no contention (no other users competing for the RAC).

Assume the RAC "response slot" is 5ms after the "request slot", i.e. following more traditional DECT framing.

Assume that resources are allocated in the following frame, i.e. a further 5ms, according to more traditional DECT framing.

NOTE 2: Pre-transmit/post-receive signal processing time is essentially "absorbed" into the inter-slot waiting times, and so does not have be additionally accounted for.

$$\text{C-plane latency} = 5 + 5 + 5 = 15 \text{ ms}$$

This scenario comfortably meets the 15ms target requirement.

Scenario 3

In the case of FP-initiation, the FP needs to page the PP first (on the "beacon"), and then the PP establishes as in either of the 2 previous scenarios, or using a form of "direct setup" initiated by the FP.

Assuming that the system is not loaded (single user), then there is no paging contention and so the FP can perform high priority paging on the beacon transmission immediately. Paging/beacon multiplexing schemes have not been designed yet, and so additional latency maybe present in practice. This requires further study.

7.5.3 User Plane Latency

There are a number of key points that can be gleaned from the ITU-R WP5D requirement:

- The latency is measured from ingress point at layer 2/3 boundary until egress point of layer 2/3 boundary. OSI layer 2 is the Data Link Layer, and OSI layer 3 is the NWK layer. However, since much of layer 2 is performed in s/w it is difficult to compute processing time for this - since any implementation could simply use a faster processor to reduce the latency. So, for the purpose of analysis it is assumed that Sw processing delay is essentially 0 ms.

- The stated requirement is only for small packets, and it can be assumed that such packets do not require fragmentation/reassembly by Data Link Layer, so there is no additional contribution to delay from this perspective.

- The analysis is for a "one-way latency, in either up-link or down-link for both up-link and down-link". This basically means that the analysis is only required for unidirectional data transfer and both directions are required to meet the requirement (but separately).

- Finally, the requirements assume unloaded conditions (i.e. single-user/no contention) and small packets (e.g. 0 byte payload + IP header)

Scenario 1

Slot configuration of alternating up-link/down-link slots. Transmission duration is 1 slot, using appropriate packet type such as HE (High Efficiency), and a robust MCS. Assume no re-transmission is required, i.e. no HARQ process needed.

The random "waiting" latency is 0 ms in best case, and 2 slots in worse case (i.e. the transmit opportunity is just missed, and so there is to wait 2 slots). So, on average the random "waiting" latency is 1 slot (0,416 ms). Transmission time is 1 slot, plus pre-transmit (assume 1 symbol) and post-receive (assume 2 symbols).

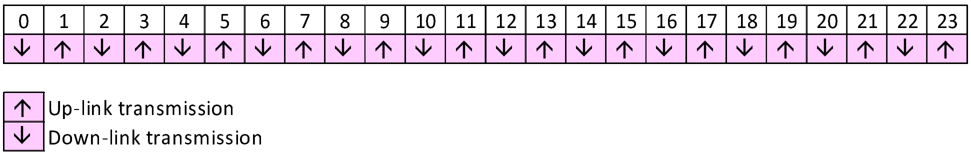


Figure 7: U-plane latency diagram (alternating uplink/downlink)

$$\text{U-plane latency} = 0{,}416 + 0{,}416 + 3 \times 0{,}0416 = 0{,}9568 \text{ ms}$$

This is within the 1ms target requirement.

NOTE 1: The above assumes that packets are arriving randomly, and this is the average case (i.e. 1 slot latency on average). However, in the case of "full buffer" scenario, which is allowed within the simulation/evaluation criteria then only the 1st packet arrives at random, after which it is assumed that packets are in the buffer and so are available for immediate transmission. As such, in the full-buffer scenario, the random "waiting" latency (after the 1st packet) will be 0ms, and so the total U-plane latency under such conditions is 0,5408 ms.

The model assumes that HARQ is not needed, which maybe the case if robust MCS are, e.g. BPSK ½ rate, or QPSK ½ rate.

NOTE 2: An HE (e.g. type III) packet can carry 8 data symbols, which yields 26 bytes using BPSK ½ rate, or 52 bytes using QPSK ½ rate, using 1 DECT frequency channel.

The data requirement is for 32 byte packet. On top of this the PHL and MAC header over-heads, and any CRC/MIC also needs to be factored in, if used. It seems that QPSK ½ rate coding may be sufficient for 1 DECT frequency channel, and of course higher data rates can be achieved by using multiple DECT frequencies and/or spatial streams.

7.6 HARQ

7.6.1 Overview

7.6.1.1 Simple HARQ

Hybrid automatic repeat request (hybrid ARQ or HARQ) is a combination of high-rate forward error-correcting coding and ARQ error-control.

In standard ARQ, redundant bits are added to data to be transmitted using an Error-Detecting (ED) code such as a Cyclic Redundancy Check (CRC). Receivers detecting a corrupted message will request a new message from the sender.

In Hybrid ARQ, the original data is encoded with a Forward Error Correction (FEC) code, and the parity bits are either immediately sent along with the message or only transmitted upon request when a receiver detects an erroneous message. The ED code may be omitted when a code is used that can perform both Forward Error Correction (FEC) in addition to error detection, such as a Reed-Solomon code. The FEC code is chosen to correct an expected subset of all errors that may occur, while the ARQ method is used as a fallback to correct errors that are uncorrectable using only the redundancy sent in the initial transmission.

Hybrid ARQ performs better than ordinary ARQ in poor signal conditions, but in its simplest form this comes at the expense of significantly lower throughput in good signal conditions. There is typically a signal quality cross-over point below which simple hybrid ARQ is better, and above which basic ARQ is better.

The simplest version of HARQ, Type I HARQ, adds both ED and FEC information to each message prior to transmission. When the coded data block is received, the receiver first decodes the error-correction code. If the channel quality is good enough, all transmission errors should be correctable, and the receiver can obtain the correct data block. If the channel quality is bad, and not all transmission errors can be corrected, the receiver will detect this situation using the error-detection code, then the received coded data block is rejected and a re-transmission is requested by the receiver, similar to ARQ.

However, there are much more advanced ways to utilize HARQ, such as "soft-combing" rather than the simple HARQ described so far.

7.6.1.2 Hybrid ARQ with "Soft-Combining"

If incorrectly received coded data blocks are stored at the receiver rather than simply discarded, then when the re-transmitted block is received, it might be possible to compute the correct data by combining the two blocks. This is called Hybrid ARQ with soft combining. While it is possible that two given transmissions cannot be independently decoded without error, it may happen that the combination of the previously erroneously received transmissions gives us enough information to correctly decode.

There are two main forms of soft-combining:

- Chase combining:
  - Every re-transmission contains the same information (data and parity). The receiver uses maximum-ratio combining to combine the received bits with the same bits from the previous transmission(s).
- Incremental combining:
  - Every re-transmission contains different information to the previous one (same data/different coded bits). Different coded bit are usually generated by puncturing the encoder output. Thus at every re-transmission, the receiver gains extra information.

7.6.1.3 "Stop-and-Wait" or "Selective Repeat"

HARQ can be used in stop-and-wait mode or in selective repeat mode. Stop-and-wait is simpler, but waiting for the receiver's acknowledgment reduces efficiency. Thus multiple stop-and-wait HARQ processes are often done in parallel in practice: when one HARQ process is waiting for an acknowledgment, another process can use the channel to send some more data.

7.6.1.4 "Adaptive Re-transmission" vs "Non-adaptive Re-transmission"

Since the data is being re-transmission, there is an option to transmit it again using the same or different modulation and coding. Non-adaptive uses the same modulation and coding; adaptive uses different modulation and coding. Adaptive re-transmission allows better behavior in case of degrading channel quality.

7.6.2 HARQ in DECT-2020

Since IMT-2020 does not impose any specific requirements to use HARQ, and a sophisticated HARQ system does not seem to be essential in order to meet the target requirements (for mMTC and URLLC) then it appears to be not necessary to define a *sophisticated* HARQ system at the onset for DECT-2020.

Nevertheless, it should be noted that since the DECT-2020 PHL utilizes channel coding scheme (e.g. BCC or LDPC) to correct errors on the received OFDM symbols (which is an essential function to avoid loss of subcarriers due to fading) and it also uses ARQ for retransmission of bad packets (for acknowledged data transfer services) then, by definition, DECT-2020 employs HARQ (Simple Type I).

NOTE 1: HARQ is only possible where there is an ACK/NACK back from the receiver. As such, HARQ is not possibly for pure broadcast-based channels, such as beacon transmissions.

NOTE 2: HARQ might not be possible or desirable for some data channels, e.g. unacknowledged data or where low latency is required (i.e. cannot wait for re-transmission for low-latency audio).

The simplest approach for HARQ utilizing soft-combining is to use "stop-and-wait" protocol (aka "alternating bit protocol"), with "chase combining" (i.e. the re-transmissions contain the same data and parity bits).

Figure 8 shows transmission of a data packet, with errors and re-transmission, but no soft-combining. The transmitter will continue to transmit the packet until it is received correctly (and ACK'd) or until a time-out.

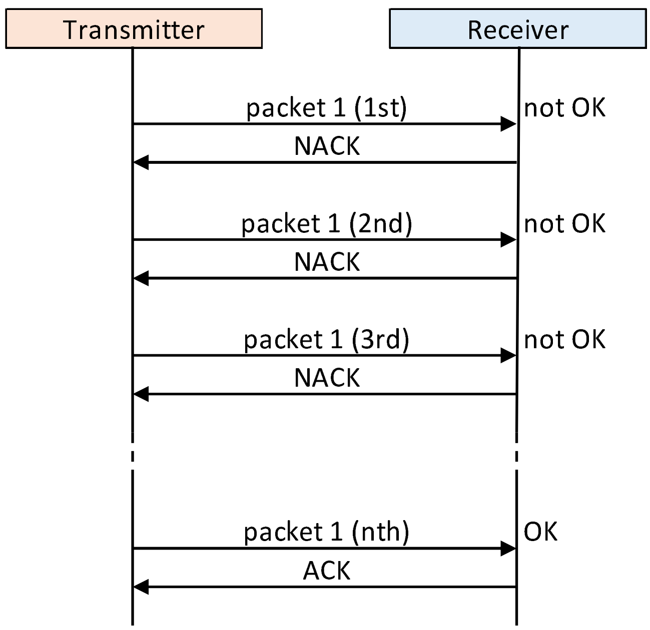


Figure 8: Re-transmission after errors (no HARQ)

Figure 9 shows soft-combining HARQ, using stop-and-wait and chase combining. The receiver is able to combine the 2 transmissions of "packet 1" in a way that produces a corrected packet.

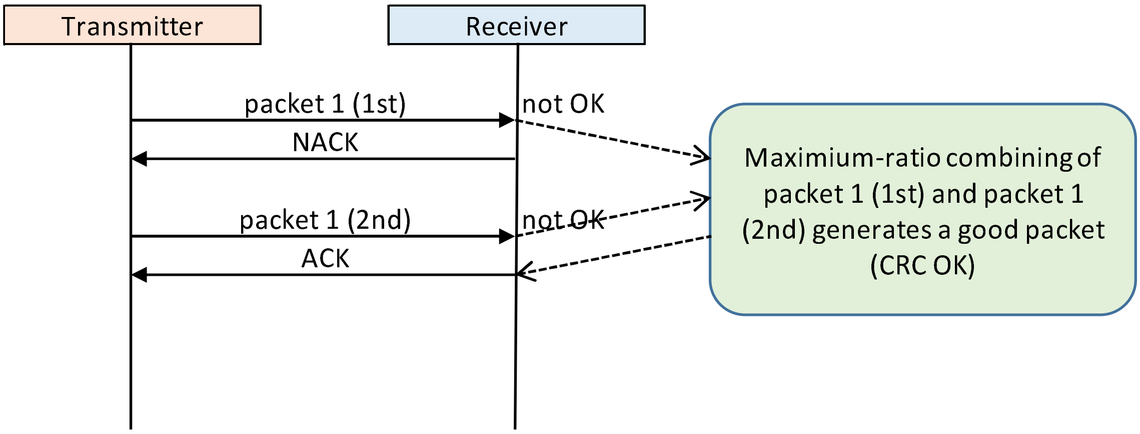


Figure 9: Re-transmission after errors (HARQ with soft-combining)

More complex approaches, including "selective repeat" and "incremental redundancy" are left for further study.

7.6.3 HARQ Implementation Considerations

The HARQ mechanism in the PHL is usually implemented as an adaptation of the "stop-and-wait" (SAW) process. Once a packet is send from a particular process, it waits for an ACK/NACK. Till it receives ACK/NACK, the process will be in-active state and will not process other packets. This would significantly impact the throughput, and so multiple SAW processes are used in parallel. For example, in LTE 8 SAW/HARQ processes are employed. When the 1st process is waiting for an ACK, the 2nd SAW process will send data, and so on with the remainder of the eight processes. The MAC layer manages these HARQ (SAW) processes.

There are implementation considerations regarding the number of SAW/HARQ processes. For example, memory to buffer the received packets, and each process needs to be identified by some address/process id so that the receiver and transmitter are synchronized as to which process is being used for a particular packet.

Detailed design and analysis of the HARQ mechanism requires further study.

7.7 PHL Header Design

7.7.1 Design Rationale

The DECT-2020 PHL packet formats contain a Header Field (HF) which maybe 1 or 2 OFDM symbols depending on packet type/variant.

A major goal of the HF is that it should contain everything needed to decode the rest of the transmission properly. This is particularly true for the "long-preamble" packet formats, or any packet that does not rely on previous negotiation or scheduling.

However, for packets which are scheduled and have parameters that are either known in advance (e.g. because of the type/service) or because of prior negotiation then the requirements on the HF are less (because some parameters are already known).

Ideally the HF should be as small as possible. It should be an integral number of symbols (e.g. 1 or 2). It should be transmitted using a robust MCS. It should be protected by a CRC (since wrongly decoding the HF could lead to misinterpreting the whole packet).

7.7.2 Header Field Contents

7.7.2.1 Header type/version/extension

The ability to support different "types" of header (e.g. for different packet types, or evolution of the design) would be advantageous. A typical type/version field could be 3 bits, allowing 8 possible types/versions, or 7 with 1 reserved for "extension". Alternatively, a single extension bit could be used to allow the basic design to be extended at a later date.

7.7.2.2 Modulation Coding Scheme (MCS)

The DECT-2020 PHL design specifies 12 MCS codes. For example, Table 3 shows the data rates for these MCS at 1,728 MHz channel with 1 spatial stream.

Table 3: Modulation Coding Scheme (MCS) summary

MCS	Modulation	R	N_{BPSC}	N_{SD}	N_{SP}	N_{CBPS}	N_{DBPS}	Data rate
0	BPSK	1/2	1	52	4	52	26	624
1	QPSK	1/2	2	52	4	104	52	1 248
2	QPSK	3/4	2	52	4	104	78	1 872
3	16-QAM	1/2	4	52	4	208	104	2 496
4	16-QAM	3/4	4	52	4	208	156	3 744
5	64-QAM	2/3	6	52	4	312	208	4 992
6	64-QAM	3/4	6	52	4	312	234	5 616
7	64-QAM	5/6	6	52	4	312	260	6 240
8	256-QAM	3/4	8	52	4	416	312	7 488
9	256-QAM	5/6	8	52	4	416	-	-
10	1024-QAM	3/4	10	52	4	520	390	9 360
11	1024-QAM	5/6	10	52	4	520	-	-

NOTE: During review of ETSI TR 103 514 [i.26], the possibility of removing the 1024-QAM modulation was discussed (it is hard to achieve in practice) and instead allowing a higher number of spatial streams (up to 8).

Clearly, for 12 MCS index, 4 bits should be used (with some codes un-used). However, this could be reduced if some of the MCS are not used in all cases.

7.7.2.3 Number of Spatial Streams

The number of supported spatial streams should be 6 or 8. Clearly 3 bits should be sufficient to encode the number of spatial streams used by the transmission.

7.7.2.4 Channel Coding Algorithm

The DECT-2020 PHL design assumes the use of Binary Convolutional Codes (BCC) as default, with the additional future possibility of using Low Density Parity Check (LDPC).

1 bit should be used to indicate which channel coding scheme is used.

NOTE: BCC requires some "padding bits" and "tail bits" (all zeros) in order to allow the coder to function. LDPC does not require this, but is more computationally expensive.

7.7.2.5 Transmission Bandwidth

The DECT-2020 PHL design allows for transmissions to occur on multiple contiguous channels (i.e. 1, 2, 4, 8, 12, 16).

NOTE: The number of channels are a power of 2. In the case of the 12 channel BW, this is achieved by using part of total BW of 16 channels - specifically designed to give ~20 MHz bandwidth.

Assuming that all of these BW options are required, then 4 bits are needed.

7.7.2.6 Transmission Length

Length could be defined in bytes, OFDM symbols or slots.

Bytes: Because of the variable nature of the MCS, transmission BW and transmission length, the possible length in bytes actually has quite a large range. Assuming N_{SS} = 6, MCS = 11, the N_{DBPS} = 46 800 and so with a 12-slot transmission, the max data = (46 800 × 10 × 12) ÷ 8 = 702 000 bytes! This would need 3 bytes to specify the length. Of course, this theoretical size is probably never going to be used in practice.

OFDM Symbols: This is a more natural and flexible way to define the length. It allows for variable length packets (e.g. not just half-slot/full-slot). The N_{SS}, MCS and transmission BW are now irrelevant for defining the length of transmission. Assuming a 12 slot transmission, the maximum length in OFDM symbols = 10 × 12 = 120, which can be represented as 7 bits.

Slots: Assuming a 12 slot transmission, this can be represented as 4 bits. However, 1 bit would also be needed to indicate whether the last slot was full-slot/half-slot; and another bit to indicate if the last slot required a guard symbol or not. In total it would require 6 bits to represent the length using slots.

In conclusion, it seems representing the length as a number of OFDM symbols is the most flexible scheme, and only requires 1 more bit than if using slots.

NOTE: The length of payload (expressed in bytes) is still required, but this information can be in the MAC header, and not necessarily needed in the PHL header.

7.7.2.7 Extras

Padding required to make the total number of bits a multiple of the number of coded bits in an OFDM symbol.

When BCC convolutional coding is used, then typically 6 "tail" bits are required to return the encoder to the "zero state." This procedure improves the error probability of the convolutional decoder, which relies on future bits when decoding and which may be not be available past the end of the message.

NOTE 1: There may be some techniques to mitigate the requirement for these "tail" bits. This requires further study.

NOTE 2: LDPC does not require these "tail" bits.

If there is capacity, it might be useful to use some HF bits for MAC information (e.g. ACK, HARQ, etc.). This requires further study.

7.7.2.8 CRC

The HF should be protected by a CRC. This allows the receiver to ensure that the HF is decoded properly. This is important since an incorrectly decoding packet could not only affect the current packet but also others. For example, suppose the length field was incorrectly decoded, resulting in the PHL receiving additional symbols.

There are various options for CRC length, depending on the required protection. The HF is relatively small (in terms of bytes) and so, it is assumed that a short CRC will provided adequate protection. This requires further study.

7.7.3 Header Field Configurations and Size Estimates

Configuration 1

Typical header for "long-preamble" type of packet, i.e. it contains everything needed to decode the contents, without any previous negotiation or assumptions.

Table 4: Header field size (configuration 1)

Name	Size (bits)	Comment
Header type/version	3	Allows for flexibility/future-proof extensions, etc.
MCS	4	Allows up to 16 MCS
Number of spatial streams	3	Allows up to 8 spatial streams
Channel coding algorithm	1	BCC or LDPC
Bandwidth	4	Allows up to 16 aggregated channels
Length	7	In OFDM symbols (up to 128 symbols, i.e. 12 slots)
Tail bits	6	(Only if needed)
CRC	8	
	36	**Total**

Configuration 2

Reduced header, assuming 1 spatial stream and BW of 1.

Table 5: Header field size (configuration 2)

Name	Size (bits)	Comment
Header type/version	3	Allows for flexibility/future-proof extensions, etc.
MCS	4	Allows up to 16 MCS
Channel coding algorithm	1	BCC or LDPC
Length	7	In OFDM symbols (up to 128 symbols, i.e. 12 slots)
Tail bits	6	(Only if needed)
CRC	8	
	29	**Total**

Configuration 3

Reduced header, assuming 1 spatial stream and BW of 1, but with additional optimization/restrictions to reduce the size further.

Table 6: Header field size (configuration 3)

Name	Size (bits)	Comment
Header type/version	3	Allows for flexibility/future-proof extensions, etc.
MCS	3	Allows up to 8 MCS (i.e. using a reduced set of optimal MCS values)
Channel coding algorithm	1	BCC or LDPC
Length	6	In OFDM symbols (up to 64 symbols, i.e. 6 slots max)
Tail bits	6	(Only if needed)
CRC	6	Reduced CRC protection
	25	**Total**

Other optimizations are of course possible, especially in the case where the connection is pre-negotiated, in which case the HF can be quite minimal as it only needs to cover the parameters that may change from packet to packet.

Configuration 4

An optimized format is proposed here with the goal of being able to implement all major requirements fitting into 26 bits (the capacity of a symbol at MCS=0).

Table 7: Header field size (configuration 4)

Name	Size (bits)	Comment
Header type/version/extension	2	00 normal value. All others reserved for future-proof extensions.
MCS (B field)	4	Allows up to 16 MCS
Number of spatial streams (B)	3	Allows up to 8 spatial streams
Channel coding algorithm	-	Not used (If needed, will be coded in the header type)
Bandwidth	2	Allows arbitrary number of aggregated channels. See text for coding
Length	5	Length of an L burst in half-slot units. Maximum length is 16 slots
A1 field	2	Insertion or not of the A1 field
CRC	8	
	26	**Total**

The coding of the different bits is as follows:

Header type/version/coding: 2 bits coded to '00'. All other values are reserved for further extensions. They can be used for indicating any alternative channel coding algorithm.

MCS: the full range of MCS values is allowed. See discussion at the end of this clause.

spatial streams: up to 8 spatial streams. See discussion at the end of this clause.

Bandwidth: the following coding is proposed to efficiently code any bandwidth value:

00	Single carrier transmission
01	Lower carrier in a multi-carrier transmission
10	Upper carrier in a multi-carrier transmission
11	Intermediate carrier in a multi-carrier transmission

NOTE 1: Half bandwidth is not considered in the proposal.

Length: Indicates the length of the burst coded in half slot units. The field assumes the worst case of L-S approach. The maximum length codifiable is 16 full slots

00000	Half slot
00001	Single full slot
any other value	Length is = value + 1 half-slots
11111	maximum value = 32 half-slots = 16 full slots

A1: indicates the possible insertion of an A1 field with reduced MCS and no MIMO

00	No A1 field
01	A1 is 1 symbol
10	A1 is 2 symbols
11	A1 is 3 symbols

NOTE 2: The rule for determination of MCS for A1 field is a separate topic. It may be equal to A0 or may be a value (reduction) taken into account the B field MCS.

With this design, all major cases are coded into 26 bits. Therefore it is implementable on an A0 field transmitted with MCS = 0 (BPSK, R=1/2).

The following additional remarks are made:

- It is assumed that at least MCS = 0 and MCS = 1 will be usable in field A0 with auto detection: normal operation will use MCS = 1 being MCS = 0 exceptional for limit radio conditions.
- If MCS = 1 is used, then an additional container of 26 bits will exist in the A0 field. This may be used for privileged MAC information or commands. Example, the MAC packet sequence number or an identity.
- With MCS = 0, the mechanism of indicating A1 insertion may be used to add a robust transmitted symbol that may carry critical MAC information.

- The "length" assumes the L-S approach, which is the worst case. In I-O-C approach, only one bit would be needed to indicate "continuation" with (potentially) another bit needed is the special case of I symbol with half-slot duration is implemented. Then, 2 bits would be spare and could be put to other uses.
- With L-S approach the maximum codifiable length is 16 full slots. This can be easily extended (e.g.to 23) by restricting half-slot granularity at long bursts.

Further optimizations may be done splitting the design on two variants: one to be transmitted over MCS = 0 (that should be limited to 26 bits) and another to be used over MCS = 1 (where more bits are available):

- The MCS = 0 version (since it is intended for low quality radio conditions) may restrict the use of the higher MCSs and higher MIMO configurations (or MIMO at all). Then some bits may be reserved for additional functions or for the MAC sequence number.
- The MCS = 1 version, intended for normal use, may code the whole set of configurations with some additional flexibility and with the MAC sequence number.

7.8 Proposal of unification of the initial part of all random access PHY layer packets and additional observations on the physical layer

7.8.1 Proposal of modification to the "long format" packet format

7.8.1.1 Rationale

At this design stage there are three variants of packets that can be used in a true (unannounced) random access. They are:

- RAC (Random Access Channel).
- ULE.
- "Standard packet" - long format.

All packets use the same PHY layer CP. They differ in the size of the synchronization field and the inter-slot space:

- "Standard packet" - long format" uses a 2 x STF field and a 1 x inter-slot space.
- RAC and ULE use 14/9 STF and 4/9 inter-slot space.

The CTF field is duplicated in all designs.

The inter-slot space in time (s) is 41,7 us for the standard long format and 18,5 us for the RAC and ULE types.

Significant simplification at MAC layer (and also at PHY layer) will be possible if the initial part of all packets with random access capability is unified. In this context, the "initial part" means the beginning of the packet until the first A symbol that carries the PHY layer header.

The simplification is evident, i.e. in the implementation of the Rx side and in the listening window (currently there are two possible starting points for random access packets).

On the other hand, as long as RAC and ULE packets exist in a system, there is no point at all in keeping an extended inter-slot space associated only to standard long packets.

The STF field of RAC/ULE packets (9/14 T_{SYM}) should be enough since RAC and ULE can operate with it.

7.8.1.2 Change request to PHY layer

It is proposed to unify the format of the initial part of the standard long packets to mimic the current RAC and ULE packets.

Therefore, the standard long packet format (initial slot) is re-defined as follows:

- Inter-slot space will be 4/9 T_{SYM}.
- STF will be 14/9 T_{SYM}.
- CTF will be unchanged (2 x T_{SYM}) .
- First data symbol (formerly named HF and renamed A0 - see clause 7.9) will be the same.
- Packet formats will only start to diverge after the A0 symbol.

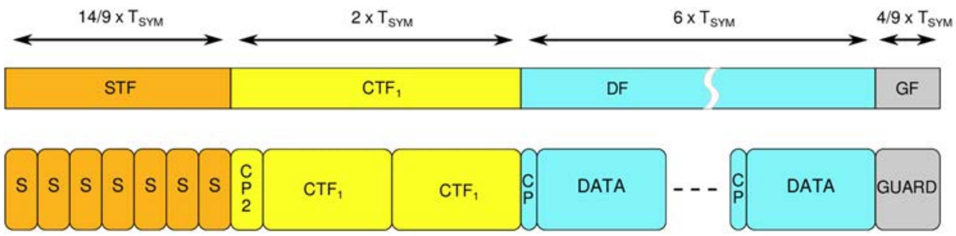


Figure 10: Proposed format formal for all random access packets (see next for change in convention for the guard time)

7.8.2 New editorial conventions - position of the inter-slot guard and new A/B terminology

To simplify descriptions the following changes to editorial conventions will be used:

1) Slots are assumed to start with the inter-slot time.
2) The following terminology is used for the formerly named HF/Data symbols:
    a) First data symbol (formerly HF) is named A0. It is transmitted with MCS 0 or 1.
    b) There may exist additional "A" symbols, transmitted with the same low MCS as A0. When used, they are named A1.
    c) Symbols transmitted with high MCS are named B symbols.
    d) The CTF for MIMO (when used) is assumed to be placed after last A and before first B (however this is subject to discussion).

This terminology applies to all packet formats.

NOTE 1: The PHY header is typically carried inside the symbol A0. However other fields may exist in A0, e.g. lower MAC headers.

NOTE 2 The low MCS part of the slot ("A1" symbols) may be more than one symbols. This is required, e.g. in beacon bearers (see clause 7.3). It is believed that this feature may also be useful in RAC and in ULE packets.

NOTE 3 The number of "A1" symbols is coded in the PHY layer header (now a part of A0) together with other information of packet construction.

7.8.3 Proposal of extension of the "A" symbols

Some packets require extended "A" symbols to transmit information with robust MCS. This has already been identify for the beacon format (see clause 7.3) but is also foreseen in the ULE and even RAC formats.

Therefore, it is allowed to construct the packets with more than 1 "A" symbol (formerly named HF). When this happens, the first symbol is named A0 and all other A1. A0 always exists. A1 may exist or not.

The following principles are used:

- The presence of A1 symbols and their number is indicated in the PHY layer header that is transmitted in A0.
- A1 uses the same MCS as A0.
- A0 is coded and should be decodable independently without waiting for A1. In multicarrier channels each A0 carrier x symbol is an independent unit from coding structure.
- On the other hand, all A1 symbols (in a single-carrier slot) are coded jointly as a block.
- The case of A1 coding in multi-carrier is left for further study.
- Solutions may considered using the multi-carrier subcarrier map (more efficient) or coding each carrier separately (easier to decode).

7.8.4 Further observations to the PHY layer

7.8.4.1 General

The following further observations are made to some PHY layer topics. They may be the subject of further investigation and change requests.

7.8.4.2 STF time is perhaps too large

Even with proposed reduction to 14/9 x T_{SYM}, the proposed STF is large in terms of absolute time. It should also be taken into account that there is a reference timing provided by the beacon and that there has been a previous synchronization process.

It is recommended to further study the topic at PHY layer simulator and reviewing the solutions used by other technologies.

7.8.4.3 CTF time is perhaps too large

The duplication of CT to allow easy detection is common practice in other technologies. However, when translated to absolute time, our design uses a very large time. It is recommended to study the solutions used by technologies with comparable symbol time (e.g. LTE).

7.8.4.4 The current design of the standard long format packets requires knowing the length of the whole packet at the beginning

This is consequence of considering the packet a single variable-sized packet instead of a concatenation of standard sized "slots". From MAC perspective it may be better considering the packet as a "burst" of standardized-size packets with the end of the burst signaled only by a flag at the beginning of the last element. Nevertheless there are pros and cons for both approaches.

7.8.4.5 The current design of the standard long format packet is probably too weak (channel tracking) for long packets

Current design allows the transmission of packets of up to 23 or 24 slots without intermediate tracking symbols. Depending on channel change and tracking strategy, this may be excessive. Analysis requires a channel model simulator.

7.8.4.6 MIMO training in long format is inefficient and does not take into account the available frequency resolution

ETSI TR 103 514 [i.26] proposes up to 5 additional symbols used just for MIMO tracking. However a different design may be possible by splitting subcarriers in the same symbol for training each special stream. It is believed that with 27 kHz subcarrier resolution, and in the radio conditions compatible with high MIMO configurations, this approach should be possible.

7.8.4.7 The current design of the standard short packet contains unnecessary STF and may be optimized

The rationale is as follows: the HE format can work without STF symbols and two occurrences of HE slots are separated up to 10 ms (they occur in the same carrier). The short format is intended for continuation of WLAN transmissions after an initial transmission started by a long format packet. Separation between the last slot of the long format packet and the beginning of the short format is less than 10 ms and both packets also use the same carrier. Therefore, there appears to be no reason why a short format needs STF fields and the HE does not.

Short format is not a random access format. It is a continuation format to be transmitted following a long format packet. Reception of short format packets may happen only at expected time windows and with absolute slot and bit synchronization related to the previous long format packet. They should also use the same carrier. From radio perspective the case is similar to HE packet reception.

Therefore, it is proposed to remove all or part of the STF field in short packet format to get an additional data symbol. The inter-slot space may be unified with all other packets.

Proposal for short format:

- Inter-slot space: 4/9 T_{SYM}.
- STF or extension of 1st symbol (CTF1) CP (to be decided by the PHY layer team): 5/9 T_{SYM}.
- CTF1 (including MIMO tracking): 1 T_{SYM}.
- Data symbols (usually all B type): 8 T_{SYM}.

7.9 Proposal for Modified PHL Packet Format and Channel Estimation

7.9.1 Background

It was observed that there were some inconsistency in short pre-ample packet formats. The short pre-ample packet type was designed to support "more or less" continuous transmission (i.e. continuous transmission with some discontinuities or interruptions) where channel estimation for HFS is done from previous packet.

It has been argued that it does not make much sense as the channel estimation is old (one more slots old), and at least one CTF is provided for data part anyhow. Thus re-ordering by HFS and CTF symbols the channel estimation can be updated for HFS and for data without no additional cost in overhead. Secondly this improves similarities between different packet designs reducing implementation and testing burden.

This principle can be extended to also cover other packet formats.

Finally, during discussion of the control header of the packet (HFS-field), it was found that single symbol can carry quite limited amount of content and utilizing complete symbol for channel estimation has a significant overhead. Thus, a scheme to reduce this overhead and same time increase the control header size is considered.

7.9.2 Different preambles in different packet types

ETSI TR 103 514 [i.26] defines the Long and Short preamble packet type as shown in Figure 11 and Figure 12 respectively.

The first part of the packet is the synchronization training field (STFS), which is 10/9 symbols for short packet format and 2 symbols for long format. It is proposed to reduce STFS field to 14/9 symbols, which is roughly 64,8 us. This can be seen very reasonable approach considering that in 802.11 system the STF field is 8 us, and the Primary Synchronization Signal (PSS) is single symbol having time of ~36 us with 30 kHz SCS in 3GPP NR. The PSS is transmitted with 5 ms or 20 ms repetition, but detection can be done as low as -6 dB SNIR level with 4,3 MHz TX BW. The BW is 2,5 times more than 1,728 MHz DECT channel bandwidth but as STFS is proposed to be longer than PSS it compensates, at least partly, for the reduced BW. Additionally, it is not considered that DECT-2020 device needs to be able to detect STFS fields as low levels as SNIR of -6 dB.

Thus packet synchronization with 14/9 symbols should be well within reasonable SNIR condition when STFS sequence is well designed. However, link simulations to confirm this argumentation would be desirable.

After STFS field there is CTF field for channel estimation so that HFS field can be detected reliably in long preamble packet format. In the current short format there is no CTF, but again in beacon packet format CTF field exists as shown in Figure 11.

After HFS there is N-1 symbols of channel estimation where number of N-1 is currently depending on number of MIMO streams uses for data transmission. (In the current short format this there is N symbols due to position of first CTF symbol).

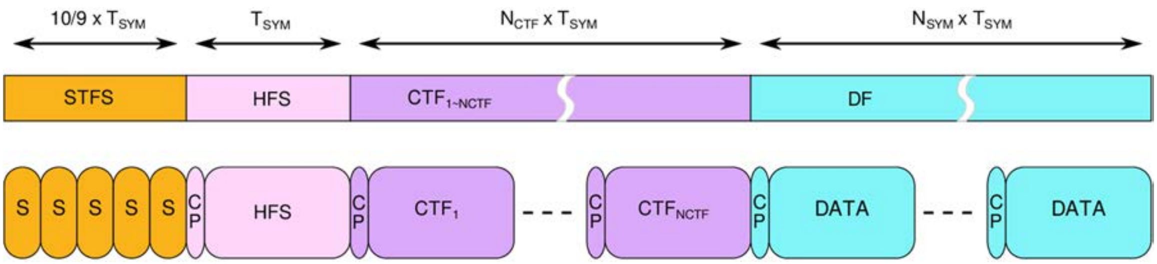


Figure 11: Short preamble packet type

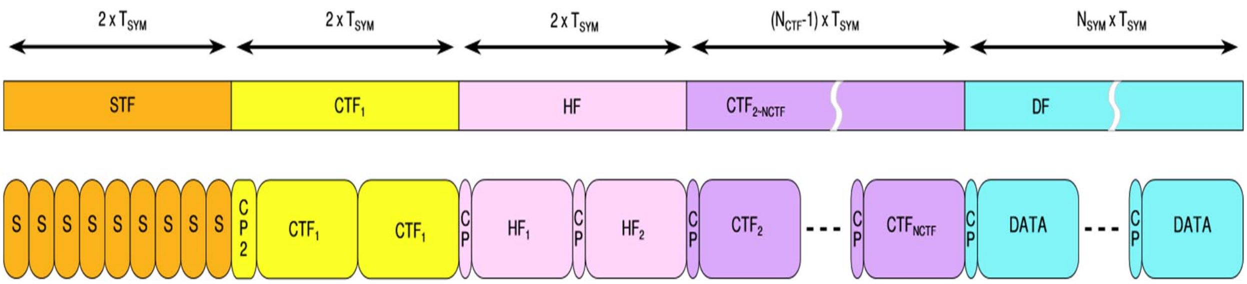


Figure 12: Long preamble packet type

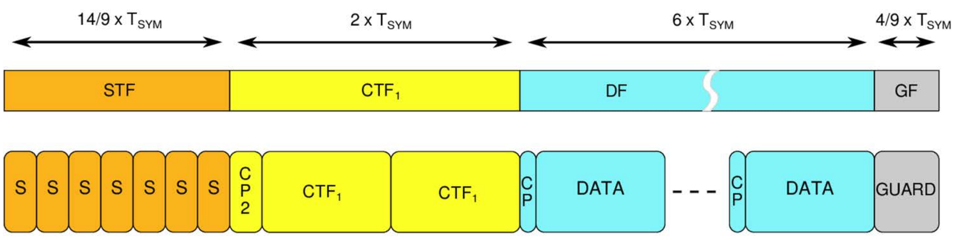


Figure 13: Beacon C/L downlink packet type

The problem of the current definition of short packet format is that it does not define any channel estimation symbol(s) before the HFS field. The format would require to first buffer the HFS samples and receive first CTF symbol before decoding HFS. Then information of number of CTF symbols would be only become available, as the remaining part of the CTF field will contain channel estimation symbol for all other MIMO streams.

The second problem of the packet formats is that if receiver needs to know advance, which format it could receive at different time moments, the packet format detection on the fly is rather difficult. Finally different formats simply introduce additional implementation burden with no clear use case. Rather it would be clearly beneficial if there would be a common PHY packet format for any type of transmission that any unsynchronized receiver would be able to receive without pre-knowledge of the used packet format.

If the STF length of the long preamble packet is reduced to 14/9 symbols such optimization is possible.

It is believed that design as shown in Figure 14 would actually support all unsynchronized packet transmission and remove need for defining long packet format, short packet format, beacon, random access or ULE packet formats separately. This proposal does not impact high efficiency packet format even tough clearly similarities between CTF/HFS symbols and RPF field could be considered.

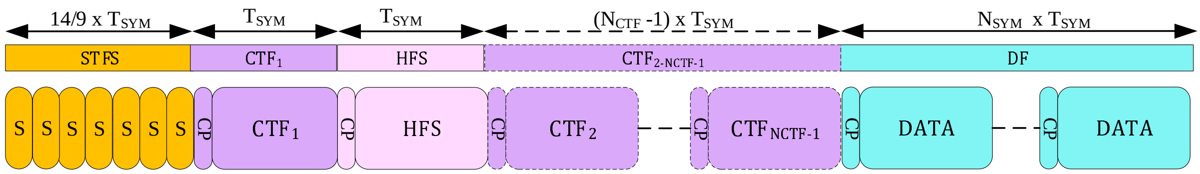


Figure 14: Proposed Generalized packet format

The modified general packet structure operates as follows:

- STFS field is a fixed sized.
- CTF_1 field is used for channel estimation for decoding HFS as well as data fields of the first antenna port of the single or multi stream transmission.
- HFS field is used to signal PHY layer parameters so that rest of the packet can be decoded.
- When transmission is single stream only the CTF_1 - CTF_{NTCF-1} symbols can be omitted as shown in Figure 15.
- CTF_2 - CTF_{NCTF-1} symbols are transmitted when 2 or more MIMO streams are transmitted.

The modification does not increase the overhead of the packet format compared to previous long packet format or beacon packet. Compared to previous short format the STFS field is slightly increased. However, this overhead increase can be compensated (or even further reduced) by optimizing CTF field as discussed in next clause (see clause 7.9.3).

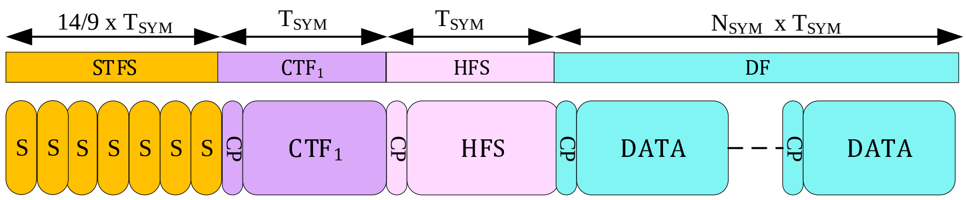


Figure 15: Generalized packet format for Single stream transmission

7.9.3 Optimizing usage of CTF field

In current packet formats the CTF symbol is included in all the subcarriers for channel estimation. This is same design as in IEEE 802.11g/n/ac [i.20]. However, in 802.11 the subcarrier spacing is 312,5 kHz which is more than ten times wider than DECT-2020. This means that the symbol is also more than ten times shorter than in DECT-2020 and carrying proportionally less energy. Due to larger CP, DECT-2020 is much more robust for delay spread resulting narrower coherence bandwidth than in 802.11 systems. However, studies like [i.29] indicates that in micro urban environments like railway station, the delay spread would be less than 0,1 us. When using common approximation of coherence bandwidth:

$$B_c = \frac{1}{2\pi\tau}$$

where τ is the RMS delay spread the 0,1 us corresponds 1 591 kHz, so even taken very robust approximation of 1,0 us delay spread, the coherence bandwidth is still 159 kHz, i.e. covering roughly 6 of DECT-2020 subcarriers.

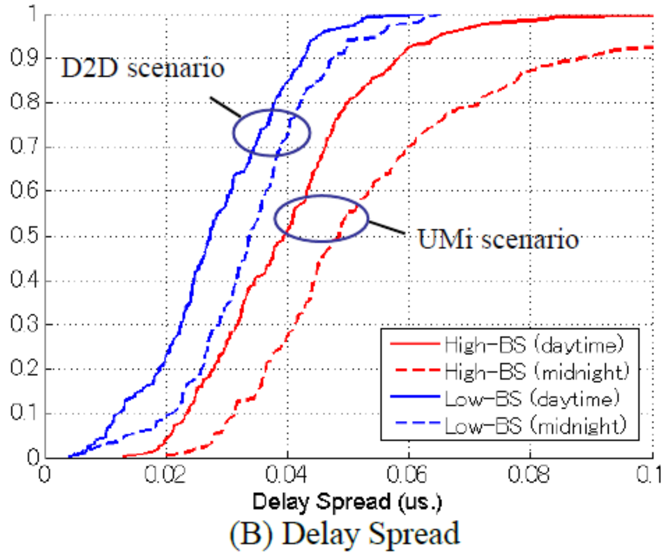


Figure 16: METIS: Measured Delay spread values from a square in front of Shibuya station in Tokyo

Thus, using CTF fully for channel estimation is not necessary to get accurate and sufficient channel estimation information, rather using less subcarriers could be considered. This is also the case in 3GPP NR master information block broadcast transmission, as well as in the case of UE dedicated data transmission. For the master information block transmission 3GPP NR uses 20 physical layer resource blocks (PRBs) which contains 12 subcarriers, resulting bandwidth of 3,6 MHz for 15 kHz subcarrier spacing and 7,2 MHz with 30 kHz subcarrier spacing. Every, fourth sub-carrier is used for channel estimation and 3 subcarriers are used for data transmission as shown in Figure 17.

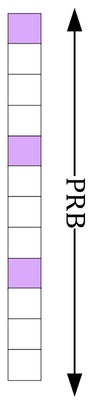


Figure 17: Single PRB (12 subcarrier) used for MIB transmission in NR

Applying same scheme for DECT-2020 CTF and HFS transmission, the HFS capacity from single channel would be increased from 52 subcarriers to 91 subcarriers, as total 52 subcarriers of CTF symbol 13 would be used for channel estimation purposes and 39 for data delivery. With BPSK (1/2 coding) this would result 45 bits and with QPSK (1/2) coding 91 bits of control or data information. Still the channel estimation would be able to track channel and the coherence bandwidth would be more than 108 kHz, which is still 3 time more than 802.11n and at same as 3GPP NR with 30 kHz subcarrier spacing for most robust data transmission. It should be noted that 30 kHz numerology will be used in all wide scale NR deployments in cities and urban areas on frequency bands between at 1,8 GHz and 4,4 GHz.

Similarly this approach can be used to reduce CTF overhead in MIMO transmissions, i.e. 4 MIMO streams could be divided into single symbol, and if number of streams is 2 or 3 the free part of the symbol could be filled with data symbols. This and above is illustrated in Figure 18.

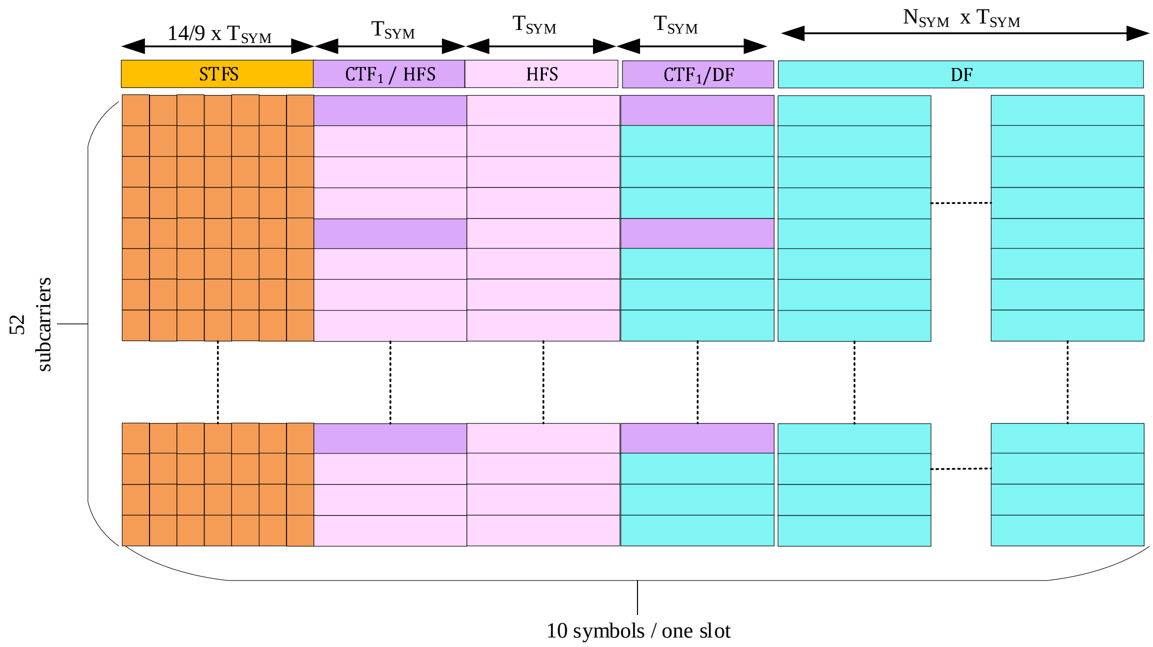


Figure 18: Illustration of generalized packet format with single stream MIMO transmission

The reception of two modes, BPSK (1/2) and QPSK (1/2) can be solved simply by requiring that receiver performs two decoding attempts, one with BPSK and other with QPSK and checking which CRC is passing. Alternatively, rules when either mode can be used, could be defined. However, coming up such rules and implementing those would be most likely more complex than simply performing two decoding. In LTE or NB-IoT or 3GPP NR, significantly higher amount of such blind decoding is mandated.

Finally, it is noted that the only reasonable manner to improve packet transmission efficiency of the DECT-2020 packet format is to reduce CTF overhead and multiplex Physical layer data or higher layer data in frequency domain with DECT-2020 numerology.

7.10 Possible ARQ/HARQ strategies in DECT-2020

7.10.1 General

In this clause the different possible strategies for ARQ/HARQ operated (at least in part) at MAC layer are analysed and proposals are given for the different types of traffic.

7.10.2 Possible ARQ mechanisms

7.10.2.1 General

By the phrase "operated (at least in part) at MAC layer" it is meant that the ARQ would be triggered by the CRCs and quality feedback given by the MAC layer.

The following options are possible:

- Traditional MAC ARQ.
- Hybrid ARQ.
- Retransmission or the content (that can also be defined as mixed MAC/DLC retransmission).

7.10.2.2 Traditional MAC ARQ

In this mechanism a retransmission query operated at MAC layer is established. To do that, all MAC packets should be labelled with a MAC packet numbering. This numbering may be as simple as modulo-2 (i.e. 1 bit), however additional flexibility is achieved if a wider range of numbering is implemented. See clauses 7.10.3 and 7.10.4.2.4 for discussion on the numbering range and properties.

When a packet is not received correctly, the quality feedback (or lack or quality feedback) triggers retransmission. Typically, the whole MAC packet is retransmitted. The retransmitted packet should be typically formatted with identical PHL construction (MCS and MIMO settings, etc.).

The retransmitted packet is identified by the packet number. The numbering range limits the transmission of new packets until a successful retransmission is made, or it is definitively waived. In many designs the transmitter may decide to waive the option of retransmit and "advance" to a new packet even is retransmission is still possible at MAC layer (this is the case in legacy DECT).

In case of multiple errors, a timer typically limits the number of retransmission attempts (in such a case the handling of the error is handed over to higher layers).

The case of "lack of feedback" (i.e. error in the reception of the feedback) can be handled in different ways. Some designs consider it as a "bad CRC". Others may allow certain flexibility to avoid blockings. Legacy DECT, in fact, allows certain flexibility.

The definition of "Traditional MAC ARQ" is that the packet should be correctly received in order for it to be valid.

This mechanism is already implemented in DECT with Mod-2 (i.e. 1 bit) packet numbering.

7.10.2.3 Hybrid ARQ

The hybrid ARQ operates in identical way to traditional MAC ARQ, but it uses the channel coding power and properties and so, in some case, it is able to reconstruct a packet from two incorrectly received retransmissions. Otherwise it is identical to MAC ARQ.

Since DECT-2020 inherently uses channel coding technology, the target is that all MAC ARQ will be Hybrid ARQ, unless this is precluded by certain specific situations. Therefore the DECT-2020 MAC ARQ is, by definition, HARQ except when this is not possible.

7.10.2.4 Retransmission of the content

This can be considered a mixed MAC/DLC retransmission. When a packed is incorrectly received, instead of retransmitting the whole packet at MAC layer, the Tx may choose to retransmit only certain types of traffic. For example, only retransmitting signalling traffic and not voice traffic.

This is not a MAC retransmission, it is mostly a DLC retransmission. From MAC perspective the packet number "advances". The retransmitted traffic segments are identified by their DLC headers and numbering.

This mechanism may allow different PHY packet construction (i.e. MCS and MIMO setting) and this is one of the reasons of its use.

No hybrid reconstruction is possible (unless coding is applied at DLC layer, what is not the case in current DECT-2020 assumptions).

An advantage of the mechanism is that PHY construction of the MAC packet (i.e. MCS and MIMO setting) may be changed (since there is no retransmission of the MAC packet, only the contents). This possibility may require additional features at DLC layer (i.e. re-segmentation of PDUs after reduction of MCS/MIMO).

The typical use case is retransmission of certain privileged channels (signalling) when main traffic does not require retransmission (i.e. voice traffic or time critical traffic).

This is a mixed MAC/DLC scheme. The difference with a pure DLC retransmission is that in the pure DLC case retransmission requests or packet acknowledgements that would trigger retransmission are DLC layer structures.

Legacy DECT already implements this possibility (with some singularities) for high layer signalling traffics (channels C_F and C_S). For U-plane traffic, legacy DECT is basically a pure-MAC + pure-DLC design.

7.10.3 Elements to be taken into account in ARQ/HARQ design

7.10.3.1 General

Based on literature, the design should paid special attention to the following critical elements:

- The size of the basic unit for retransmitting.
- The range of MAC numbering.
- The protection of the MAC numbering.
- The identities of the transmitter/receiver and their protection.

7.10.3.2 The size of the basic unit for retransmission

In DECT-2020 this decision applies specially to WLAN traffic (but can also be applicable to scheduled multi-bearer traffics). There are two basic possibilities: either to consider the "burst" as the retransmission unit (of variable size), or standardizing a fixed size retransmission unit, that in our case would be slot based (full of half slots).

7.10.3.3 The range of MAC numbering

Certain cases may be handled with a simple Mod-2 scheme (as in legacy DECT), but in multi-bearer transmissions this reduced range may limit the flexibility (or may simply not operate).

7.10.3.4 The protection of the MAC numbering

To get optimal operation and benefit from the hybrid ARQ, it would be desirable to transmit the MAC packet numbering in a protected manner and external to the segment under HARQ coding. In that way, the number can be observed in case of CRC failure and the Rx can attempt the proper recombination. However this is not always possible. In DECT-2020, this may be easily done for the initial WLAN packet (i.e. using the field/symbol A0), but this may not be possible in for WLAN continuation packets and in all packets for scheduled services (based in the current design). Therefore, either the basic design is changed or alternative methods of protection of the numbering should be envisioned.

However, it should be said that lack of protection of the numbering does not - per se - prevent the operation of the MAC ARQ. It only may prevent the "hybrid" recombination (what is still a limitation).

7.10.3.5 The identities of the transmitter/receiver and their protection

Certain traffic types, namely random access traffic, may require identification of transmitter/receiver to avoid confusion and recombination of traffics really sent to different nodes. In many cases, simplified identities may be used.

Ideally the identities should be transmitted in a protected manner and external to the segment under HARQ as discussed for the MAC numbering. However this is not always possible, and when possible, it may be unacceptably costly in terms of capacity (i.e. using the A0 field).

Not having identities protected may limit the cases of hybrid ARQ. However "regular" ARQ is normally possible if the identities have normal protection (i.e. are observable only after decoding the packet).

It may also cause certain cases of bad ARQ/HARQ association and combinations. However a good CRC mechanism at higher layers (e.g. a MIC at DLC) may be used to filter the cases of erroneous recombination, making the identity less critical (however it may still be needed for routing).

In scheduled services, the fact that transmission is always repeated in the slot at a given rate, makes less critical the need for continuous transmission of identities (in fact legacy DECT only sends the sender ID from time to time).

7.10.4 A possible design approach

7.10.4.1 General

Before entering in the different cases, the following general design decisions are taken:

1) The granularity for ARQ/HARQ will be at slot level. A long WLAN burst will be consider split into slots each one with their own CRC, MAC numbering:

   - Slots will be normally full slots, but half slots will also be supported.

Rationale: Although PHY layer may operate with arbitrary sized packets, the design of MAC mechanism requires some standardization. Also, a retransmission of a long packet would require finding a similarly sized free combination of slots what may be impossible or causing extra waiting times.

2) Each ARQ unit (slot) is reportable separately by the quality feedback mechanism.

3) A further CRC or MIC will exist inside the coded block. This mechanism may be implemented either at MAC or at DLC and will be capable of debugging residual cases of erroneous recombination.

The following two design decisions are compromises caused by current design. They are limitations that may reduce the efficiency of HARQ but have to be accepted by pragmatic reasons:

1) MAC packet number will be in the B field:

   - This means that the packet number can only be observed after decoding the packet. This is a limitation that will impact HARQ efficiency, however has to be accepted due to current PHY design.

2) Identities will be in the B field only be observed after decoding the packet, and not necessarily transmitted in each packet.

7.10.4.2 Solutions for scheduled traffic

7.10.4.2.1 General

The design of an HARQ for scheduled traffic is simpler than for random traffic types. Also, scheduled services are better candidates to profit from HARQ mechanisms. Therefore a solution for HARQ in scheduled services is proposed as first priority in the first release of DECT-2020.

7.10.4.2.2 Example scenario

To show all possibilities, a complex example scenario with asymmetric bearers, mixed bearer types and quality feedback roundtrip time different from 10 ms are proposed for the discussion.

Slots >	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24
carrier0																								
Carrier1																								
Carrier2																								
Carrier3																								
Carrier4																								
Carrier5																								
Carrier6																								
Carrier7																								
Carrier8																								
Carrier9																								

NOTE: See colour legend in clause 7.1.5.

Figure 19: Example slot/traffic allocation

7.10.4.2.3 Discussion on the scenario

The scenario shows asymmetric uplink scheduled traffic with 7 slots every 5 ms. It is assumed that only slot formats of 4,3 us CP are used (types 3 and 4).

The following slots and packet types have been allocated for the uplink traffic:

- Packet type 3, single carrier in slot positions 3, 8, 15 and 20.
- Packet type 4, single carrier in slot position 4 and 16.
- Packet type 3, dual carrier in slot positions 6 and 18.
- Packet type 4, dual carrier in slot position 7 and 19.

The quality feedback roundtrip time is set at 5 ms. The quality of slots 3, 4, 6, 7, 8 is reported in slot 13. The quality of slots 15, 16, 18, 19, 20 is reported in slot 1.

Packets use the O and C formats described in clause 10.3. Some slots may use MIMO. MCS and MIMO settings are considered static.

Only full slots are used in the figure, however the solution should be general and support the cases of half slots described in clause 10.3.

7.10.4.2.4 Simple solution proposed for scheduled traffic

A simple however effective solution for scheduled traffics is the use of a Mod-2 retransmission query bearer-by-bearer. This solution would be very similar to what is already implemented in legacy DECT and has the advantage of being able to support all cases of the figure. In addition to that, this solution is mostly tolerant to the "issue" of having the MAC numbering inside the B field, as it will be discussed below.

Therefore this is the solution proposed for the first release of DECT-2020.

Operation will be very similar to Ip_error_correct in legacy DECT. The main difference is that the roundtrip time of the quality feedback and ARQ mechanism will be configurable at setup. Otherwise it is near the same.

In the example the roundtrip time of the quality feedback and ARQ mechanism is set at 5 ms.

This means that, for example, slot 3 is reported in slot 13 and that in case of retransmission, it will be retransmitted in slot 15. The same applies to all other slots.

Each packet (either 3 or 4 format) will have a MAC header with the MAC sequence numbering. This number may be as small as one bit (Mod-2). Additional bits may be allowed to have some flexibility, but not so many, e.g. 2 or 3 bits are enough. Otherwise it will impact the HARQ capability.

The disadvantage of the mechanism is that it has a fixed allocation to the slotxcarrier. It is not possible to retransmit (at MAC layer) slot 3 in slot 20. This is the same case as in legacy DECT. However, retransmissions using a different slot are possible when advancing the slot at MAC layer and retransmitting the content at DLC layer. This is the same case as in legacy DECT.

A very important advantage of the schema is that, if Mod-2 is used, it allows the correct operation of the Hybrid ARQ even with the limitation of having the MAC sequence number embedded into the B field.

Discussion: After an erroneous reception and reporting, in the next received packet over the same slot (according to the configured ARQ roundtrip time) the receiver may find the following cases:

1) The packet is correctly received. Then the Rx may see if the packet is a retransmission of the previous one or an advance (named jump in legacy DECT terminology) and may route the packet accordingly. Therefore, this is the end of the problem.

2) If the new packet is also incorrectly received, the Rx may only need to recombine it with the previous incorrect packet and see if a correct packet can be built. The packet number should be the expected one at previous packet and internal CRC (or MIC) correctly found. Otherwise, it is a new error that has to be newly reported to the sender.

A pure Mod-2 scheme requires immediate retransmission at the first opportunity. If the packet is advanced and a new fresh one is sent, then there is no further possibility of retransmission. However, in turn, this makes the "hybrid" operation simple and feasible in all cases.

Using more than 1 bit for MAC packet numbering, allows flexibility and a variable time for retransmission may be allowed. However it makes the process more complex for the hybrid mechanism at Rx. This is due to the limitation of not having the sequence number externally visible.

Therefore, a pure Mod-2 with fixed slot-to-slot retransmission sequence is proposed as option for the first release of DECT-2020.

Advantages:

- Supports mixed slot formats (type 3 and 4 in the example).
- Supports multicarrier-slots (with the PHY layer optimized subcarrier mapping designed for this case).
- In theory it may support different MCS and MIMO settings slot by slot.
- Only one sequence number bit is needed (in theory).
- Fast.
- Simple basically it is the same as legacy DECT.

Disadvantages:

- Fixed inflexible scheme: retransmission has to be immediate. If this is not the case and a new packet is sent (advance) no retransmission is possible.
- Fixed slot operation. However slot crossing is still possible by advancing the slot and doing "retransmission of the content" on a different slot. However this cannot be Hybrid retransmission.

7.10.4.3 Solutions for packet-mode traffic

7.10.4.3.1 General

The ARQ/HARQ solution is, as general rule, more complex in packet-mode traffic. Also, not all cases of packet-mode traffic may profit from HARQ. A typical case is initial transmissions (either WLAN, ULE or RAC) where lack of quality feedback may be caused by an access collision and the correct response should be applying the collision handling algorithms and not performing ARQ (that may aggravate the access collision, if it has been the issue).

In addition to that, the decision about using an I-C-O approach or an L-S approach for burst formatting has strong influence in retransmission process and strategy. Both strategies have "issues".

The L-S approach has a fundamental issue for retransmissions. This is the need to find a free space in the slot-carrier grid in order to place a retransmitted L or S format "burst" of identical size to the initial transmission. This may not always be immediately possible.

The I-C-O allows independent re-transmission of slots, which adds flexibility for retransmission. However it has the issue of the coding of the MAC sequence number, which does not have easy solutions.

7.10.4.3.2 Additional considerations

Before applying ARQ/HARQ some considerations are needed to determine when ARQ/HARQ is the right strategy.

Considering the typical case of initial transmissions of either WLAN, ULE or RAC traffic (using either an L format or an I-C format) the following situations may happen:

- Complete failure or lack of quality feedback. This means that the other peer reports no reception at all (no A0 field) or is not reporting anything. In that case a collision may have happened. The correct response should be applying the collision handling algorithms and not performing ARQ (that may aggravate the access collision, if it has been the issue).

- Correct reception of A0 field and no reception of A1 and B field. This would be a usual case if A1 has been coded conservatively, but may happen if A1 is coded more aggressively. Retransmission is a debateable response. One possible response seems to be reducing the MCS (and also the MIMO setting) to a more conservative setting and retransmit the content. Another response is HARQ retransmission and relying on HARQ properties to recover the packet.

- Correct reception of A0, potentially correct reception of A1 and no reception of B field. This may be a very usual case. The case may have been caused by a random error, but may typically be caused by an excessively aggressive selection of the MCS and/or MIMO settings. One possible response seems to be resetting the MCS (and also the MIMO setting) to a more conservative setting and retransmit the content. Another response is HARQ retransmission and relying on HARQ properties to recover the packet.

As seen, ARQ over the first burst of a random transmission seems to be a debateable practice.

Retransmission would be correct in the following case:

- In the initial transmission, using I-C-O format, the A0 is correctly received, some of the B segments are also received correctly and only some B segments are reported as erroneously received.

In that case, attempting ARQ/HARQ on the badly received segments seems to be correct.

Note that this case is not detectable in L-S approach. This is one of the examples why I-C-O approach seems to be better for ARQ than L-S approach.

On subsequent transmissions (using either O-C format or S format), ARQ may have sense. The general rule is that some packets transmitted with the same MCS and MIMO should have been correctly received. Otherwise there may be a huge mismatch in MCS and/or MIMO settings compared with channel reality and is better reconfiguring these settings to a more conservative values before reattempting retransmission.

In any case O-C bursts have advantage over S packets due to the possibility of retransmitting only erroneous segments (slots).

7.10.4.3.3 Possible approaches for L-S approach

As discussed before, the L-S approach is not very friendly for ARQ/HARQ retransmissions that typically would never happen in an initial (L) packet (however they may happen in S packets). In addition to that, there is the constrain of the need to find a suitable space in the grid for retransmitting a potentially long burst. On the other hand the transportation of the MAC sequence number and the design of the mechanism may be simpler.

A MAC sequence number of several bits can be used. Values of 4-6 bits seems enough. This extended range will add flexibility to the ARQ/HARQ mechanism allowing not immediate retransmissions. This is important since the L-S format may need to wait until a suitable frame space for a long packet is found.

Since the L-S approach requires coding the burst length at the beginning and externally to the B packet, the same mechanism may be used for the sequence number. The options are:

- For L packets, using the A0 field.

- For S packet, using the same (not decided yet) mechanism used for the length indicator. This is not designed yet but options are:

  - Using some subcarriers in the CTF symbol.

  - Using some bits in the STF (5/9 T_{SYM} in current design). See discussion in clause 10.2.2.6, notes 3 and 4.

  - Adding a header symbol (as "old" design). (Not recommended due to impact in performance, but listed as last chance option).

  - Using some bits in the first B-symbol, protecting them, and encoding them separately to the rest of B bits.

  - Inside the B-field. This has the drawback of making difficult the hybrid combination that has to be done based on trial and error. However it may be possible in many cases.

The operation of the ARQ relatively simple and based on a quality feedback. A MAC numbering mod-2 requires continuous feedback, however the feedback would be very simple. Extended numbering range would allow non continuous feedback, however this has other problems in packet mode transmissions. Mod-2 is an option for this type of traffic using L-S formats.

7.10.4.3.4 Possible approaches for I-C-O approach

I-C-O is more friendly for ARQ/HARQ retransmissions that may even happen in the initial burst (see clause 7.10.4.3.2). The basic problem in the I-C-O approach is the transmission of the sequence number.

Two basic strategies are possible:

- Protecting some bits in the first B symbol and encoding them with great redundancy and separately to the rest of B bits.
- Inside the B-field. This has the drawback of making difficult the hybrid combination that has to be done based on trial and error. It may happen if the number of retransmissions is small. Otherwise it is not realistic. It is more problematic than with L-S format.

Nevertheless, it has to be noted that when the number of re-combinations is impossible to handle, the ARQ mechanism continues working. The only drawback is that the "hybrid" reconstruction cannot be processed. But retransmissions correctly received can be processed. It will not stop the operation. It is only a performance matter.

EXAMPLE: An O-C burst of packet numbers 2-3-4-5 is sent. The feedback indicates correct reception of packets 2, 3, 5 and bad reception of packet 4. The Tx retransmits a new OC burst with packets 6-4-7-8 (note the placement of packet 4). The receiver receives packets 6-bad-bad-8.

SOLUTION: The Rx can still try to reassemble the two badly received packets with the bad packet of previous burst. One of the re-assemblies provides a valid packet and the Rx can now see that sequence number (that was inside B field) is 4. So HARQ has worked and only the next packet need to be requested in next ARQ feedback.

A MAC sequence number of several bits can be used. Values of 6-8 bits seems enough. This extended range will add flexibility to the ARQ/HARQ mechanism allowing not immediate retransmissions.

The operation of the ARQ is as in previous case. I-C-O requires independent feedback per slot. NACK or GoBackN schemas can be used. Extended numbering range would allow non continuous feedback, however this has other problems in packet mode transmissions. Mod-2 is, in principle, not an option for I-C-O formats in packet service due to the length of the bursts.

8 Protocol Stack Architecture

8.1 Introduction

One approach for the protocol stack architecture for DECT-2020 would be to treat DECT-2020 only as a new PHL/MAC, and so keep the same design for DLC and NWK layers. Whilst this might minimize the effort required, since the upper layers have a large degree of reuse, it does reduce flexibility of the design as it imposes some design features that are not necessary as well as imposing other restrictions.

In the following clauses, two very similar concepts are introduced. These two concepts are loosely based on the architecture of LTE/LTE-NR, specifically with the introduction of a "convergence layer". Since it is assumed that data traffic will be heavily biased towards IP protocol, this layer is essential for header compression, packet numbering and handling ciphering and integrity. A further addition to the architecture is the "routing layer", which can provide efficient packet routing behavior for mesh-like networks.

8.2 Concept 1

8.2.1 Overall Protocol Stack

In Figure 20, the view on protocol stack for direct PP to FP communication is shown. Figure 21 depicts the protocol stack for relay, multi-hop PP to PP and i.e. general Mesh communication. Notable the protocol layers are the same for both options expecting that in multi-hop PP to PP and Mesh communication routing layer is needed. In direct PP-FP communication routing layer can be for e.g. considered to be in transparent mode or routing layer can be considered as a configuration option of the convergence layer. Additionally when protocol layer support Mesh network with PP to FP, FP to PP and multi-hop PP to PP communication there is no need to define separate operations with relays, self-backhauling or D2D operations as those are subcases of Mesh networking.

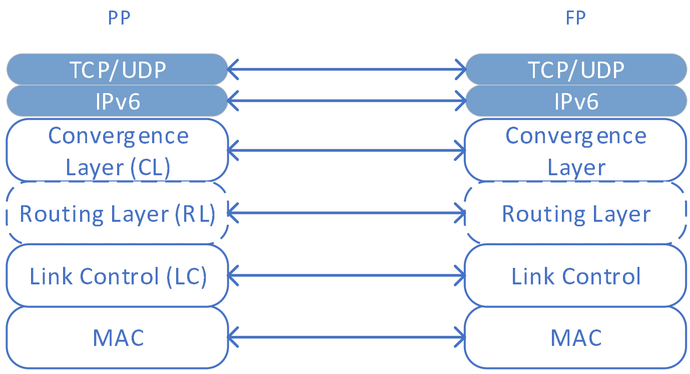


Figure 20: Protocol layer for PP to FP communication

As shown in Figure 21, it is proposed that in mesh communication the MAC and link control protocols are separate for each link, allowing medium access and link control functions operate independently at each link. This allows link data rate, channel access time and packet size selection etc. to be done optimum manner at each link and no convoluted decision taking account minimum capability of each hop is needed. This design also allows directly e.g. self-backhauled FPs implementations where link between two FP devices is higher throughput than radio links to a PPs. Additionally all packet re-transmissions are done between each link and re-transmissions are not propagating between multiple links.

Thus, the basic functionality of the MAC and Link Control functionality in mesh communication and PP to FP on these protocol layers is expected to be same.

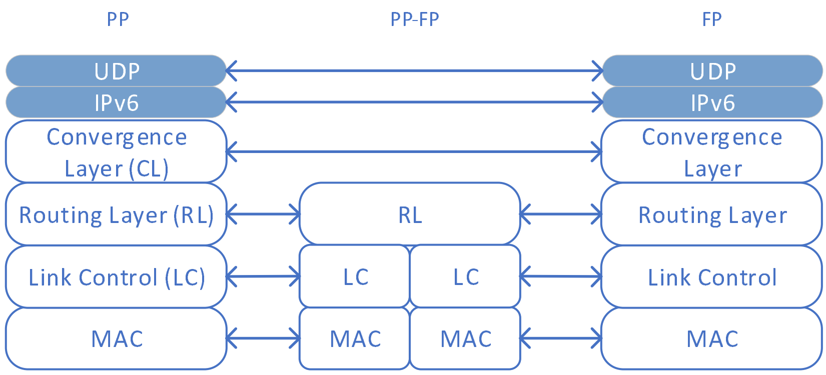


Figure 21: Protocol layer for relay, multi-hop and Mesh communication

The routing layer is then used in mesh communication to forward packets forward towards final destination. The routing layer packet contains both final destination and original source address. Finally convergence layer is used for necessary packet format conversion, header compression functions and provide port numbering for multiple native services from single device in case that application does not use UDP/TCP port numbering.

The current DECT utilizes security at MAC layer, however, LTE and NR utilize security in the PDCP, i.e. packet convergence layer. The benefit of convergence layer security is that security functions can be done base on application layer PDU and security procedures can be performed independent from actual transmission procedures, i.e. "offline". This reduces needed protocol functions that are performed just before transmitting a packet. Additionally this allows end to end security between node and FP in a mesh network and any node between these nodes cannot manipulate or read the actual data, adding extra security to the system. The drawback of this is that all link control and MAC control headers are send in plain text and without integrity protection.

However, the if ciphering and integrity protection is done in MAC layer the number fields send in plain text can be minimised and only fields that are needed in receiver for deciphering or does not need security at all are send in plain text. This is very suitable phenomena in mesh network as node receiving data from other node can check whether sender is valid and forwards only valid data forward. Such function makes DoS type of attack more difficult for mesh network as packets that do fail integrity will not be distributed any further in the system. Additionally performing security at MAC layer keeps final destination and original source address ciphered and not visible in the radio interface as plain text.

Thus protocol should be defined such a manner that security can be initiated at both layers when necessary.

8.2.2 Protocol Functions

8.2.2.1 General

Based on above discussion the following function to each protocol layer in DECT-2020 is proposed, the list is not necessarily complete and should be amendment when needed.

Additionally protocols should be re-designed completely without any legacy limitations from existing DECT standard. Finally the need for introducing a separate control signalling protocol - (RRC in LTE and NR) should be discussed separately as here mainly user plane functions are addressed.

8.2.2.2 Convergence layer

- Header compression.
- Sequence numbering.
- Duplicate detection.
- End to end Ciphering and integrity protection.

8.2.2.3 Routing layer

- Packet addressing with source and destination addresses.
- Packet routing.

8.2.2.4 Link control layer

- Sequence numbering.
- Segmentation of SDU to PDU to match Physical layer requirements.
- Acknowledge mode ARQ to recover lost PDU during radio transmission.
- SDU discard in the transmitter.
- Duplicate detection.
- SDU reassembly.

8.2.2.5 MAC

- Channel access, both contention-based and scheduled access should be supported for user plane data.
- Mapping data to/from physical layer.
- Concatenation and multiplexing of Link control data to physical layer packet.
- Priority handling.
- Fast feed acknowledgements and potentially HARQ.
- Link specific ciphering and integrity protection.
- Link specific packet addressing.
- Sending and receiving MAC control elements.

8.2.3 Layer 2 Protocol Details/Considerations for Mesh Operation

8.2.3.1 MAC functions and PDU format

As described in clause 8.2.2.5, the identified functions for MAC are:

- Channel access, both contention-based and scheduled access should be supported for user plane data.
- Mapping data to/from physical layer.
- Concatenation and multiplexing of Link control data to physical layer packet.
- Priority handling.
- Fast feed acknowledgements and potentially HARQ.
- Link specific ciphering and integrity protection.
- Link specific packet addressing.
- Sending and receiving MAC control elements.

Mesh network operation needs ciphering and integrity protection below routing layer so that any node receiving data from any other node can ensure that the sender is legitimate and forwards only valid data. Such function makes Denial of Service (DoS) type of attack more difficult for mesh network as packets that fail integrity will not be distributed any further in the system.

If MAC security is not always used (simple star network topology) as security is only done at convergence layer there is need to have separation between when MAC security is used and when not. Additionally, when transmission occurs in a common contentious resource, the MAC layer needs to indicate the receiver and sender address so that only the intended receiver processes the data forward. However, if transmission occurs in a resource that is configured explicitly for certain device the receiver and sender address could be omitted.

To achieve these objectives, it is considered that the first byte of the MAC packet is used to indicate what kind of common MAC header is used. This common MAC header is used for whether MAC security is used and whether common MAC header contains transmitter and receiver addresses. The first byte can be used to indicate also the option that no common header is used. These options are presented in Figure 22. The detailed coding of the first byte is FFS, and naturally this approach supports other common MAC header options also.

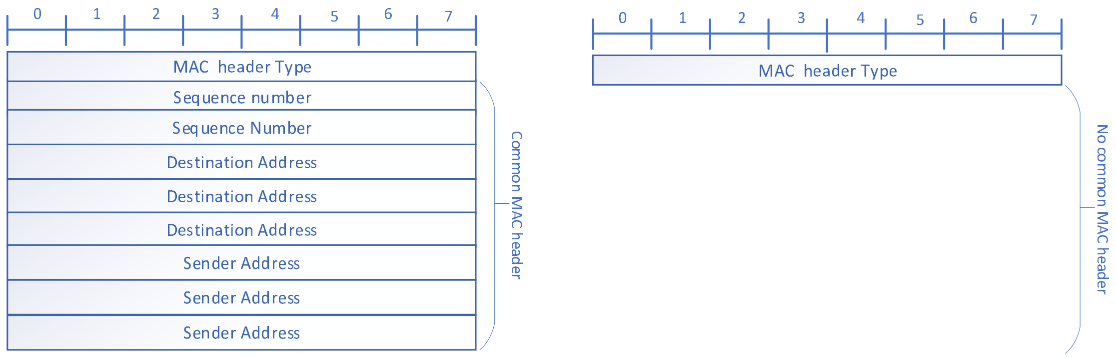


Figure 22: Common MAC header options

After the common MAC header, MAC multiplexing headers would be used to multiplex and concatenate different kind of user plane data as well as MAC control elements used for e.g. scheduling and link control features. The multiplexing can be achieved by introducing general MAC multiplexing or sub-header as shown in Figure 23.

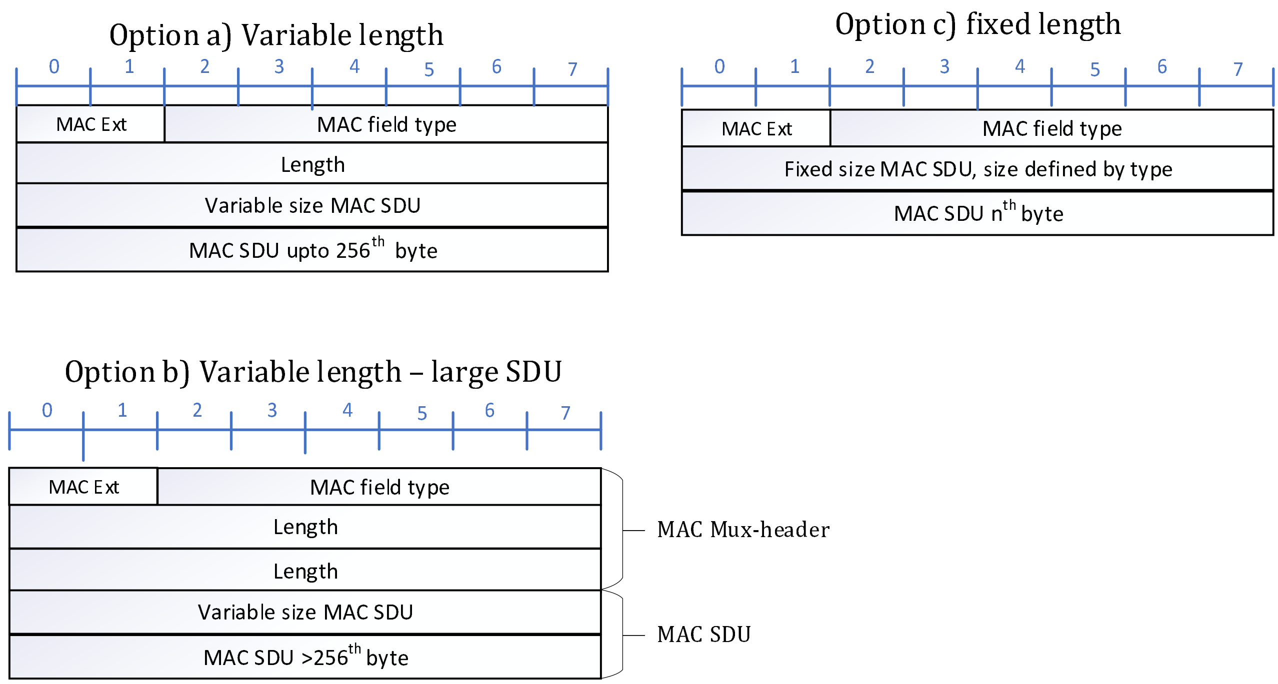


Figure 23: MAC multiplexing PDU with different header options

The Max Ext field is used to indicate whether 8 or 16 bit length or no length at all is used MAC SDU to indicate MAC SDU length. In case that length is not used the MAC field type not only defines what is the content of the MAC SDU but also defines fixed size of the SDU. This option can be useful in certain MAC control elements where size is fixed.

The overall MAC PDU is then pushed to the transmitter's PHY layer processing by including the common MAC header and different set of MAC multiplexing PDUs as shown in

Figure 24. Overall this process assumes that single MAC PDU is mapped to single Physical layer resource block and obtains the same PHY layer processing, e.g. channel coding. In case of MIMO transmission each MIMO stream would obtain its own MAC PDU, in this manner both MU-MIMO and SU-MIMO are similar from PDU processing point of view. Furthermore, when fast ARQ or HARQ is used the single MAC PDU is the unit that is acknowledged and in case of retransmission the MAC PDU is retransmitted.

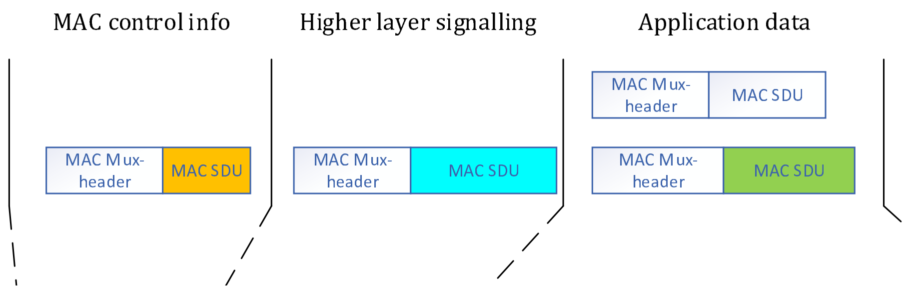


Figure 24: MAC PDU structure

The benefit of this approach is that transmitter can pre-process MAC Mux headers and corresponding SDU ready before actually knowing when it is able to transmit and how much data the transmission can carry. When that information and the needed common MAC header size is known the MAC PDU processing can start, as well as the MAC can start feeding the PHY processing. Even though the number of different flows and data types being multiplexed the transmitter only needs to manipulate the last MAC SDU by chopping the SDU to suitable length at DLC layer where segmentation is expected to happen. Thus, transmitter's DLC layer needs to be dimensioned for single MAC SDU chopping and not for chopping all MAC SDU based on the most demanding multiplexing scenario.

For including MAC multiplexing PDUs to a single MAC PDU it is expected that MAC should use a simple absolute priority scheme. Within the scheme, the transmitter first fills all data from flows having highest priority and when all data is included it starts including data from the next highest priority flow. For MAC control information the different rules could be defined and when these rules apply the MAC control info is the highest priority. The priorities are shown in Table 8. The number of different signalling and application data priorities should be further discussed.

Table 8: Priority handling

Traffic type	Priority
MAC Control info	1st (Included only based on rules)
Higher layer signalling	2nd
Application layer data prior 1	3rd
Application layer data prior 2	4th

8.2.3.2 Routing layer

It is expected that Link control layer will implement a typical sliding window ARQ protocol supporting segmentation. Additionally, configurations for unacknowledged and transparent mode are expected to be supported. It is expected that link layer should operate per link also in MESH networking. Thus, in this clause, a high level view of routing layer is presented. The routing layer should be used in MESH networking and could be considered transparent when simple star network operation is used.

The routing layer PDU header also contains set of control information used for routing and other purposes which are now left for future study. If desired, the number of these control information elements and length of each element can be made variable size by using the same header principles as in MAC layer.

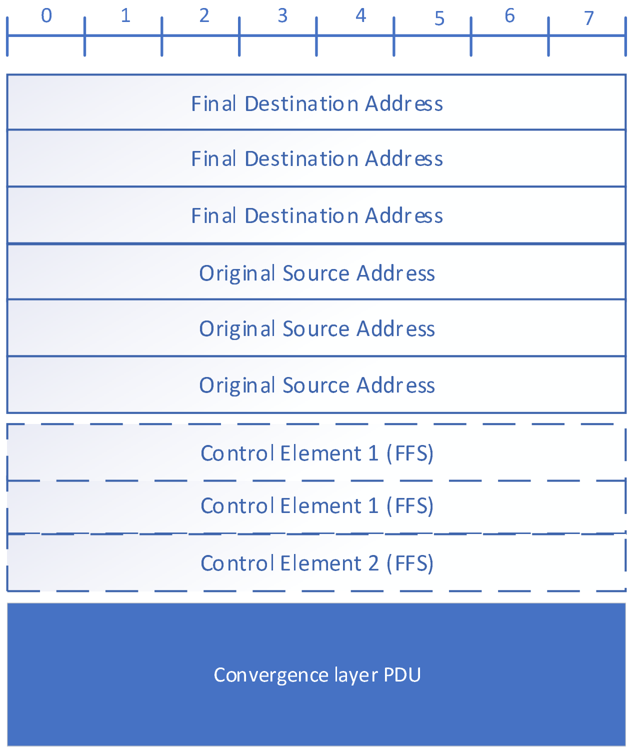


Figure 25: Routing layer PDU structure

8.3 Concept 2

8.3.1 Overview of Legacy DECT Protocol Architecture

For reference, the legacy DECT protocol stack follows the basic architecture as shown below. It consists of a Control Plane (C-Plane), which is used for signaling messages (e.g. setting up and maintaining links, etc.) and a User Plane (U-Plane) which is used to transport the user data once the connection has been established.

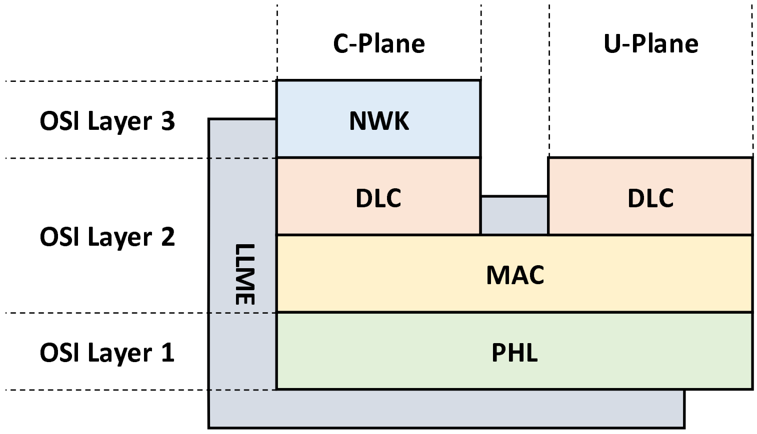


Figure 26: Legacy protocol architecture

The protocol stack consist of 4 layers, which map on to the OSI approximately as shown in Figure 26.

8.3.2 Overview of DECT-2020 Protocol Architecture

The working assumption has been that the changes required for DECT-2020 are mainly in PHL and MAC, with only minor changes to the higher layers. However, a study of security requirements and other modern protocol stacks such as LTE/LTE-NR suggest that some modifications are desirable. In particular, a modified structure has been propose, adding 2 new sub-layers to the OSI Layer 2 (between DLC and NWK). This is depicted in Figure 27.

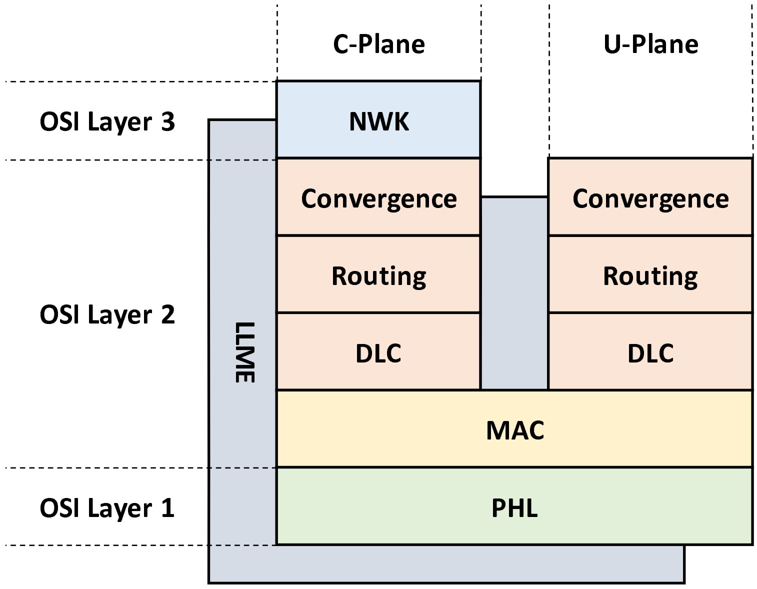


Figure 27: Proposal for DECT-2020 protocol architecture

The NWK, DLC, MAC and PHL have similar roles as before, but of course modified according to the new radio. The two new sub-layers are the "Convergence Layer" and the "Routing Layer".

Convergence Layer:

The Convergence Layer (CL) is analogous to the Packet Data Convergence Protocol (PDCP) of UMTS, 3GPP, LTE, etc. This layer provides sequence numbering of higher layer SDUs, as well as end-to-end ciphering and integrity (when required).

NOTE 1: Having security and integrity in this layer (rather than in MAC layer as in legacy DECT) has a number of advantages, e.g. better suited to repeaters/mesh systems, less real-time critical, integrity checks can be performed on bigger SDU sizes (more efficient), etc.

NOTE 2: LTE assumes all data on U-Plane is in the form of IP packets (IPv4 or IPv6), and in LTE the PDCP also provides header compression for IPv4 and IPv6 using Robust Header Compression (ROHC) which replaces the IP header (20-40 bytes) with a much smaller "token" (2-3 bytes). However, this case is not clear for DECT-2020, and it might be preferable to move such header compression to IWU layer, which would allow any format of data to be transferred (e.g. "raw" voice codec data, IP packets, proprietary formats, etc.).

NOTE 3: Having the security (ciphering and integrity) at the level above DLC, is similar to how it is done in DECT ULE.

The security topic is covered in more detail by another study (see [i.28]).

Routing Layer:

The Routing Layer (RL) was introduced to provide a packet routing function. This feature is potentially useful for creating "repeater" type of systems, as well as "mesh" networks. The RL provides unique source and destination address which can be used to route packets between nodes, assuming the availability of some form of "routing table". Locating the "Routing" layer below the "Convergence" layer means that there is no need to de-crypt/re-encrypt data prior to routing, since the "Convergence" layer provides end-to-end ciphering.

NOTE 4: In the trivial case, the routing is transparent (i.e. simply routed up/down the stack).

NOTE 5: Overhead of extra identities needs to be understood and mitigated if possible.

NOTE 6: Further study is required.

8.3.3 Inter-working with Legacy DECT

A DECT-2020 FP will be required to simultaneously support both DECT-2020 PPs and legacy DECT PPs. Since the radios of legacy DECT and DECT-2020 are vastly different, this is best accomplished by having totally separate PHL and MAC to handle the legacy DECT devices. Furthermore, with the newly proposed protocol stack architecture it would seem more appropriate to keep the lower layers separate by either: a) having totally separate legacy DECT and DECT-2020 protocol stacks and providing the required connectivity with an inter-working layer; or b) having common NWK layer and separation below (i.e. OSI Layer 1 & 2).

Figure 28 shows the option with totally separate protocol stacks, with legacy DECT (on the left) and a DECT-2020 (on the right).

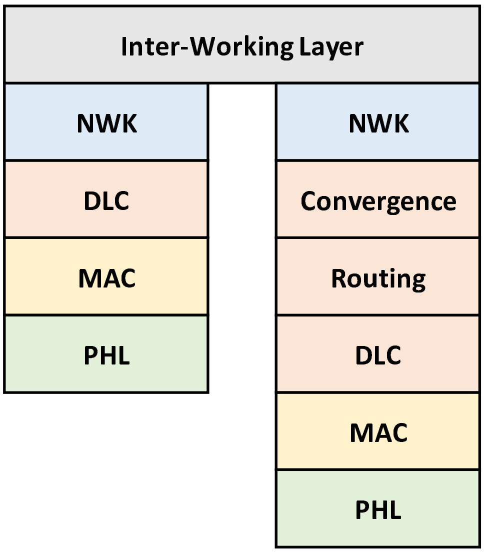


Figure 28: Inter-working of legacy DECT and DECT-2020 (option 1)

Figure 29 shows the option with common NWK layer and separate OSI Layer 1 & 2, with legacy DECT (on the left) and a DECT-2020 (on the right).

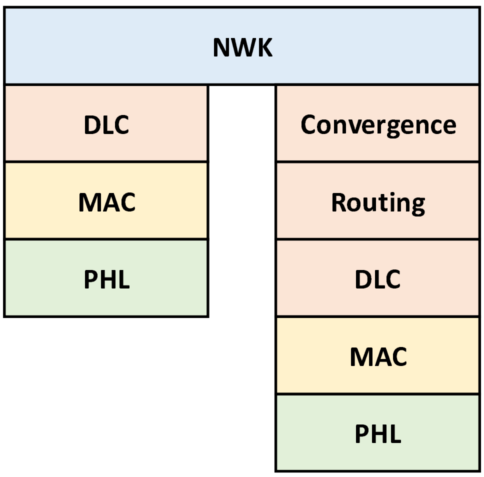


Figure 29: Inter-working of legacy DECT and DECT-2020 (option 2)

In both cases, the following features need to be provided for inter-working:

- Synchronization of time-frame (slot & frame).
- Shared knowledge of used/available slots for transmission.
- Ability to connect calls between legacy DECT and DECT-2020 handsets, e.g. intercom/conference, case.

In the case of totally separate protocol stacks there also needs to be a shared subscriber database, or at least some way to ensure that subscriber numbers and identities are unique between the legacy DECT and DECT-2020 sub-systems.

NOTE 1: It is assumed that the PP is either a legacy DECT PP or a DECT-2020 PP, and so it does not have to (and indeed, is not able to) roam or handover between the coverage area provided by the legacy DECT component and the DECT-2020 component.

NOTE 2: Using completely separate protocol stacks gives more freedom to the design of the DECT-2020 protocol stack, e.g. not having to retain support for un-used services, or redundant features in the implementation.

8.3.4 Inter-working with 3GPP-5G

Non-3GPP access networks can be connected to 5G core network via a Non-3GPP Inter-Working Function (N3IWF). The N3IWF interfaces to 5G core network control-plane functions and user-plane functions via N2 interface and N3 interface, respectively.

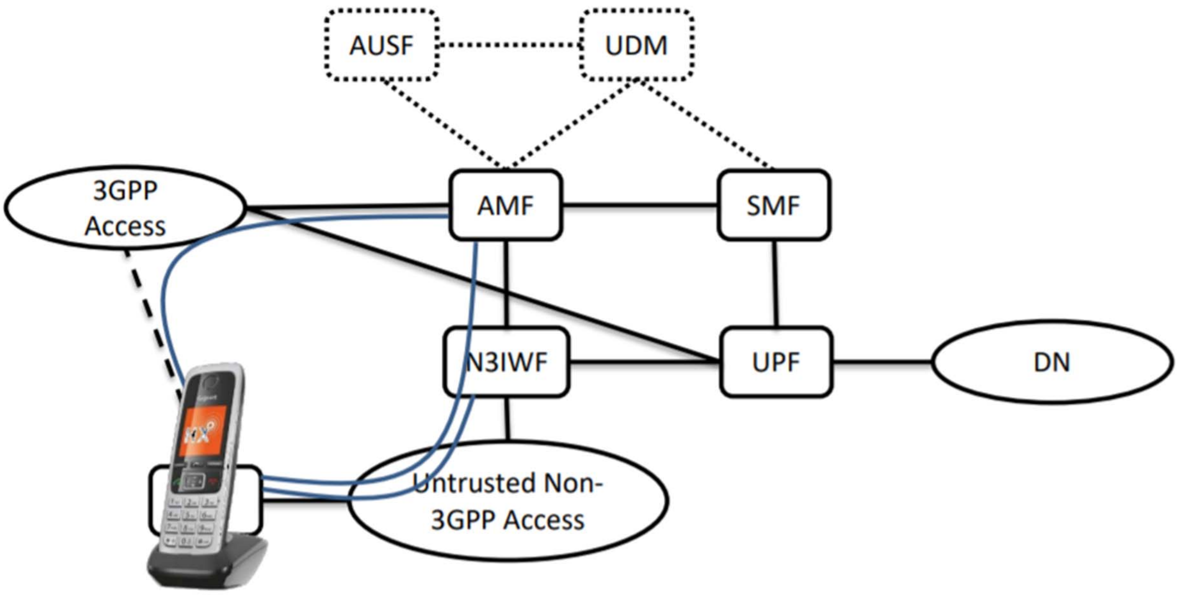


Figure 30: Interworking of DECT and 3GPP-5G

The DECT will require the concept of SIM/eSIM in order to access the 3GPP-5G network, but this can be situated in the FP, which will then act as a proxy for PPs.

NOTE: WiFi proposes to use same technique in order to access 5G network.

It is also possible to interface with 3GPP-5G as a trusted network, but this is considerably more complex. There is a separate study on the interworking with 3GPP-5G and so the above information is only included for basic information.

8.3.5 Detailed Protocol Architecture

8.3.5.1 General

Logical Channels:

The majority of communication through the stack takes place over logical channels. These group similar types of data together (e.g. paging data might be sent from the higher layers via the paging channel).

Service Access Points (SAPs):

The logical channels enter and exit the protocol layers at defined Service Access Points.

Control-Plane and User-Plane:

Conceptually there is a split of control messages and user data messages, through the C-Plane and U-Plane respectively.

NOTE: Legacy DECT has names for the logical channels and SAPs. These may, or may not, be adopted for DECT-2020.

8.3.5.2 NWK Layer

Figure 31 shows a simplified reference model for the NWK layer, which exists only as a C-Plane entity.

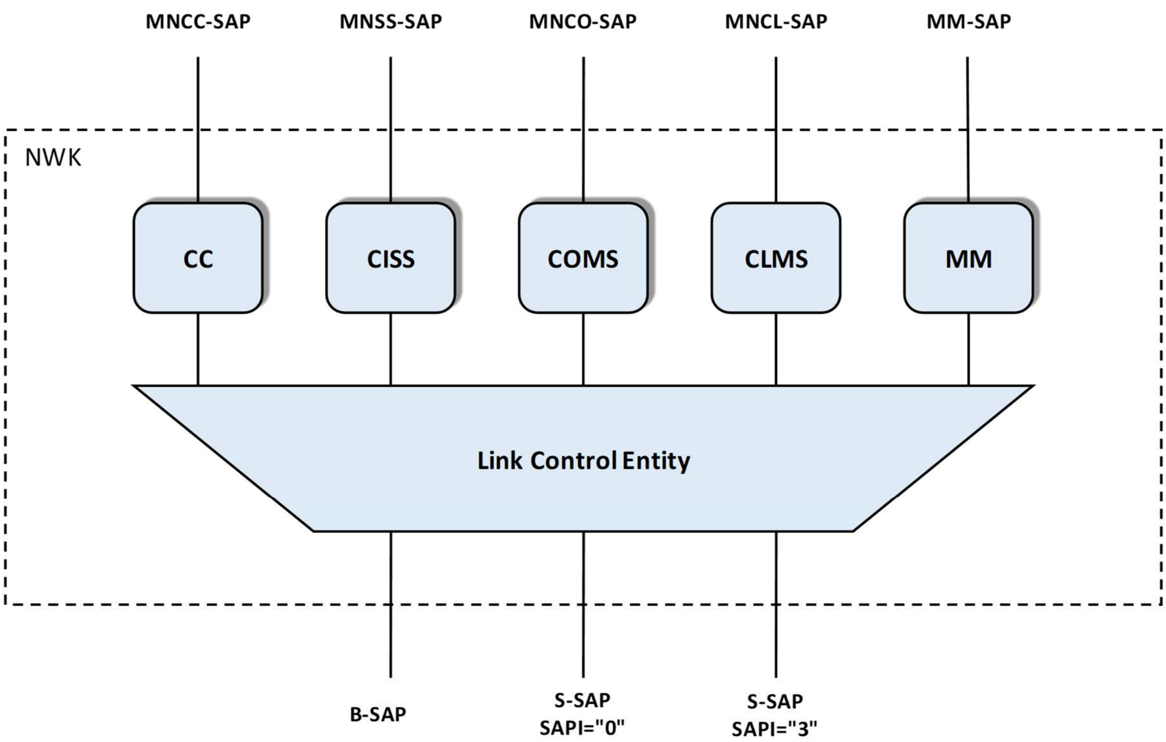


Figure 31: NWK layer

Call Control (CC):

State machine and protocol for Call Control services. Separate instances of the CC exist per subscriber. Multiple instances of a CC can also exist per subscriber, with each instance being allocated a different Transaction Identifier (TI).

CC is used for establishment and handling of voice calls, data connections, etc.

NOTE 1: This is typically a circuit-switched service. Although the CC may also be suspended/resumed for intermittent services. DECT ULE used a permanent virtual connection to avoid call control setup for each packet, or use of suspend/resume protocol which was deemed to be inefficient and/or too complex. For DECT-2020, this schema should be considered in order to accommodate packet data transfers more efficiently (typical of IP based systems), i.e. need an efficient/low-latency suspend/resume mechanism.

Call Independent Supplementary Services (CISS):

State machine and protocol for Call Independent Supplementary Services. Separate instances of the CISS exist per subscriber. Multiple instances of a CISS can also exist per subscriber, with each instance being allocated a different Transaction Identifier (TI).

CISS is used for NG-DECT ([i.31], [i.32], [i.33]).

NOTE 2: It is possible that multiple instances (per subscriber) are not required, and a single instance will do.

Connection Oriented Message Service (COMS):

State machine and protocol for Connection Oriented Message Services. Separate instances of the COMS exist per subscriber. Multiple instances of a COMS can also exist per subscriber, with each instance being allocated a different Transaction Identifier (TI).

COMS is used for some proprietary features.

NOTE 3: It is possible that multiple instances (per subscriber) are not required, and a single instance will do.

Connectionless Message Service (CLMS):

State machine and protocol for Connection Oriented Message Services. Only 1 instance of CLMS entity exists.

CLMS is intended for point-to-multipoint broadcast of system data. It is used by NG-DECT (currently for "Base station name" broadcast during registration procedure).

NOTE 4: CLMS broadcast exists as either CLMS-FIXED (which uses A-field P_T channel) or CLMS-VARIABLE (which uses connection-less bearer procedures). In general, only CLMS-FIXED is widely implemented. For DECT-2020 the low-rate/high-latency CLMS-FIXED service is probably undesirable, and CLMS-VARIABLE should be investigated.

NOTE 5: If a connection exists then CLMS-VARIABLE may be sent directly over the connection rather than via a separate connectionless bearer.

Mobility Management (MM):

State machine and protocol for Mobility Management services. Separate instances of the MM exist per subscriber. Only two instances of MM can exist per subscriber, and these have to be initiated in opposite directions (i.e. maximum of one FT-initiated instance and one PT-initiated instance), with each instance being allocated a different Transaction Identifier (TI).

MM is responsible for a host of services, including: identity procedures; authentication procedures; location procedures; access rights procedures; key allocation procedure; parameter retrieval procedures; ciphering related procedures.

An important aspect for DECT-2020 will be the authentication and ciphering state.

Link Control Entity (LCE):

The LCE is the lowest entity in the NWK layer. It performs the supervision of lower layer link states for every data link end-point, including the generation of page messages, upward routing of messages based on Transaction Identifier (TI) and Protocol Discriminator (PD), downward routing of messages to different data link end-points, queuing of messages, reporting link failures to higher NWK layer instances, etc.

Service Access Points (SAPs):

MNCC-SAP: For CC protocol messages.

MNSS-SAP: For CISS protocol messages.

MNCO-SAP: For COMS protocol messages.

MNCL-SAP: For CLMS protocol messages.

MM-SAP: For MM protocol messages.

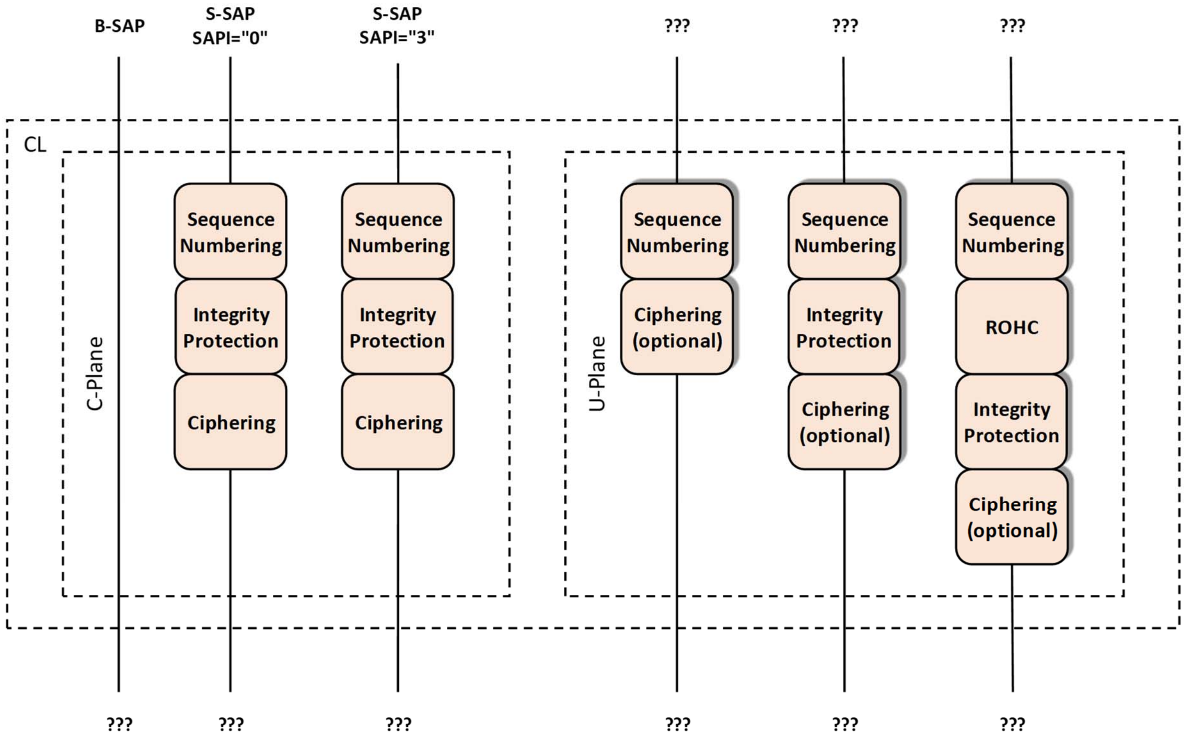
S-SAP (SAPI = "0"): For connection-oriented messages.

S-SAP (SAPI = "3"): For connection-less messages.

B-SAP: For system broadcast/paging messages.

8.3.5.3 Convergence Layer

Figure 32 shows a simplified reference model for the Convergence Layer.



NOTE: ??? means "SAP to be defined" (an interface still to be defined).

Figure 32: Convergence layer

C-Plane:

The broadcast/paging logical channel is passed through the CL transparently. The other logical channels of the C-Plane have the following functions: Sequence numbering; integrity protection; ciphering.

U-Plane:

Several logical channels should be provided, offering a variety of service types. An FP-PP connection may support multiple U-Plane connections at a time, possibly providing different QoS; these maybe mapped on to the same or different physical channels by the lower layers.

NOTE 1: Robust Header Compression (ROHC) is shown here for one of the channels transporting IP data. However, as discussed earlier this could also be handled as IWU function.

NOTE 2: The total range of U-plane service types is TBD, and needs further study. However, it is anticipated to be considerably less than the 15 or so that are currently defined by the DECT base standard.

Sequence Numbering:

A header will be added to each SDU with an N-bit sequence number. This sequence number allows for in-order delivery of SDUs (which might be optional for some services), as well as discard of duplicate SDUs. The sequence number is also one of the inputs to the ciphering process.

NOTE 3: The value of "N" is TBD, and depends on the window size. Suggested values 8, 12, 16 bits. The header format can be left flexible to allow for a range of values and possible future modification.

NOTE 4: A service with no sequence numbering may be needed (this might also mean no ciphering)

Integrity Protection:

A process such as AES CCM should be used to provide a Message Integrity Check (MIC) (synonymous with Message Authentication Code (MAC)) which is appended to the SDU.

NOTE 5: LTE seems to only perform integrity check on C-Plane messages (rationale behind this is unclear).

Ciphering:

A process such as AES CCM should be used to provide ciphering of the SDU.

NOTE 6: The MIC is part of the ciphering function.

NOTE 7: Some U-plane services may not require ciphering, e.g. because they are ciphered end-to-end by the application, or perhaps for low latency services.

Service Access Points (SAPs):

S-SAP (SAPI = "0"): For connection-oriented messages.

S-SAP (SAPI = "3"): For connection-less messages.

B-SAP: For system broadcast/paging messages.

8.3.5.4 Routing Layer

The Routing Layer requires further study, particularly with respect to the operation of the routing feature itself. There are a number of open points, e.g. how are connections to routable nodes maintained; how does paging work, etc.

In the meantime, for the default transparent operation (i.e. all packets pass straight through without any re-routing) it is assumed that the logical channels are connected 1:1 and there is a small header added to the SDU (or possibly it can be a single bit in existing header). A full header containing additional source and destination addresses is only added when active routing is actually required.

8.3.5.5 DLC Layer

Figure 33 shows a simplified reference model for the DLC Layer.

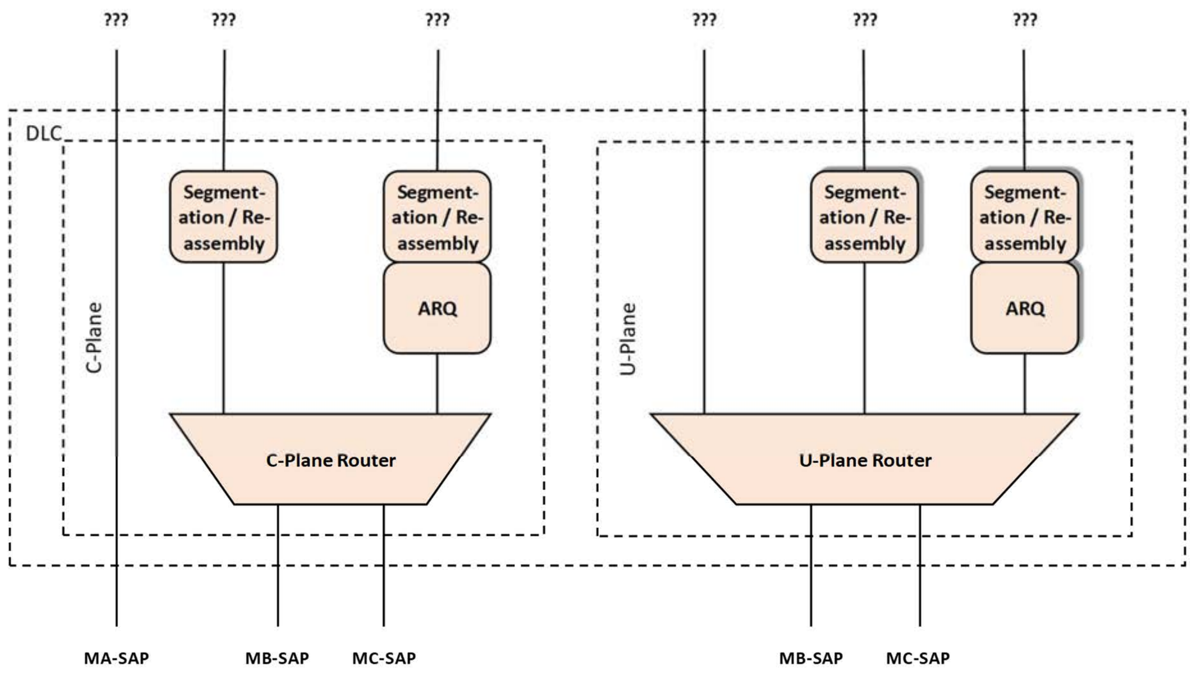


Figure 33: DLC layer

C-Plane:

The broadcast/paging logical channel is passed through the CL transparently to the MA-SAP of the MAC layer. The other logical channels of the C-Plane have the following functions: Segmentation of one or more SDUs into suitably sized PDUs/re-assembly of one or more PDUs into one or more SDUs; Automatic Repeat Request (ARQ) mechanism for missing PDUs.

NOTE 1: Connection-less messages do not use ARQ since there is no back-channel to report packet errors. So, the connection-less channel is basically an "unacknowledged" channel.

C-Plane Router:

Each C-plane link may be routed to any suitable MAC connection.

U-Plane:

The U-plane is handled similar to the C-Plane, with the following functions: segmentation of one or more SDUs into suitably sized PDUs/re-assembly of one or more PDUs into one or more SDUs; Automatic Repeat Request (ARQ) mechanism for missing PDUs. U-Plane DLC can also provide Transparent Mode (i.e. no segmentation, no ARQ), Unacknowledged Mode (i.e. with segmentation/re-assembly, no ARQ) and Acknowledged Mode (i.e. with segmentation/re-assembly, ARQ).

NOTE 2: Although very similar to C-Plane, parameters such as PDU size, window size, sequence numbering, and ARQ mechanism may differ.

U-Plane Router:

Each U-plane link may be routed to any suitable MAC connection.

Segmentation:

On the transmission side, one or more SDUs are segmented (and/or concatenated) into one or more "suitably sized" PDUs. Each PDU has header with sequence number (this is different to the SDU sequence number) and other control information.

NOTE 3: What is a "suitable size"? Unlike legacy DECT, the capacity of the physical bearer is not fixed (due to possibility of varying MCS, and also because multiple logical channels can be sent over same physical channel). As such, the MAC may need to re-segment the PDUs further. This means that the optimal PDU size is not hugely important, and is more down to granularity of messages/trade-off with efficiency vs ARQ latency (i.e. small PDUs → lower latency/higher overhead; large PDUs → higher latency/lower overhead). Alternatively, segmentation into PDUs of exact size is left until last minute when exact size of MAC transmit buffer is known!

NOTE 4: LTE also performs concatenation of SDUs as well as segmentation. However, in LTE-NR, they have removed this possibility, and rely on MAC concatenation for better management of Quality of Service (QoS) and latency. The option to concatenate in DLC or not is still an option for DECT-2020.

Re-assembly:

On the receiver side, one or more in-sequence PDUs are re-assembled into one or more SDUs. Missing PDUs can be requested using the ARQ mechanism. Duplicate PDUs can be discarded.

Automatic Repeat Request (ARQ):

Missing PDUs can be requested from the peer entity using an ARQ mechanism.

NOTE: Details of ARQ mechanism are TBD, but should be something like LU10 in legacy DECT.

Service Access Points (SAPs):

MA-SAP: For system broadcast/paging messages.

MB-SAP: For connection-less messages/data.

MC-SAP: For connection-orientated messages/data.

8.3.5.6 MAC Layer

Figure 34 shows a simplified reference model for the MAC Layer.

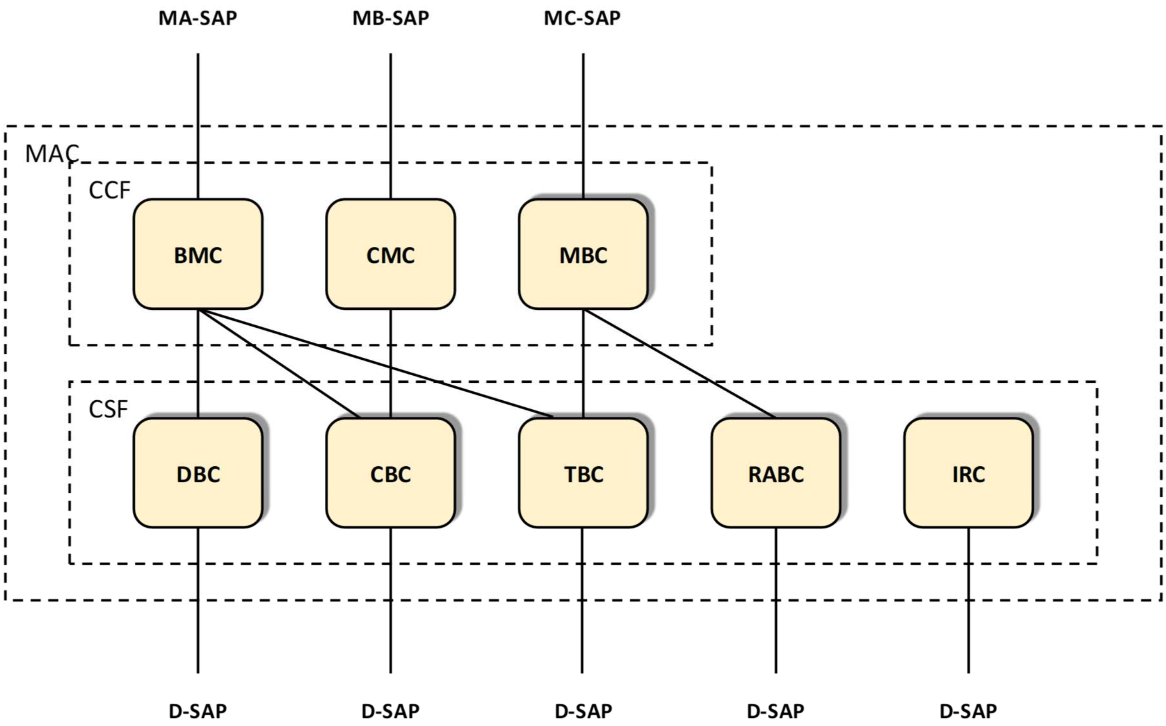


Figure 34: MAC layer

Cell Cluster Functions/Cell Site Functions:

Legacy DECT has a structural split of the MAC into Cell Cluster Functions (CCF) and Cell Site Functions (CSF). This split facilitates multi-cell systems, such as for a PBX, since each CSF instance has its own RF part which can be remote from the rest of the system.

NOTE 1: For DECT-2020, it is an open topic if such architecture has to be supported.

Even without a physical split into CCF and CSF, the logical split does serve some purpose since it allows a hierarchy of control, e.g. the Multi-Bearer Controller (MBC) can control multiple Traffic Bearers, etc.

Broadcast Message Controller (BMC):

The BMC provides a set of continuous point-to-multipoint connectionless services. These are used to carry internal logical channels, and are also offered to the higher layers via the MA-SAP. These services operate in the direction FT to PT, and are available to all PTs within range.

The BMC provides its services on one or more dedicated physical bearers (managed by the DBC), and it can also be carried by the connection-orientated and connection-less physical bearers (managed by the TBC and CBC).

Connection-less Message Controller (CMC):

The CMC provides connection-less point-to-point or point-to-multipoint services to the higher layers via the MB-SAP.

These services may operate in both directions between one specific FT and one or more PTs.

The CMC provides its services on one or more dedicated physical bearers (managed by the CBC).

Multi-Bearer Controller (MBC):

Each instance of MBC provides one of a set of connection oriented point-to-point services to the higher layers via the MC-SAP. These services may operate in both directions or in one direction between one specific FT and one specific. PT. Each service instance provides a connection (a connection oriented service) between one FT and one PT.

The MBC provides its services on one or more dedicated physical bearers (managed by the TBC).

Beacon Bearer Control (DBC):

The DBC provides the management and control of one physical channel for the BMC.

NOTE 2: For now, the term "Dummy Bearer Control" (DBC) is retained since it is so commonly used in DECT. However, in DECT-2020, this is not a "dummy" in shape or form. A better term might be to call it "Beacon Bearer Controller" (BBC).

Connection-less Bearer Control (CBC):

The CBC provides the management and control of one physical channel for the CMC.

NOTE 3: As well as carrying the connection-less service as required by the higher layers and the CMC, the CBC may also carry the services of the BMC.

Traffic Bearer Control (TBC):

The TBC provides the management and control of one physical channel for the MBC. The TBC also handles the Hybrid Automatic Repeat Request (HARQ) process for the physical channel.

NOTE 4: As well as carrying the connection-orientated service as required by the higher layers and the MBC, the TBC may also carry the services of the BMC.

Random-Access Bearer Control (RABC):

The RABC handles the initial setup/grant using contention-based channel access mechanism.

NOTE 5: This could be considered as a specialized feature of a TBC, but there are some important differences so maybe it is better to have a separate entity.

NOTE 6: LTE has entity called RACH (Random Access Channel) which handles its initial random access procedure. This only seems to exist on UE side (I think because the request is sent on random access channel, but the response is sent on downlink shared channel).

NOTE 7: As well as being used for initial setup/grant, it is also anticipated that this mechanism is used to transfer small amounts of data, e.g. ULE or similar.

NOTE 8: Should HARQ be used on this connection? Seems like it might not be suitable since this is potentially a contended channel.

Idle Receiver Control (IRC):

The IRC handles various scanning, search, synchronizations functions that control the receiver when not involved with a bearer.

Multiplexing:

The DBC/CBC/TBC entities have the task of multiplexing their various logical channels onto the limited resources of the physical cannel. The multiplexing takes into account the capacity of the physical channel and the priority of the data (for QoS purposes).

Hybrid Automatic Repeat Request (HARQ):

HARQ is combination of Forward Error Correction (FEC) which is the channel coding scheme of the PHL, with Error Detection (ED) and fast reporting by MAC layer back to the sending side. The sending side can then resend the same data or different data depending on HARQ type and mode.

HARQ can only be performed for connection-orientated physical channels (i.e. there needs to be a back-channel in order to send the ACK/NACK).

Various types of HARQ/modes could be supported:

- Type I HARQ: data + FEC + ED bits sent each transmission, and same data is resent on an NACK.
- Type II HARQ: data + ED bits are sent, and FEC bits are only sent in case of a NACK.
- HARQ with "soft-combining": In case a packet and its re-transmission are both received with errors, it might be possible to reconstruct a good packet by using technique such as maximal-ration combining.

NOTE 9: At least Type I HARQ should be supported, with possibility to support other techniques.

NOTE 10: It will not be possible to use HARQ for all services, especially when low latency is important.

Service Access Points (SAPs):

MA-SAP: For system broadcast/paging messages.

MB-SAP: For connection-less messages/data.

MC-SAP: For connection-orientated messages/data.

D-SAP: Interface to the PHL.

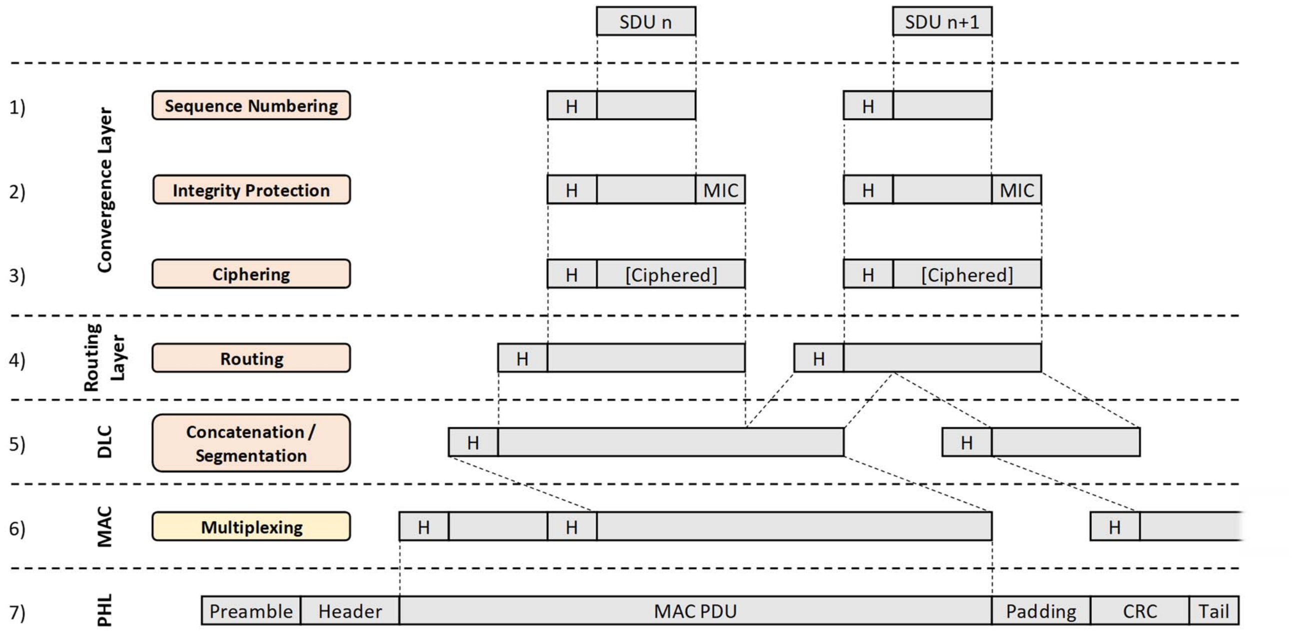
8.3.5.7 Physical (PHY) layer

PHY layer is out of scope of the present document. Refer to ETSI TR 103 514 [i.26] for description of the PHY layer.

8.3.6 Operations

8.3.6.1 Data Flow

Figure 35 represents the data-flow for a "slice" through the protocol stack.



NOTE 1: The header field added with sequence number.
NOTE 2: Message Integrity Check (MIC) generated by likes of AES CCM and appended to data.
NOTE 3: Ciphering provided by the likes of AES CCM.
NOTE 4: The overhead of the routing layer header field should be minimized, e.g. 1 bit header (possibly "stolen" from Convergence layer header), and only if this bit = 1 is an extended header is used (e.g. 2 × 16 bit IDs).
NOTE 5: The DLC header could be quite large and indicates PDU sequence number and other control information as well the length of data for each segment. In the example, there is both concatenation and segmentation.
NOTE 6: The MAC can multiplex multiple channels according to the physical channel capacity and data priority. In the example, a small SDU from another source (e.g. MAC control message) was also added to the transport block. Excess data will have to be re-scheduled for next transmission opportunity.
NOTE 7: Each block of data has its own header indicating the block type and length. The total length of MAC PDU can be computed from the length fields in the individual block headers, i.e. there is no need for a separate total length indication.
NOTE 8: Padding (with zeros or special "fill" data) is required so that the total length of MAC PDU + "Padding" + CRC + "Tail" can be coded as an integer number of OFDM symbols. Preamble and PHL Header depend on the PHL packet type/format.
NOTE 9: Addition of "Padding" and CRC could also be a MAC function.

Figure 35: Data-flow through the protocol stack

Figure 35 shows the transmission process going from top to bottom. The reception process is the logical reverse of this, of course with PDUs being re-assembled into un-concatenated SDUs, etc.

8.3.6.2 MAC Multiplexing

An important topic is the multiplexing operation that takes place in the MAC layer.

Legacy DECT has a hard split of the D-field into separate A-field (used for control fields such as MAC control and C-Plane data) and the B-field (used for U-Plane data). Different control fields are multiplexed onto the A-field according to "T-Mux rules" (i.e. time-multiplexing based on DECT frame number). The B-field is usually only for a single channel (e.g. voice or data), and so typically there is no multiplexing on the B-field.

However, as discussed elsewhere, this hard split of the D-field into A-field and B-field is inefficient for DECT-2020; since a permanent A-field is wasted bandwidth when it is not used - this is especially so for URLLC multi-bearer type of connections.

Elsewhere, the DECT concept of "E+U mux" is discussed. This would allow a variable/flexible split of the D-field into subfields carrying either "A-field type" data (i.e. C-Plane or MAC data) and "B-field type" data (i.e. U-Plane data).

Looking at similar protocols such as LTE, they also adopt a flexible approach, and allow the MAC transport block (i.e. MAC PDU) to be constructed from several SDUs, or varying sizes. These component SDUs come from several sources (logical channels), for instance MAC control SDUs, or multiple user data SDUs. Each logical channel source has an associated priority and bit rate (Prioritized Bit Rate (PBR)). The resulting multiplexing/priority scheme ensures that logical channels are handled in priority order, according to a target bit-rate, whilst still ensuring that lower priority channels receive some level of service.

Figure 36 shows how LTE MAC PDU is filled from 3 channels with different priority and PBR: Channel 1 is used first up to its PBR value; then Channel 2 is used up to its PBR; then Channel 3 up to its PBR (actually, it had less data than its PBR allowed); the MAC PDU still has capacity and s Channel 1 has another opportunity, etc.

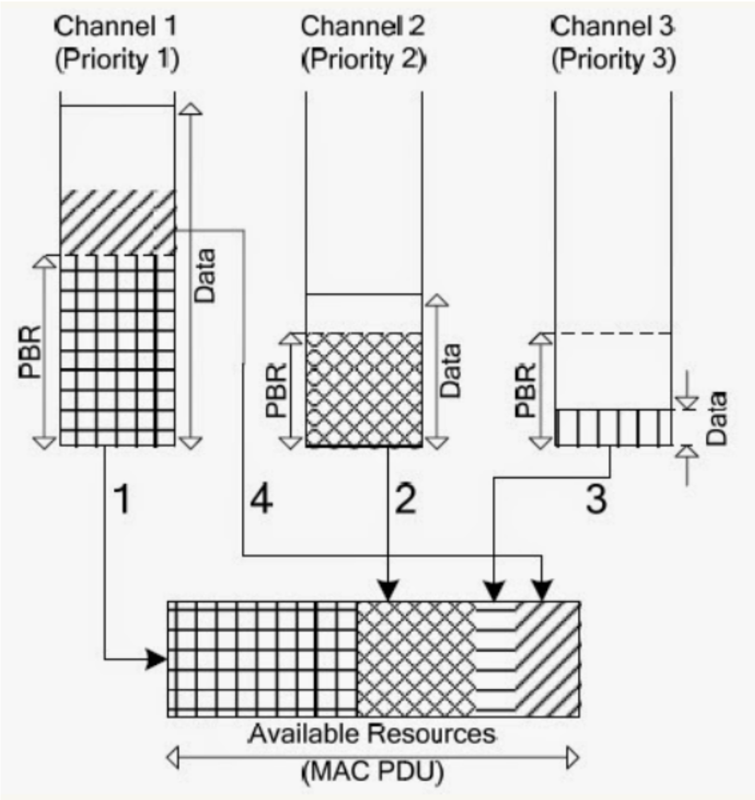


Figure 36: LTE MAC PDU filled from different priority channels

NOTE 1: In LTE, the MAC PDU header contains data indicating how many "data blocks" of data are present, what channel they come from, and the length of each block.

NOTE 2: In LTE-5G, the structure was modified, and each "data block" has its own header with channel and length information. This allows more opportunity to parallelize the processing of the blocks.

NOTE 3: One outcome of this process is that SDUs are segmented, and may only be partially transmitted. For instance, in the above example, not all the available data for Channel 1 or Channel 2 was transmitted in this MAC PDU, and it will have to be scheduled for the next transmission opportunity.

A similar approach can be taken with DECT-2020. Extending the concept of "E+U mux" as introduced previously, to handle multiple fields in a flexible manner.

Figure 37 shows a MAC PDU being filled by multiplexing 3 SDUs (from different logical channels, e.g. MAC control, C-Plane data, U-Plane data). Due to the allocated bit-rate limit, SDU 1 and SDU2 cannot be transmitted fully, and are segmented (the remainder will be sent on the next opportunity). Each block is preceded by a header ("Hdr") which identifies the type of block (i.e. which logical channel it came from), the length of data, and whether further blocks follow.

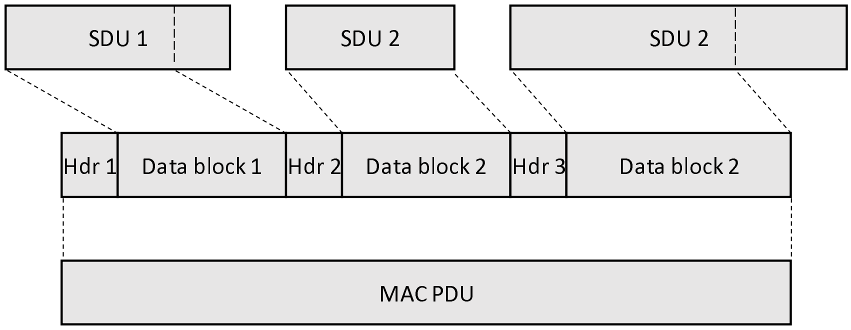


Figure 37: MAC PDU being constructed from different SDUs

It is possible to provide a lot of flexibility by configuring the channels' priority appropriately.

8.3.7 Security

The proposed protocol architecture has both data integrity and ciphering performed in the Convergence Layer. There are some "pros" and "cons" for consideration of ciphering at MAC layer (as in legacy DECT) or in higher layers.

One of the original assumptions was that it would be more efficient to handle data integrity at the higher layers, since a large SDU split into several PDUs only requires the addition of a single MIC at the higher layer, whereas it would require the addition of a MIC per PDU when performed at the MAC. However, it can also be argued that many small SDUs can be concatenated into 1 PDU, and this would only require the addition of a single MIC at the MAC layer, but it would require the addition of a MIC per SDU when performed at the higher layer. Which scenario is more important depends on the expected size of SDUs and PDU capacity.

In any case, there are other benefits to have data integrity performed at the higher layers, and this does still seem to be the best option.

Legacy DECT performs ciphering function at the MAC layer. One of the main criticisms of this technique is due to the way that the DECT multi-frame number is used as part of the IV. This means that any repeater or mesh-like systems have to decrypt and then re-encrypt the data if the relay incurs a change of frame number (which is normally the case).

Using a sequence number as part of the IV overcomes this issue (as long as the sequence number or part of it is transmitted with the data). The use of a sequence number also has other benefits for packet-based data systems (e.g. ensuring SDUs are deliver in-order, discarding duplicate SDUs, etc.).

However, for low-latency streamed data (such as non-IP-based voice) the addition of a sequence number does not seem natural. Of course, there is also the opportunity to not use ciphering for low-latency streamed data.

NOTE: One LTE source suggests that ciphering is performed at MAC layer for "Transparent Mode" RLC services, probably because of the lack of sequence number, etc. It is not clear how this works in terms of setting the IV, etc.

It has also been suggested that another benefit of MAC layer ciphering/integrity checks, is that it would enable received data to be checked for authenticity before being routed to other nodes - this would be particularly useful for mesh networks, to prevent Denial of Service (DoS) type of attacks. This requires further study.

It would certainly seem a good idea to have a flexible system which allows data to be ciphered were most appropriate.

The security topic for DECT-2020 is handled by a separate Technical Report (see ETSI TR 103 637 [i.28]).

8.3.8 Summary Analysis

On the face of it, it seems that the protocol architecture can be very similar to legacy DECT. The NWK layer is potentially un-changed (except for new messages, IEs, etc.). However, there is also the option to start with a "clean slate", which might make things simpler in many ways.

One way to ensure "compatibility" or "re-use" of existing standard might be to define some kind of "encapsulation mode" for NWK protocol messages, which enables the legacy style message structure to be encapsulated and sent without modification via the "new" protocol.

The Convergence Layer is clearly new, and has no legacy counter-part (although in principal the idea of a layer above DLC for ciphering is what the LU14 services does for DECT ULE).

The Routing Layer is also new, and at this stage is the least understood. More work is required in this area.

The DLC layer is very similar to legacy DECT DLC. There is possibility to re-use LAPC and LU10 for example, although changes will certainly be required. However, as with the NWK layer, it might be simpler to start with clean slate.

The MAC layer follows a similar structure, but fundamentally is very different and will require many new features and procedures.

8.4 Concept 3: MAC PDU Structure

8.4.1 Overview

Depending on the used PHL packet type/format, there will be a number of OFDM symbols allocated for data payload. For example, the Long Preamble PHL packet type (LP) will have 4 preamble symbols (STF + CTF), 2 header symbols (HF) and a variable number of data symbols (DF) depending on the total packet length. Whereas the High Efficiency Full-Slot packet type (HE-FS) has 1 pilot symbol and either 7, 8 or 9 data symbols. Irrespective of the specific PHL packet type, these data symbols are used transport the PHL SDU/MAC PDU.

NOTE: In fact, not all of data symbols in the PHL packet are available for the PHY SDU, since there is typically a small number of service/tail/padding bits which are required for correct operation of the PHL. For example, to initialize the data scrambler, terminate the convolution coder and ensure that the total number of data bits is an integer number of OFDM symbols.

Figure 38, depicts a generic PHL packet format (with preamble, PHL header, PHY SDU and tail/padding). However, the MAC packet format is agnostic to the specific PHL packet format, as it just concerned about the PHY SDU/MAC PDU.

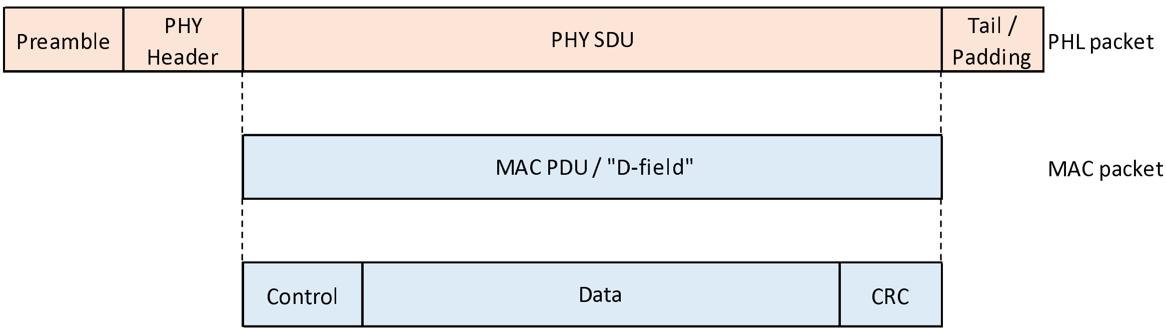


Figure 38: MAC PDU structure

The MAC PDU is more or less equivalent to the "D-field" or traditional DECT.

In traditional DECT, the D-field is further split into the "A-field", "B-field" and "X-field". The A-field is generally used for control information ("C-plane"), and the B-field for user data ("U-plane"). However, for DECT-2020, especially for low-latency bearers and/or multi-bearers, the overhead of a permanent A-field on each Traffic Bearer slot would be quite detrimental to the efficiency of a system.

In current DECT standard, for the B-field, there is the concept of transporting "E-type", "U-type" or "E+U-type". This mode of operation is signalled by the "BA-bits" on a slot-by-slot basis, and allows the B-field to carry either E-type (MAC, C_F, G_F data), or U-type (I_N, I_P, etc.) or both.

For DECT-2020, a similar concept can be employed, with some bits in the "Control" field being used to indicate the type of contents of the "Data" field, i.e. either E-type, U-type, or E+U-type, as shown in Figure 39.

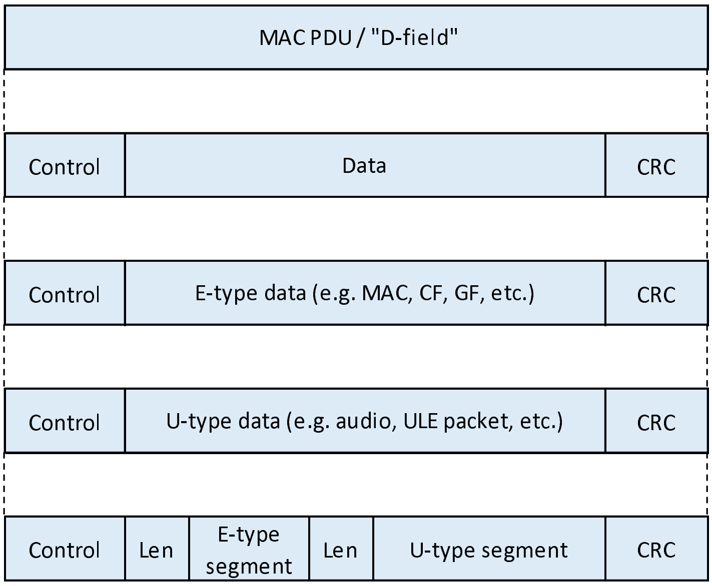


Figure 39: Flexible MAC PDU format

The flexible approach of allowing E-type, U-type or a mix of E+U-type seems quite useful. Different traffic scenarios can utilize the ability to select E-type/U-type differently. For example:

- Initial Access Bearer, prior to establishment of Traffic Bearer:
  - E-type only, since clearly no traffic at this stage.
- ULE Packet Mode:
  - E+U-type, since it is required control and data from the start.
- Regular voice/data Traffic Bearer:
  - E-type during initial call setup (e.g. transfer of CC-SETUP, CC-CONNECT, etc.).
  - U-type when data service is connected.
  - E+U-type when additional signalling is required (e.g. CC-INFO, or the CC-RELEASE):
    - When E+U-type is used the user data capacity will be less than normal, which has to be taken into account somehow.
- Low latency audio Traffic Bearer:
  - E+U-type for one of the slots, and U-type only for the others.
    - User data capacity on the E+U-type slot will be less than the others, which has to be taken into account somehow.

NOTE 1: The traditional DECT terminology of "E-type" and "U-type" is used here. However, the final terminology used for DECT-2020 may vary.

NOTE 2: When E+U-type is used the user data capacity will be less than normal. This might be a problem for services requiring a fixed throughput (e.g. audio streaming services). However, it might be possible to accommodate this in the codec (variable rate codec?) or by temporarily going to a higher MCS for the packet (only possible in some PHL packet formats). Alternatively, E+U-type can always be used which would keep data rate fixed, but be less efficient.

NOTE 3: Use of E-type during call setup means it will be faster.

NOTE 4: Some MAC packets may have specific formats, e.g. Dummy Bearer/Beacon Bearer, probably has a unique packet structure, similar to ULE Dummy Bearer.

8.4.2 Control Field

Bits in the Control field will indicate the format of the following Data field. The number of bits should be approximately 3 or 4, which would allow for 8 or 16 different Data field formats, or other special uses. This should be sufficient for current uses and also for future extension features.

In addition to the bits indicating the type of contents of the Data field, it could also contain other bits, for example to do with ACK mechanism, quality control, encryption, identities, etc. In many ways it has features of existing A-field header (i.e. BA-bits and Q-bits).

The following is a rough estimate of the number of bits required in the Control field:

- Data field format: 3-4 bits.
- Identity information (e.g. FMID/PMID): 24-32 bits.
- Encryption information:
  - Encryption on/off: 1 bit.
  - Key index: 2 bits.
  - 8 LSBs of IV: 8 bits.
- ACK/NACK/BCK bits: 2 bits.

This puts the size of the Control field in the range 40 to 49 bits, so approximately 5-6 bytes. Maybe it can be less. For example, are identities required in each packet? This is typical for WiFi type of system, but in DECT usually these MAC identities are only in MAC messages.

8.4.3 Data Field

The Data field is equivalent to the B-field, but rather than being split into fixed length subfields, it is more of a free-format, which can carry either all E-type (MAC, G_F, C_F, CL_F), all U-type (I_N, I_P), or a mix of the two, i.e. E+U-type. A length field ("Len") indicates the length of each data section (either in bytes or blocks (e.g. 8 bytes)). Multiple blocks per packet could be supported (using "more" bit and additional length field, etc.).

Different formats of Data field are possible according to a header field in the Control field. The following is a list of suggested formats:

- Beacon bearer format (a special packet format).
- All E-type format.
- All U-type format.
- E+U-type format.
- Traditional A+B fixed field format.

8.4.4 CRC Field

The CRC field is used to detect any uncorrected errors in the received data. It is not used for any security/authentication purpose. The CRC is performed on the whole D-field. If the CRC check fails then the whole packet is assumed to be corrupt, and the data should be discarded.

NOTE: The PHL layer provides Forward Error Correction (FEC). However, it is not clear if the proposed coding scheme will also detect errors that have not been corrected. If the FEC also indicates that errors still exist then the CRC could be weaker, otherwise a stronger CRC is required.

The size of the CRC field could be in the range 8 to 32 bits, depending on packet length, and FEC coding scheme.

8.4.5 Data Fragmentation

Data fragmentation will be handled by DLC layer in manner similar to current DECT (i.e. LAPC service for C-plane, and Lux services for U-plane).

However, there are a few things should be considered:

The current DECT standard assumes fixed sized data frames for the MAC, i.e. each MAC SDU (i.e. DLC PDU) will be of a fixed and constant size depending on the currently supported service. For example, when C_S is being used the C-plane fragments are 5 bytes each, and when C_F is being used the C-plane fragments are 8 bytes each. Similarly for U-plane data, the PDU size is fixed and depends on the LUx service.

For DECT-2020, a more flexible approach is needed, not only because of the possibility to mix E-type and U-type data, but also due to varying data rates by PHL layer MCS, which could possibly be changed dynamically in some cases.

For this purpose, rather than having fixed frame sizes (configured at service establishment/modification), the MAC will request the number of bytes that should be supplied each time in the "MAC_CO_DTR-ind" (or equivalent) primitive. The DLC should supply the requested number of bytes (or possibly less).

The MAC might have to perform some adaptation when transporting U-plane data over E+U-type packet. For example, a U-plane PDU might have to be split into smaller chunks, and sending each part tagged with "segmentation info" (basically a flag indicating "1st part of PDU segment", "1st and only part of PDU segment", "not last part of PDU segment" and "last part of PDU segment"). This is similar to standard DECT when using I_{PF} channel (see ETSI EN 300 175-3 [i.3], clause 10.8.4). Since these segments are not numbered, one assumes that ARQ mechanism can only occur at whole DLC PDU level.

8.5 Identities and Addressing

8.5.1 Review of other technologies

Table 9 summaries the main identities used in 3GPP (including NR), IEEE 802.11 [i.34] and legacy DECT. This information is used as reference when considering the requirements and considerations for DECT-2020 identity and addressing schemes.

Table 9: Identities in other wireless technologies

Identity	3GPP NR systems	IEEE802.11	DECT
Network ID (Transmission method)	PLMN ID containing MCC + MNC MCC = 3 digits MNC = 2 or 3 digits Total: 3 bytes, 24 bits. (Periodically in SIB)	SSID: 0 - 32 bytes (Periodically in Beacon message and Probe responses)	RFPI: containing ARI and RPN. Total: 39 bits.
Local Network ID (Transmission method)	Tracking area code: 24 bits RAN area code: 8 bits Cell ID: 36 bits (Periodically in SIB) PhysCellId: 0…1007 (Periodically in MIB)	BSSID: 48 bits, in case of Infrastructure network the BSSID is MAC address of the AP. (Every packet)	RPN: Identifies FP LA: x bits of RFPI.
Permanent Device ID (Transmission method)	PEI: Either IMEI (15 digits) or IMEI-SV (16 digits) (Initial authentication based on network request)	MAC address: 48 bits (Every packet)	IPUI: 40 bits (N-type) IPUI: 64 bits (S-type) Other exist also.
Permanent subscription ID (Transmission method)	IMSI, or Network specific identifier (Initial authentication based if no Temporary ID available)	Out of the scope. Can be IMSI in 3GPP - WiFi interworking	PARK A to E. 31- 36 bits with IPUI.
Temporary subscription ID (Transmission method)	5G-S-TMSI: 48 bits. Identifies the UE within the tracking area (Connection setup)	N/A	N/A
Radio ID (Transmission method)	RNTI: 0…65535 (16 bit) (Once per scheduled transmission in PDCCH, hashed on top of CRC)	MAC address: 48 bit (Every packet)s	TPUI: 20 bits

The 3GPP technologies including NR have very a specific addressing scheme starting from PLMN Id, down to tracking area codes and different cell IDs to identify different networks and part of the network. Clear separation between subscription and device identities is done by having IMEI and IMSI numbers. Additionally, the network specific identifier can be used, such as user ID, which provides means to operate without IMSI numbers. One key distinguishing fact of 3GPP systems is the operation on licensed spectrum without the possibility of different overlapping networks in same frequency.

Therefore different network IDs are only broadcasted in system information broadcast (SIB) messages quite infrequently. Cellular system are organized in cells, each cell is serving one area on one operating frequency. Since the communication is always between UE and BTS, where the BTS performs scheduling, there is no ambiguity between transmitter and receiver of the packet in that cell. The identification of each individual transmissions is done by Radio ID i.e. different allocated RNTI values in physical common control channel. RNTI information is also used to indicate the allocation of downlink data transmission and indicate uplink grants for UE uplink transmission.

IEEE 802.11 relies on fixed MAC address of 48 bits as well as 32 byte SSID used as network ID. MAC address is directly the ID given by the chipset or device manufacturer that is used in all packet transmissions. Originally this was fixed allowing to monitor offline any device traffic, however there are software solutions available where random address values are used, but typically these are not used by the end user. Each MAC packet may contain following addresses.

Basic service set identifier (BSSID), source address (SA), destination address (DA), transmitting STA address (TA), and receiving STA address (RA).

Frames may contain optimization not to send all address information in all frames, in most typical case is when devices are operating with access point where addresses are used as follows:

- Receiver address: 48 bits.
- Source Address: 48 bits.
- BSSID: 48 bit MAC address of the Station operating in AP.

One important aspect of IEEE 802.11 addressing scheme is that it allows overlapping systems to operate on same spectrum without any resource coordination. Each receiver can decide from received packet is the data intended for them or to be routed forward or to be discarded. Additionally, different STA's can learn other BSSIDs operating in same channel and other devices operating on those BSSID.

Using 48 bit fixed global address no coordination of address are needed inside the SSID or BSSID. The drawback is that the addressing introduces quite large overhead.

Finally, when considering IP-networks one should consider IPv6 addressing. The IPv6 is addressing is shown in Figure 40. The IPv6 address consist of 48 - 64-bits long routing prefix and 16 - 0 bits of subnet ID. These parts of the address are controlled by the IP network operator and are not necessarily controlled by the DECT-2020 network operator. After this information there is a 64-bit interface identifier that can be automatically derived from Ethernet 48-bits MAC address (which is obtained from DHCPv6 server) or randomly selected or assigned manually.

The existing Ethernet 48-bits MAC address is extended to 64-bits by adding 'FFFE'H (in Hex) into middle of MAC address and toggling the 7th most significant bit of the EUI-64, to 1 indicating that the MAC address is globally unique.

bits	48 (or more)	16 (or fewer)	64
field	*routing prefix*	*subnet id*	*interface identifier*

Figure 40: IPv6 Addressing (General unicast address format)

8.5.2 Requirements for DECT-2020

The questions for DECT-2020 system design now are: How different networks are identified and how many devices are expected in the network? Due to nature of DECT spectrum the network identification should support multiple co-located and independent networks which are all operating on the same DECT spectrum.

The second question regarding how many devices could be operating in a single DECT-2020 network? Since a single network should support extremely high quantities of FPs, and OFDM radio parameters are suitable both for indoor and outdoor deployments, the geographical area of a single DECT network could be significant, e.g. Munich city-wide network. Such smart city network could contain hundreds of thousands of devices in one system. In mesh deployment all devices can act as FP to other devices and only a limited set of devices will have fixed FP role with backend connection. Thus, the device addressing scheme should be same for FP and PP, and the device address should be independent from the role/type of the device.

Single networks could also have multiple independent locations without a continuous coverage, e.g. Berlin, Munich and Hamburg could allow devices to move between cities and connect to each city network. Therefore the backend systems needs to be able to identify which location downlink data needs to be sent.

To support unicast, broadcast and multicast services, a set of addressing scheme for broadcast and multicast should be considered. This can be achieved by selecting one address for broadcast and set of addresses for multicast use from the basic device address space.

Finally, it can be expected that DECT-2020 networks are integrated as part of common IPv6 networks and thus used addressing should be compatible IPv6 addressing.

Summary of DECT-2020 deployments and services that needs to be supported by Network and Device addressing are:

- Overlapping networks may and will exist.
- Single network could be very large - a small residential network is a special miniaturised case.
- Number of nodes in single network can be up to several millions.
- Broadcast and multicast services should be supported.
- There may be nomadic users which may move from one network location to another or one network to another network.
- IPv6 addressing scheme is anyway used in backend systems.

8.5.3 Considerations for DECT-2020

To support mobile nodes, the possibility to allocate a globally unique network ID would be clearly beneficial for identifying any network. In 3GPP this can be done by 24-bits PLMN-ID, with using global allocation scheme. In Wi-Fi the 32 bytes of the SSID are provided for this purpose; although the network owner can select freely the network ID without the guarantee of globally unique network ID. If 3GPP type of allocation scheme does not exist - higher number for Network IDs could be needed to minimize the use of same Network IDs.

On the other hand it would be beneficial if receiver would be able to detect from each packet whether the received packet is transmitted by member of the same network or if the transmission is part of some other network. It is considered that very long IDs alone are not feasible due to associated overhead. Thus the network identification could be partly similar as in Wi-Fi:

- Long Network ID can be broadcasted periodically in MAC payload and indicating the short ID.
- Short Network ID transmitted in packet header allowing packet filtering and discarding.

Proposal 1:

- Long Network ID can be broadcasted periodically in MAC payload, and it can also indicate the Short Network ID used in PHY packets.

Proposal 2:

- Short Network ID is transmitted in packet header allowing packet filtering.

DECT-2020 should also support multiple formats of the Long Network ID including the support of PLMN ID for 3GPP interworking as well as IPv6 based addressing, where a single network can be identified as IPv6 address, i.e. the router address(es) of the network maintaining list of nodes in the network. This is the "Home Agent" in device home network and "Foreign Agent" visiting network in mobile IP terms.

The size of the short Network ID should be discussed but it could be e.g. 24 bits.

It is expected that all devices will have long (64-bit) unique ID that is built based on manufactures ID, or other immutable technologies and running number i.e. Ethernet MAC address with additional extension as done in case of using IPv6 addresses or combined from Long Network ID and Node ID. One option is that in case of IPv6 the Network ID is the 32 MSB bits of the IPv6 interface identifier and Device ID is 32 LSB bits of the IPv6 interface identifier. However, the standard may allow different options, but 32-bit Device ID is sounds attractive in terms of IPv6 compatibility as well as allowing high number of devices being used in a single DECT-2020 network.

The addressing scheme for FP and PP should be the same and broadcast service as well as multicast service should be obtained by using specific address value ranges.

Proposal 3:

- Device addressing scheme should be same for FP and PP, and thus the node address is independent from the role/type of the device.

Proposal 4:

- Broadcast and multicast services are part of global addressing scheme, using single address for Broadcast and another set of addresses for multicast.

8.5.4 Proposal for DECT-2020

The following requirements are proposed:

- Overlapping networks may and will exist.
- Single network can be very large - a small residential network is a special miniaturised case.
- Number of nodes in single network can be up to several millions.
- Broadcast and multicast services should be supported.
- There may be nomadic users which may move from one network location to another or one network to another network.
- IPv6 addressing scheme is anyway used in backend systems.

The following solutions are proposed:

- Long Network ID can be broadcasted periodically in MAC payload, broadcasting is used to indicate the Short Network ID used in PHY packets.
- Short Network ID is transmitted in each packet header allowing packet filtering.
- Device addressing scheme should be same for FP and PP, and thus the node address is independent from the role/type of the device.
- Broadcast and multicast services are part of addressing scheme, using single address for broadcast and another set of addresses for multicast messages.

8.6 Beacon Bearer Contents

8.6.1 General

It is assumed that a PHL packet format, such as that described in clause 7.3.2 is used. This has the STF and CTF fields for synchronization and channel training. There may (or may not) be a Header Field (HF/A0) depending on the final design. The rest of the packet (i.e. 5 or 6 OFDM symbols) is the "Data" field and it is this data field that is the main concern of this study.

8.6.2 Sync Pattern

Legacy DECT uses a special pattern in the "sync field" which the receiver uses to lock onto and determine the bit position timing. In legacy DECT, all bearers broadcast "sync field" and "A-field" and the PP can use any available bearer to obtain a lock to the FP - the PP just needs to check the FP's identity to make sure it is the right FP.

However, in ULE, only the ULE dummy bearer contains the full information needed for the PP to lock quickly - if a PP were to lock on to a regular bearer, it would have to reject it and try again (wasting precious time and energy). To overcome this, the ULE dummy bearer uses a special "sync field" (which has a different pattern to the regular one). This allows a ULE PP to lock only to ULE dummy bearers, and so it can ignore other DECT bearers.

DECT-2020 devices will not be able to lock (meaningfully) on to just any transmission, they should only be able to lock to the beacon bearer (of which there will only be 1 or possibly 2 in multi-cell deployments). As such, some kind of "marker" may be beneficial, to identify which bearer is the "dummy bearer" (aka "beacon bearer"). This "marker" serves a similar role to the special sync word used in the ULE dummy bearer.

The sync pattern/marker could take several forms. For example, a specific coding in the Header Field (HF/A0). However, this would still require the PP to try to receive and decode all transmissions until it found the specific transmission (i.e. the beacon bearer) that it was looking for. Another solution could be to use some unique pattern of sub-carrier usage in the CTF (e.g. specific sub-carriers being zeroed) which the PHL could be determine directly (without MAC processing).

NOTE: The problem of "sync" here is referring to initial synchronization, e.g. when a PP is initially powered-on, enters a new cell, or is trying to register with a new FP. Once a PP is locked to an FP, it is assumed that the beacon bearer position can be tracked relatively robustly without the need to "re-acquire" fully.

8.6.3 Identity

The FP beacon bearer needs to carry the identity of the FP. This is required so that PP's trying to lock to "their" base can find it quickly and efficiently.

Table 10: Identity contents

Field	Size	Comment
Option 1: Identity (full)	5 bytes (40 bits)	
Option 2: Identity (partial)	16 bits	To save space, a partial identity might be acceptable, but this should always be augmented by the transmission of the full identity at some lesser frequency.

8.6.4 System Information

There are various pieces of data which could fall into this category. This information is loosely categorized as static or quasi-static. More time-varying information, such as dynamic blind slots, RAC resources, etc. is handled under the section on "MAC Layer Information" (see clause 8.6.5).

Table 11: System information contents

Field	Size	Comment
Slot number	4 bits (0-11)	Only required if the beacon bearer can be on any slot. If the beacon bearer was always on slot 0, for example, then this field would not be required. See notes 1 and 2.
Carrier number	4 bits (0-15)	Carrier number could be redundant (since the receiver should know what carrier the transmission was received on). However, it is still broadcast in legacy DECT, and it does have some utility when disambiguating carrier numbering over different RF bands. See note 3.
RF band	5 bits	It is expected that DECT-2020 will operate in one of several bands, e.g. DECT core band 1 880 MHz - 1 900 MHz, "DECT extension band" 1 900 MHz - 1 920 MHz, UPCS band 1 920 MHz - 1 930 MHz, or other IMT band. Therefore, a mechanism to identify which band is used is needed.

Field	Size	Comment
RF carriers	10 bits	Some RF carriers might be unavailable for whatever reason. This bit mask identifies which ones are used (up to 10). More bits could assigned if necessary.
Frame number	4 bits	Frame number might be required to synchronize activity like multiplexing cycle for broadcast information, etc. See note 4.
Operation "model"	2 bits	The operation "model" used by the FP allows some flexibility in how the system operates, e.g. more "WLAN-like" or more scheduled. This impacts slot availability, channel access, etc. Currently only "Model A" is defined (see clause 7.1).
Up-link/down-link configuration	4 bits	This field combines the sub-frame periodicity and preferred up-link/down-link configuration (see clause 7.1). Rather than allow full flexibility for all options and configurations it is assumed that only certain combinations will be typically used. As such, this can be coded in 4 bits (16 options).
FP capabilities	TBD (e.g. 16 bits)	In legacy DECT, there are several FP capability fields (in fact, 3 separate Q_T messages are used for this purpose). This information is very static (should hardly ever change), and is only really required when a PP wants to acquire a new FP (e.g. when registering, changing cell, etc.). An alternative to broadcasting this data continuously would be allow higher layer messages to exchange the information during registration procedures (much as the PP does today with its <<Terminal Capability>> IE. However, some basic (low-level) capabilities might still be required. For example, indicating which MCS modes are supported by the FP, etc. These should be kept to a minimum, e.g. just sufficient to allow a PP to decide how to communicate with the FP, and everything else can be deferred to higher layer messaging.
Multi-cell configuration	TBD (e.g. 4 bits)	It is assumed that some bits will be required to identify multi-cell configuration, e.g. whether the FP is part of PABX, whether a repeater is used, etc. This is left for further study.
Transmitter power	4 bits	Some capacity is provided to specify different allowed transmitter power classes (16 options), which might be important particularly when dealing with additional RF bands, etc.

NOTE 1: It is assumed that the beacon bearer is always on full-slot boundary (never half-slot).
NOTE 2: It is assumed that the beacon bearer is always in the first half of the frame (i.e. slot 0-11). If this is not the case, then a 5 bit field is required.
NOTE 3: Legacy DECT typically uses 5 bit numbers for carrier information, so that it can handle different RF bands.
NOTE 4: Legacy DECT uses a larger multi-frame number, but this is mainly for the MAC based encryption IV, which should not be required in DECT-2020.

8.6.5 MAC Layer Information

There are various pieces of data which could fall into this category. In general, this data is more time-varying than the "System Information" (see clause 8.6.4).

Legacy DECT systems use MAC layer information to indicate various things, but the most important ones are:

- "Blind slots" (i.e. which slots are busy/blind and which are free) for various configurations.
- Position of other bearers (including other beacon bearers, etc.).
- RFP status.

The "blind slots" information in legacy DECT is a bit mask, indicating which slots are busy/blind or free. In legacy DECT, the FP can only transmit to (or receive from) one PP at a time, so if a particular slot is busy then it is busy on all carriers. However, with DECT-2020 this is not the case (i.e. it can transmit to multiple PPs on different carriers in the same slot). Another point to note, is that in legacy DECT the "blind slots" are always indicating a slot-pair, e.g. slot 0 & slot 12 is always indicated together - it is not possible to indicate slot 0 as busy and slot 12 as free.

DECT-2020 needs some way to indicate busy/free resources, but this cannot be handled by a simple bit mask as is it is legacy DECT.

Another important difference between legacy DECT and DECT-2020 is that the resource allocation is performed predominantly by the PP in legacy DECT, and predominantly by the FP in DECT-2020. In fact, in DECT-2020, the FP only really needs to inform the PPs about RAC resources. Everything else is scheduled by the FP, and so the FP does not need to broadcast availability of other free slot & carriers.

There are a few main options:

1) Bit mask of everything

   - For 10 carriers and 24 slots, this would be a 240 bit matrix showing which slot & carrier combinations were available for RAC use.

2) Bit mask of specific carriers

   - If RAC use can be limited to a small number of carriers, e.g. 2-4 carriers, then it is not necessary to specify the full matrix for the bit mask. Suppose the number of RAC carriers = M, then the RAC positions can be identified using M × (4 + 24) bits, i.e. 4 bits to indicate the carrier number and 24 bits for that carrier's bit mask, per RAC carrier. So, for M = 3, this would be 3 × (4 + 24) = 84 bits.

3) Identify specific RAC positions only

   - Each RAC position can be identified by 5 bits for slot number, and a 4 bits for carrier number, i.e. 9 bits in total. If the number of advertised RAC positions = N, then N × (5 + 4) bits are needed to identify these positions. For example, suppose N = 10, this would require 10 × (5 + 4) = 90 bits.

Option (1) has the most flexibility, but if RAC channels are sparse (which is probably the case) then this does seem wasteful.

Option (2) seems viable. There are pros and cons to this limited number of RAC carriers. On the "con" side, it reduces flexibility. However, on the "pro" side it makes it quite clear which carriers should be used for RAC procedures, and this has the advantage of helping to keep these carriers clear of interference (which could also be advantageous for neighboring DECT-2020 systems).

Option (3) seems viable. Although only a small "pool" of RAC channels can be advertised at a time, they can be constantly replenished as long as the FP has available resources. The beacon bearer is broadcast once per frame (10 ms), so the pool of RAC positions can be replenished every 10 ms. Of course, the value of N sets an upper limit on how many PPs can successfully attempt RAC connections per frame. In practice, since RAC attempts are at random, there will be collisions, and a bigger value of N would reduce the probability of a setup collision.

There is clearly some trade-off between the proposed options. Nevertheless, option (2) with M = 3, or option (3) with N = 10 both seem viable, and require similar number of bits.

In addition to the RAC positions, a similar mechanism is needed to identify the position (i.e. slot/carrier) that the FP will use to send any response to the RAC request. The PP (that sent the RAC request) will need to listen on the position. However, generally the number of these RAC response positions will be a lot less that the RAC positions themselves, and it might even be possible to respond to multiple PP's RAC requests with a single RAC response (this is left for further study). Due to the much smaller number of positions, a scheme like (3) seems suitable.

Table 12: MAC layer information contents

Field	Size	Comment
RFP status	4 bits	Can be used to indicate, if an RFP is busy or not (for various service types). This is typically useful for multi-cell systems, since a PP can move to another cell, if the current RFP is "busy".
Option 1: RAC bit mask of everything	240 bits	
Option 2: RAC bit mask of M carriers	M × (4 + 24) bits	Assuming M = 3, this is 84 bits
Option 3: RAC N positions	N × (5 + 4) bits	Assuming N = 10, this is 90 bits
RAC response	O × (5 + 4) bits	Assuming O = 4, this is 36 bits.

NOTE 1: The above analysis assumes a maximum of 10 carriers is supported. This can be represented by 4 bits. If a higher number of carriers is required, then a larger bit field is required. Legacy DECT uses 5 bit field for carrier number (i.e. up to 32 carriers).

NOTE 2: The above analysis assumes that RAC setup is always on full-slot boundary (not half-slots). If half-slot support is required for RAC, then larger bit fields will be required (i.e. 48 bit mask for option 1 & 2, and a 6 bit slot number for option 3).

8.6.6 Paging Information

Paging is required in order to initiate connections with PPs. For example, when there is incoming call or data waiting to be delivered. It is used when the PP does not currently have a scheduled connection that the FP can use for communication.

When the PP is not connected to the FP, it is normally in some power saving mode, where it is receiving the FP beacon bearer less frequently (e.g. ever 4 frames, every 16 frames, or even less frequently). Thus it is usually necessary for paging to be repeated a number of times. Repeating paging data also benefits in case of interference.

Paging can be scheduled on specific frames, e.g. frame 0 (as in legacy DECT). However, if the FP knows that a PP is listening to the beacon bearer every frame, then it can page that PP on any frame. There might also be some benefit in using different paging frames for different PPs, e.g. some PPs use frame 0, some PPs use frame 4, etc. so as to distribute the load (both of the paging data, and the setup attempts from the resulting page messages).

Paging information could be done in a several ways:

- Paging identity (similar to TPUI) to indicate the paged PP:
    - Additional information could also be included, such as type of required service, or even which resources (slot & carrier) the PP should be used for establishment.
- Bit mask (similar to ULE) which can indicate several PP's simultaneously:
    - One bit is used to indicate a single PP, and this operates in a way similar to a mailbox flag, i.e. indicating that a PP needs to contact the FP in order to see what the "mail" is (which could then indicate for example, an incoming call, data available, or a request for status information).
    - Use of bit-mask in this way is quite efficient, and the bit could remain "set" for relatively long time whilst waiting for the PP to wake up.
    - Different PPs can check their mailbox flags on different frames, e.g. frame 0 is used for PPs 1-10, frame 4 is used for PPs 11-20, frame 8 is used for PPs 21-30, etc.
    - This is similar concept to how DECT ULE operates.

The two paging schemes (i.e. identity and bit mask) are complementary and could be used concurrently, if there is sufficient bandwidth in the beacon bearer.

ULE-type devices need extra consideration, due to the very long sleep periods. A system similar to ULE paging for regular DECT can be considered, but this is left as a further study.

Table 13: Paging information contents

Field	Size	Comment
Paged device identity	20 bits	PP identity such as TPUI
Additional page information	16 bits	For example, allocated resources for page response: Slot (4 bits), Carrier (4 bits), MSC (4 bits), Other (4 bits)
Paged device bit map	10 bits (or more)	Bit mask indicating the paged PPs. Bits (and frame number(s)) are assigned to the PP during some registration phase

Paging for multi-cast/connectionless downlink data is also possible, and would require additional paging addresses. In addition to the address, the FP should also supply information about the designated slot/carrier for such a connectionless downlink data broadcast. This is left for further study.

8.6.7 Total Contents Size

Using the estimates in the previous clauses, the size of beacon bearer contents can be summarized as:

- Identity ~40 bits.
- System Information ~57 bits.
- MAC Layer Information ~130 bits.
- Paging Information ~46 bits.
- Total ~273 bits (~34 bytes).

Clearly this value is very approximate, and it would be wise to provision for additional capacity in order to allow flexibility and extensibility (i.e. for future-proofing the design). Consequently this value should probably be taken as a lower bound.

A less conservative estimate could be made if it is assumed increased support for capability bits, number of advertised RAC channels and increased paging capacity, etc. For this purpose, a more realistic upper bound of 150 % of the lower value (i.e. ~410 bits) can be assumed.

8.6.8 Beacon Bearer A/B Split

8.6.8.1 A/B Capacity

Assuming a "dual mode" beacon bearer as described in clause 7.3, the beacon bearer will consist of 6 data symbols (or 5 if separate Header Field (HF/A0) is needed by the design). It is assumed that the "A-part" will be transmitted with a robust MCS (e.g. MCS 1) and the "B-part" will be transmitted with a more aggressive MCS (e.g. MCS 6). The split between A & B parts is open to debate, and will ultimately require some trade-off in the design. Table 14 shows the resultant data capacity for several configuration scenarios.

Table 14: Capacity of A/B parts for various configuration scenarios

Configuration		A-part capacity	B-part capacity
1	A-part: 6 OFDM symbols, MCS=1 B-part: N/A	6 × 52 = 312 bits (39 bytes)	N/A
2	A-part: 1 OFDM symbols, MCS=1 B-part: 5 OFDM symbols, MCS=6	1 × 52 = 52 bits (6,5 bytes)	5 × 234 = 1 170 bits (146,25 bytes)
3	A-part: 2 OFDM symbols, MCS=1 B-part: 4 OFDM symbols, MCS=6	2 × 52 = 104 bits (13 bytes)	4 × 234 = 936 bits (117 bytes)
4	A-part: 3 OFDM symbols, MCS=1 B-part: 3 OFDM symbols, MCS=6	3 × 52 = 156 bits (19,5 bytes)	3 × 234 = 702 bits (87,75 bytes)
5	A-part: 4 OFDM symbols, MCS=1 B-part: 2 OFDM symbols, MCS=6	4 × 52 = 208 bits (26 bytes)	2 × 234 = 468 bits (58,5 bytes)
6	A-part: 3 OFDM symbols, MCS=0 B-part: 3 OFDM symbols, MCS=4	3 × 26 = 78 bits (9,75 bytes)	3 × 156 = 468 bits (58,5 bytes)

NOTE 1: MSC 0 = BSPK modulation with 1/2 coding rate.

NOTE 2: MSC 1 = QPSK modulation with 1/2 coding rate.

NOTE 3: MSC 4 = 16-QAM with 3/4 coding rate.

NOTE 4: MSC 6 = 64-QAM with 2/3 coding rate.

The values shown in the Table 14 are the gross values. Additional overhead for things like headers and CRCs also need to be taken into account. In particular the A-part should probably contain an information field describing the split (i.e. how many symbols of A, how many of B) and the modulation for the B-part.

Recall that the "A-field" in legacy DECT is only 5 bytes of useful payload, so the capacity values shown in Table 14 represent a useful increase in capacity compared to legacy DECT.

The A/B split and used MCS could be fixed by the standard, or left flexible to varying degrees. For example, having fixed size and MCS for A-part; or even having variable size and MCS for A-part (the receiver might need to decode with multiple MCS to see which one works); it is assumed that the size and MCS of the B-part can be coded in the A-part.

8.6.8.2 A/B Content Split

As seen in the previous clause (see clause 8.6.7) the total size of the required beacon bearer contents has a lower bound of approximately 273 bits and an upper bound of approximately 410 bits.

Taking the lower-bound value, using configuration 1 (see clause 8.6.8.1), it would in fact be possible to carry all of this information wholly in the A-part (i.e. no time-multiplexing or B-part required), even when taking into account additional overhead such as CRC (assume 16 bits) and headers, etc.

However, the upper-bound value does not fit into a single A-part in this manner. Therefore it would be necessary to either a) restrict the estimate for the upper bound size, or b) use higher MCS with A/B split as previously suggested.

Clearly, if the upper-bound value can be restricted (e.g. by limiting the number of RAC channels or paging capacity) then it would be very advantageous to provide all the beacon bearer contents in a single A-part with a robust MCS rather than resorting to A/B split, as this would significantly reduce complexity, and offer the best possible service to all users (whatever their range/radio conditions).

However, if this is not possible, and A/B split is needed then there are several ways to do this.

Even when using the upper-bound value, a very high MCS or large size for the B-part does not seem to be required. For example, it seems sufficient to use 2 OFDM symbols at MCS 6 (468 bits) or 3 OFDM symbols at MCS 4 (468 bits). This would leave 4 OFDM symbols at MCS 1 (208 bits), 3 OFDM symbols at MCS 1 (156 bits), etc.

The B-part would contain all of the required information, plus CRC.

The A-part would need to be multiplexed in some way, and would also need some header plus CRC. The Identity and MAC Layer Information (mainly the RAC channels) will be given the highest priority (i.e. frequency of transmission) for the multiplexing scheme. The precise nature of this multiplexing is left for further study, but it would seem that there is sufficient capacity and scope to do this without too much difficulty.

8.6.9 Additional Considerations

- Multiple beacon bearers:
  - In some cases, multiple beacon bearers will be required, for example in PABX case (to ensure that one is always visible to a PP when performing handover).
  - Multiple beacon bearers can also be used for resilience (in case of interference).
  - In the case of multiple beacon bearers, each beacon should transmit the same information (with exception of obvious data such as the slot & carrier of the transmission).
- Size of bit-fields for slot numbers:
  - Legacy DECT typically uses 4 bits in order to encode a slot number 0-11, since this represents a slot-pair.
  - DECT-2020 probably should not have any such restriction of slot number (or allocation by slot-pair) built-in. Therefore, all slot number fields should probably be 5 bits.
  - **Proposal: All slot numbers in DECT-2020 should be represented as 5 bits fields.**
- Size of bit-fields for carrier numbers:
  - Legacy DECT typically uses 5 bits in order to encode a carrier number. Carrier numbers are typically 0-9 (i.e. the 10 carriers in the DECT core band 1 880 MHz - 1 900 MHz). The 5 bit field typically used can encode for 32 carriers, and this is to allow for DECT operation in alternative RF bands (see ETSI EN 300 175-2 [i.2], Annex F). In legacy DECT, although carrier numbers themselves could be in this larger range, there are normally never more than 10 allowed carriers at a time.

- Since the DECT-2020 PHY is a wideband RF design there are probably some physical limits on the maximum number/range of RF carriers that the PHY can access (e.g. limited by the maximum size of the FFT). The size of bit field used to encode carrier numbers should be based off this limit, rather than absolute carrier numbers. It is proposed that the maximum bit field size for carrier umbers should be 4 bits (i.e. up to 16 carriers) or 5 bits (i.e. up to 32 carriers).

- **Proposal: All carrier numbers in DECT-2020 should be represented as 4 bit fields.**

- This study is mainly concerned with the size/length of information content for the beacon bearer. No attempt is made here at "mapping" this information into structure in the MAC payload (e.g. as fields or channels). However this "mapping" is performed, it needs to be efficient, flexible and extensible. The use of rigid, fixed bit-fields should be avoided where possible, in favor of more extensible coding schemes.

9 Design and analysis of basic access sequences and signaling procedures

9.1 Packet mode operation

9.1.1 General

Under "packet mode operation" the following cases are considered:

- Any signaling operations using RAC (Random Access Channels), including CC entity operations intended to setup scheduled services (but not the scheduled traffic bearers part of the procedure).
- Any U-plane traffic (e.g. WLAN traffic) using standard packet formats (long and short), including the possible use of MIMO.
- ULE traffic.

All these traffic types are processed in a common way. The initial packet of all these traffics has a common structure (format of the first slot in WLAN traffic).

In this study WLAN traffic is assumed to be "single channel". The case of WLAN traffic using multiple channels is possible and considered in PHY layer design. However, the MAC operation in this case has special peculiarities and so this topic is left for a further study.

- In particular a critical decision is if multicarrier WLAN traffic is allowed in the initial access or only after some acknowledgement from the other peer.

9.1.2 Principles and assumptions

After the acceptance of the model "A" as working assumption (see clause 7.1.3), it is possible to start the design of a common MAC access strategy for all packet mode traffic types.

The general principle is a preset slot classification (decided by the FP) as "preferred-downlink", "preferred-downlink" or "flexible" (see clause 7.1.3.5 and Figure 4). This pre-selection and the associated message/response sequence should be performed by the FP and announced in some way using the broadcast mechanism.

Figure 4 shows a possible example based on a 2,5 ms basic roundtrip delay and equally spaced "preferred" slots in both directions. This is not the only possibility and different values of the "basic roundtrip delay" are possible.

In this context, "basic roundtrip delay" means that this would be the basic delay in a two-way/two messages signalling operation.

9.1.3 Equally spaced vs. non equally spaced

9.1.3.1 General

The equally spaced "preferred" slots shown in clause 7.1.5 are just one of the possibilities. Other approaches are possible. For instance, placing the "preferred slots" together allows a longer "burst" for flexible slots. The convenience on one of other configuration depends on the expected traffic type:

- Equally spaced maps are convenient for signalling (RAC) and ULE traffics since give a balanced time for processing the data. This is convenient for signalling (especially high layer signalling) and ULE traffic.
- Not equally spaced mappings (in the limit, where both "preferred" slots placed together) are convenient if the expected traffic pattern is WLAN since it provides space for long frames. However signalling traffic may be less efficient since there may not be enough time at one of the peers to process the messages. This may typically require a wait interval at least, or NWK layer messages. Also scheduled traffic may be perturbed as result of the tendency by the FP to place scheduled traffic closed to the preferred bearers (see the rule in clause 7.1.4.3).

In short, equally spaced mappings are convenient for signalling and for scheduled services. Non equally spaced are convenient if the dominant traffic is WLAN (at the expense of adding delays of making more difficult the process of the others).

9.1.3.2 Example on non equally spaced mapping

See Figure 4 for associated legend.

Slots >	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24
carrier0																								
Carrier1																								
Carrier2																								
Carrier3																								
Carrier4																								
Carrier5																								
Carrier6																								
Carrier7																								
Carrier8																								
Carrier9																								

Figure 41: Example of non-equally spaced mapping

9.1.3.3 Example on wideband access (two step approach)

See Figure 4 for associated legend.

Slots >	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24
carrier0																								
Carrier1																								
Carrier2																								
Carrier3																								
Carrier4																								
Carrier5																								
Carrier6																								
Carrier7																								
Carrier8																								
Carrier9																								

Figure 42: Example on wideband access (two step approach)

9.1.4 The message/response sequence

The basic concept in the model "A" is that there is known relation between slots used for placing a message and slots where the response to the message will be received. This "sequence" is set by the FP and communicated to the PP by means of some part (to be designed) of the broadcast mechanism.

In the sketch described in clause 7.1 any random access message in uplink direction (RAC, WLAN or ULE) will be replied by the FP in the first "preferred downlink" slot. In general reply may be done in the same carrier, however this is not always possible (slotxcarrier may be in use by other service). Therefore special rules are needed. The same applies to potential wideband WLAN accesses. In any case this has to be solved by the broadcast mechanism.

9.1.5 Initial access (uplink)

9.1.5.1 RAC, ULE and WLAN single slot traffics

For any single slot traffic the best strategy would be a random selection of any of the possible slotxcarrier combinations in the interval before the next downlink "preferred" slot. By "possible" slotxcarrier it is meant any slot announced by the FP as usable, and assuming channel selection done by the FP, and announced via the broadcast mechanism. A random component should be used to spread the traffic. Several options are possible depending on the balance between access speed and collision avoidance. A last time scan of the selected carrier may be possible but it compromises the access speed.

The scan policy of the FP should also be taken into account. The FP may not being listening for setup in all slotxcarrier combinations. This has also to be part of the broadcast mechanism.

9.1.5.2 WLAN multi slot traffic

Similar strategy may be used by WLAN multi slot traffics, however the use of random component may take into account the expected length of the packet.

9.1.5.3 Response

The basic principle is that the PP should listen for a reception ACK in the expected slotxcarrier. If this is not the case a collision should be assumed.

9.1.5.4 Continuation after response

After successfully reception of a response, the PP may continue the transmission. Several strategies are possible:

1) A new random selection is done for the next uplink transmission.

2) The FP indicates the slotxcarrier where the transmission should continue.

3) A preferred uplink slot is used.

The three strategies each have pros and cons.

A special and interesting case happens when the process is repeated with a duration longer that a frame (or at least crossing the beacon position). In such cases, and assuming strategy 2), it would be possible to mark the selected slotxcarrier as non-usable in the broadcast channel selection aid information. In such a case the transmission becomes "semi-scheduled" and protected from further access collisions (from the same system). This is an option that may be further investigated.

9.1.5.5 Collision and erroneous responses

If no response is received, a collision may have happened. An appropriate algorithm with random components for a back-off time should be used to control the situation.

If partial responses are received (e.g. A-field well received, B-field not received), a retransmission or a new transmission with a lower MCS seems to be the right strategy expected.

The field A1 (if used) may be transmitted with a lower MCS and used as aid for selecting the proper MCS:

EXAMPLE: The Tx sends A0 with MCS=1, A1 with MCS=6 and B with MCS=10.

- If A0 is not received, then a collision may be expected. The proper collision handling algorithm should be run and the attempt repeated.
- If A0 is received but neither A1 nor B is received, the Tx should repeat the attempt (without collision handling back off) with a very conservative MCS, lower than 6. For example, A1 MCS=1 and B MCS=3.
- If A0 and A1 are received but B is not received, the Tx should repeat the attempt reducing B field MCS, but not lower than 6, since 6 works.

Similar strategy may be used by WLAN multi slot traffic, however the use of random component may take into account the expected length of the packet.

9.1.5.6 Wideband WLAN uplink transmissions

The PHY layer design supports multi-carrier transmissions with optimized subcarrier mappings (see ETSI TR 103 514 [i.26]). They are expected to be used in WLAN operation. The PHY design allows wideband initial access (long format). However, the convenience at MAC layer of allowing such access is debateable. A more conservative design would be a two-step approach where the transmitter always starts the transmissions as single-channel and should wait for at least one acknowledgement message to expand the transmission to wideband.

Another factor to be taken into account is the ability of the FP to be in "wideband" listen for setup state. While this may be an option in high-end FPs, the possibility does not seem to be an option for battery limited PPs. Therefore general use of unexpected wideband setups seems not to be a design option.

A realistic design would be allowing only "unexpected" single-carrier setups with the option to shift to wideband after first acknowledgement (with the permission to expand bandwidth indicated in the acknowledgement message). The FP capability to accept wideband setups may continue for negotiated interval. After then, only single carrier setups are allowed.

9.1.5.7 Limitations in FP setup capability

A FP may establish special limitations in order to avoid being in continuous listen for setup state. In legacy DECT, the listen for setup is limited to one carrier. DECT-2020 allows any configuration to be build, assuming that it can be announced via the broadcast mechanism.

9.1.6 Downlink transmissions - introduction to the problem

9.1.6.1 PP setup capabilities

The initial establishment of downlink transmissions is far more problematic due to the need for the PPs to be in listening for setup state. Continuous listening for setup over multiple carriers may seriously impact the battery duration. On the other hand a very "slow" response for downlink traffic (e.g. using normal DECT paging) will not be acceptable for WLAN traffic and would jeopardize technology performance. It may also produce a very unbalanced design with different response times depending on direction.

The proper and efficient handling of downlink setups is a promising area for technology differentiation.

The following options are envisioned (listed from higher to lower performance):

- The PP is in multi-carrier listen for setup and is able to accept even wideband WLAN access.
- The PP is able to accept only single-carrier setups, however it may be listening on two or more carriers (given options to the FP).
- The PP is only listening on one carrier and only accepts narrowband access.
- As above, but the PP only listens on some slots of the frame.
- The PP is not in listening for setup state, but can recognize the paging messages in the beacon.
- The PP is in a slow listening for paging state (such as ULE devices).

Due to the wide range of products to be implemented with the technology and also to the dynamic nature of the traffic, a dynamic system is proposed, where the PP may change from one state to another based on the following inputs:

- Previous activity.
- Messages over the paging channel.
- Messages over the bearers.
- A timer.

Proper coordination and communication between PP and FP is needed to make the FP aware of PP immediate setup capabilities.

9.1.6.2 Downlink access procedure

Assuming that the PP is in setup capability in the slotxcarrier selected by the FP, the access procedure seems to be similar to the uplink case.

MAC signalling may be different with potentially more FP control on the indications for returning the ACK and for further transmissions.

The collision problem in such transmissions is largely reduced (the FP is not going to collide) however there even exists the possibility of collision from other systems and from other portables in the same system if they attempt an uplink access over a "flexible direction" slot.

An additional case to be taken into account is the potential need for the FP to do access attempts to several PP at the same time. This may be supported due to the C/L nature of packet mode transmissions, but the proper discriminators are needed.

9.1.6.3 Example of possible MAC semantics for a downlink access

Case:

The FP has received higher layer for traffic for two PPs: a huge burst of WLAN traffic for one of them and only minor traffic for the other. Unfortunately both PPs are in listening for setup over the same slotxcarrier and the FP cannot simply delay the traffic due to slot map and delay requirements.

Solution:

A possible solution would be the FP to issue a RAC packet with MAC content addressed to both PP. This will, of course, impact higher MAC layer design. The packet would contain several MAC channels with the following semantics:

- Source identification (this may be common information).
- Destination identification 1, plus several MAC instructions addressed to PP 1.
- Destination identification 2, plus several MAC instructions addressed to PP 2.

The "several MAC instructions" may contain:

- Identification of the slotxcarrier to send the response that would be different for both PPs to avoid a return collision.
- Instructions about which setup detection state should be used after this acknowledgement. These instructions could also be different. For instance, PP 1 may be requested to be in listening for wideband setup, and the PP 2 in listening for single-carrier setup. The FP can assign the slotxcarrier positions appropriately to avoid overlapping in this new packet.

After reception of the ACKs, the FP may separate downlink setup attempts to both devices.

NOTE: In general, it is assumed that all packet mode messages would contain MAC messages, even if the transmission is WLAN traffic.

9.1.7 Operation under low traffic conditions

Under low traffic conditions the problem of collisions and slot usage by other devices does not exist (or is limited). Therefore sequences of WLAN transmissions and signalling operations happen at the roundtrip cycle defined by the model "mode". With the example of model A given in clause 7.1.5, this roundtrip time is 2,5 ms and this is the basic timing for signalling operation.

The implementation may add additional delays due to processing considerations. For instance, with the example given in clause 9.1.3.2 of "uneven" placement of preferred bearers, it is unlikely that one of the sides may process the messages using only the inter-slot time. This would result in increased delay.

The main critical problem under low traffic conditions is the fast selection of the proper MCS and MIMO configuration, and the reconfiguration in case of errors in the setup messages. Plain ARQ or HARQ with the same MCS and MIMO is not a good strategy if such configuration has not been previously used successfully. In general the MCS/MIMO configuration used in a first message has to be conservative, but erroneous reception may still happen. The proper algorithms, timing and strategies for reducing MCS and MIMO are required. The field A1 may be used as a "tester" to speed up selection of MCS.

Conversely, when a transmission with a given MCS and MIMO is running well, the Tx may choose to try a higher MCS and MIMO configuration. Again, the proper dynamic algorithms are needed.

9.1.8 Operation under high traffic conditions

Additional problems will arise when the system is operated under high traffic conditions. These problems are a consequence on the competition for recourses between the different devices active in a cell and will range from restriction in usable combinations of slotxcarriers to the handling of potential access collisions. The effective control of all phenomena and effects under high traffic conditions will add additional complexity to the MAC layer.

The topic is complex enough and will require detailed study in a further project phase. At this stage of the design, the present document lists and identifies some of the issues that will arise under heavy traffic situations. All of them should be properly studied and solved in the MAC architecture phase:

- Reduction in the number of access slotxcarriers due to existing services increasing access collision probability.
- Strategies for prevention of access collisions.
- Strategies for resolution of access collisions.

- Competition for the return channels - case of having several RAC transmissions allocated to the same return channel.
- Blind slot due to scheduled services and communication to PPs - via beacon broadcast or other mechanisms.
- Long duration packet mode transmissions - possibility and convenience to reserve resources to prevent access collisions - broadcast.
- Strategies for channel scanning, scanning cycle and last minute scan to detect used resources by RAC/ULE/WLAN transmissions.
- Case of PPs unable to observe the "fast" part of the beacon and their possible actions.

As noted, all these cases are left for further study during the MAC architecture phase.

10 Revised Physical layer formats

10.1 General

As result of the progress in the MAC area, some Physical layer formats have been revised. The present clause provides an update on current understanding. This clause can be considered a potential change request to the PHY layer (ETSI TR 103 514 [i.26]).

10.2 Formats for packet-mode non-scheduled services

10.2.1 Overview

The following formats are included under this category:

- Format for RAC (Random Access Channels) - mostly intended for signalling operations.
- Formats for ULE packets.
- Formats for WLAN traffic over single-carrier channels.
- Formats for WLAN traffic over multi-carrier channels.

MIMO is usable in all cases.

10.2.2 Basic principles and basic changes

10.2.2.1 General

The following common principles and changes are proposed:

1) New common design framework.
2) Packets are now aligned to slot grid (with half slot resolution).
3) Reduced unified inter-slot space.
4) Common position of inter-slot boundaries and new time reference.
5) Reduced STF field.
6) No duplication of CTF field.
7) Possibility of inserting some "A1 slots" with reduced MCS in the initial slot of the packets.
8) Simplified MIMO training fields.

9) The length of long bursts of WLAN packets does not need to be known at the beginning.

10.2.2.2 New common design framework

A common design framework is created for all cases of the category. The different uses (ULE, RAC, WLAN) are just A common design framework is created for all cases of the category. The different uses (ULE, RAC, WLAN) are just particular configurations of the common design.

This simplifies the design of the receivers with a common initial section of all packets. In addition, it adds flexibility and opens the door to any intermediate solution, e.g. ULE packets with MIMO.

New design:

Three new fundamental packet formats (with full slot length) are created. All existing packets are considered composed of these fundamental building elements.

Additional variants with half slot length are also proposed.

The new packets are:

- Common initial (I) packet.
- Immediate continuation (C) packet.
- Open continuation (O) packet.

Relationship with previous packets:

The relationship with the previous packets proposed in ETSI TR 103 514 [i.26] is as follows:

- A former "long format" is now built with one I packet and a variable number of C packets.
- A former "short format" is now built with one O packet and a variable number of C packets.
- RAC and ULE are typically implemented with an I packet. However they may also be implemented (new possibility) with an O packet is they are used in the same carrier of a previous transmission with last activity less that a given time.

The framework is completely flexible and not constrained to the examples. For example, a MIMO packet may include continuation (C) packets if needed.

Alternative design:

Nevertheless, it is recognized that the previous approach of long and short formats with common coding for the whole burst of B symbols has advantages at coding level (however at the expenses of several issues). It is considered that it is not possible to select definitively one or the other approach at this stage of the project. Therefore a redesigned version of the "old" approach of long/short packet is also provided in the present document. These are the L and S formats that are described in clause 10.2.3.4. The selection between one or the other approach would have to be taken in a further project stage.

10.2.2.3 Packets are now aligned to slot grid (with half slot resolution)

While the arbitrary length (with symbol resolution) of the long/short formats proposed in ETSI TR 103 514 [i.26] may have certain advantages, it is very inconvenient for MAC design (e.g. consider ARQ). Therefore, the arbitrary length long/short formats are now split in sections with full and half slot resolutions. There is no real PHY change for any packet aligned to slot or half-slot boundaries. However arbitrary position/length packets are no longer allowed.

10.2.2.4 Reduced unified inter-slot space

Proposed inter-slot space in ETSI TR 103 514 [i.26] "standard packets format" is considered excessive. It is also variable (long, short and RAC formats had different inter-slot space). This cannot be technically justified.

The inter-slot space is now unified to 4/9 T_{SYM} in all cases.

T_{SYM} is = to 41,67 µs (72 units at nominal sampling). Therefore 4/9 T_{SYM} is = 18,52 µs (32 units).

NOTE: If this value were considered too small for certain applications, the inter-slot space can be expanded, if needed, at the expense of the STF (still very large).

10.2.2.5 Common position of inter-slot boundaries and new time reference

The position of inter-slot boundaries is unified for all cases to simplify implementations. Slot time reference is now set at the beginning of the inter-slot space.

10.2.2.6 Reduced STF field

The STF field proposed by ETSI TR 103 514 [i.26] is considered excessive. The new length is set to 14/9 T_{SYM} for all initial random access packets, and to 5/9 T_{SYM} for the "S" format (formerly named short format) and for the "open continuation packet".

NOTE 1: The value of the STF field is still considered conservative and further reductions may be possible in the future.

NOTE 2: Also suggested is the possibility to move part of the STF field (e.g. by 1/9 T_{SYM}) to extend the CP of the following CTF field.

NOTE 3: To compare these values with the preambles in "legacy DECT", each T_{SYM} = 41,7 µs = 48 DECT symbols = 48 GFSK bits. Therefore, assuming the same roll-off factor, the proposed STFs equals to 74,7 "DECT bits" (long) and 27,7 (short) respectively. With ideal roll-off (un-implementable in practice) the values would be T_{SYM} = 72 "ideal bits", long STF = 112 "ideal bits", short STF = 40 "ideal bits". With a more realistic roll off = 0,3 the values would be T_{SYM} = 55,4 bits, long STF = 86,2 bits, short STF = 30,8 bits.

NOTE 4: Previous calculation also shows that the preambles are still very conservative.

10.2.2.7 No duplication of CTF field

Duplication of CTF field proposed in ETSI TR 103 514 [i.26] is considered too conservative and excessive. CTF is no longer duplicated.

NOTE: Also suggested is the possibility to move part of the STF field (e.g. by 1/9 T_{SYM}) to extend the CP of the following CTF field.

10.2.2.8 Possibility of inserting some "A1 symbols" with reduced MCS in the initial slot of the packets

Optionally, a small number of symbols (e.g. from 0 to 3) after A0 may be transmitted with and independent coding and a more robust MCS than the B field. This is done to further support ULE and signalling operation, as well as early detection of feasibility of the MCS setting.

The value of the MCS for A1 is open to further investigation. The simple option is using the same MCS as A0. However a more promising solution would be coding it with a reduced MCS in relationship with the MCS used for the B field and defined by a table in the standard (e.g. "x" MCS steps less than B field).

The use of MIMO (or not) in A1 is noted as a design possibility.

10.2.2.9 Simplified MIMO training fields

Using up to 5 symbols for MIMO training (as proposed in ETSI TR 103 514 [i.26]) is considered excessive and not fitting with the frequency axis resolution provided by a 27 kHz subcarrier spacing. A single symbol is considered enough. In case of continuation formats (formerly "short format"), the same initial CTF is assumed to be able to keep MIMO tracking. This is supported by the comparison with the HE formats described in the same ETSI TR 103 514 [i.26] that can operate with a single CTF.

10.2.2.10 The length of long bursts of WLAN packets does not need to be known at the beginning

Current design assumes that total length is coded in the PHY header. This is considered inconvenient for long burst when end of the burst is not known at beginning of transmission. A new design is proposed where a MAC header field is supposed to exist at the beginning of each "continuation" packet (see clause 10.3.2) indicating the termination of the burst. This is also saves bits in the initial header field (A0).

10.2.3 Packet formats and diagrams

10.2.3.1 Full slot packets

10.2.3.1.1 Void

This clause is intentionally left blank.

10.2.3.1.2 Common initial (I) packet

This format is common to all RAC, WLAN and ULE packet sub-formats.

Structure:

NOTE 1: Time reference is the beginning of the inter-slot space.

4/9 T_{SYM} = inter-slot guard.

14/9 T_{SYM} = STF.

1 T_{SYM} = CTF.

1 T_{SYM} = A0 (formerly HF).

0-1 T_{SYM} = CTFM = MIMO training CTF.

NOTE 2: Not always included. Used only if MIMO. One symbol is assumed to be enough for all MIMO cases (using alternate subcarriers for each space stream).

NOTE 3: The position of this field is open. It can be before the A1 or after the A1. In the first case, A1 may use MIMO.

0-3 T_{SYM} = A1.

NOTE 4: Not always included. Used only as option in some packet types. The number of A1 symbols is assumed to be within 0 and 3.

N2 T_{SYM} = B.

NOTE 5: The number of B symbols that is necessary to complete the full-slot length of the packet.

EXAMPLE 1: For a packet with no A1 symbols and no MIMO, B = 6.

EXAMPLE 2: For a packet with 2 A1 symbols and MIMO, B = 3.

NOTE 6: There is the possibility to use part of the subcarriers in the CTF1 symbol to code additional critical information in addition to the A0 symbol.

Packet diagram:

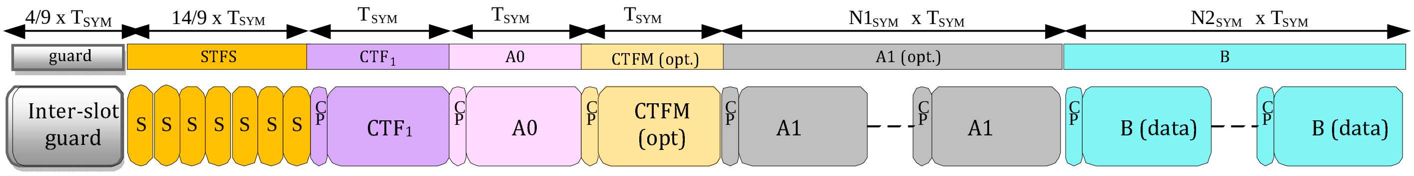


Figure 43: Common initial (I) packet

10.2.3.1.3 Immediate continuation (C) packet

This packet is to be used only immediately after a common initial (I) packet, an open continuation (O) packet, or another continuation (C) packet (on the same carrier(s)).

O packets do not allow to change the MCS or the MIMO setting that should be as defined by the previous I packet.

Structure:

10 T_{SYM} = B

NOTE: There is no guard interval.

Packet diagram:

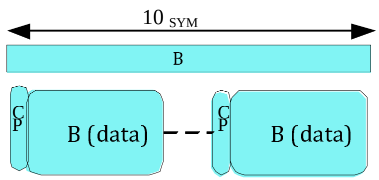


Figure 44: Immediate continuation (C) packet

NOTE 1: There is neither inter-slot space nor header (A0) field.

NOTE 2: There is no PHY layer header.

NOTE 3: However, a MAC header field is assumed to exist in the first B symbol. This header may indicate the MAC packet number and the end/continuation of the burst.

NOTE 4: MCS for the B field and MIMO configuration are assumed to be the same previously set by the previous I packet.

NOTE 5: There will a limit of the maximum length in slots of a C burst due to tracking considerations. An O packet should be used is this limed is exceeded.

10.2.3.1.4 Open continuation (O) packet

This packet is to be used when there has been a previous transmission on the same carrier and within a maximum limit after the end of previous I, O or C packet. This limit is for further study (but it will be less than one frame in any case).

Structure:

NOTE 1: Time reference is the beginning of the inter-slot space.

4/9 T_{SYM} = inter-slot guard

5/9 T_{SYM} = STF

1 T_{SYM} = CTFM

NOTE 2: With MIMO tracking capability. It is assumed to be enough even with MIMO by comparison with HF format).

8 T_{SYM} = B

Packet diagram:

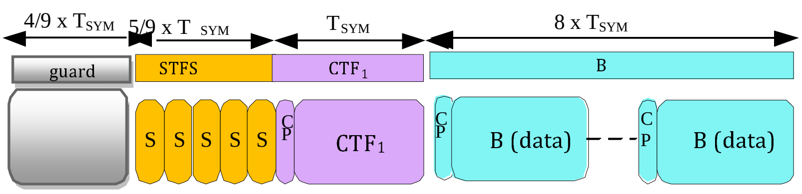


Figure 45: Open continuation (O) packet

NOTE 1: There are no A0 nor A1 fields.

NOTE 2: There is no PHY header.

NOTE 3: However, a MAC header field is assumed to exist in the first B symbol. This header may indicate the MAC packet number and the end/continuation of the burst.

NOTE 4: MCS for the B field and MIMO configuration are assumed to be the same previously set by the header of a last previous I packet.

NOTE 5: There is the possibility to use part of the subcarriers in the CTF1 symbol to code some critical information (normally placed in the A0 symbol).

10.2.3.2 Half slot packets

10.2.3.2.1 Void

This clause is intentionally left blank.

10.2.3.2.2 Initial packet - Half slot (IH)

This format is only allowed in RAC channels as an isolated packed (i.e. no continuation). If the intention is continuing the burst, then use an initial packet full slot and then a continuation packet (either half or full).

Structure:

NOTE: Time reference is the beginning of the inter-slot space.

4/9 T_{SYM} = inter-slot guard

14/9 T_{SYM} = STF

1 T_{SYM} = CTF

1 T_{SYM} = A0 (formerly HF)

1 T_{SYM} = B

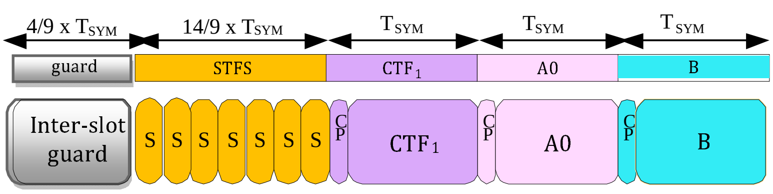


Figure 46: Initial half-slot packet (IH)

NOTE 1: There is only one B symbol. Due to the use of the packet, it is assumed to be transmitted with a robust MCS.

NOTE 2: This format is only allowed in RAC channels as an isolated packed (i.e. no continuation). If the intention is continuing the burst, then use an initial packet full slot and then a continuation packet (either half or full).

NOTE 3: MIMO training is not allowed.

NOTE 4: A1 is not allowed.

NOTE 5: This packet allows carrying only one B symbol. However it also trains the channel, so it may allow an O or OH packet if the next transmission.

10.2.3.2.3 Immediate continuation packet - half slot (CH)

This packet is intended to align bursts with the half slot grid. It can only be used after an I, O or C packet and should be the last one in the burst. Otherwise, the C packet should be used.

Structure:

5 T_{SYM} = B

NOTE: There is no guard interval.

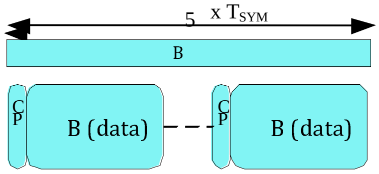


Figure 47: Immediate continuation half slot packet (CH)

NOTE 1: There is neither inter-slot space nor header (A0) field.

NOTE 2: However a MAC header field is assumed to exist in the first B symbol. This header may indicate the MAC packet number and the end/continuation of the burst.

NOTE 3: MCS for the B field and MIMO configuration are assumed to be the same previously set by the previous I packet.

10.2.3.2.4 Open continuation packet - half slot (OH)

This format is only allowed in RAC channels as an isolated packed (i.e. no continuation). If the intention is continuing the burst, then an open continuation packet full slot should be used and then a continuation packet (either half or full).

This packet is to be used when there has been a previous transmission on the same carrier and within a maximum limit after the end of previous I or C or CH packet. This limit is for further study (but it will be less than one frame in any case).

MIMO is not usable.

This packet is to be used when there has been a previous transmission on the same carrier and within a maximum limit after the end of previous I or C packet. This limit is for further study (but it will be less than one frame in any case).

Structure:

NOTE: Time reference is the beginning of the inter-slot space

4/9 T_{SYM} = inter-slot guard

5/9 T_{SYM} = STF

1 T_{SYM} = CTFM

NOTE: With MIMO tracking capability. It is assumed to be enough even with MIMO by comparison with HF format.

3 T_{SYM} = B

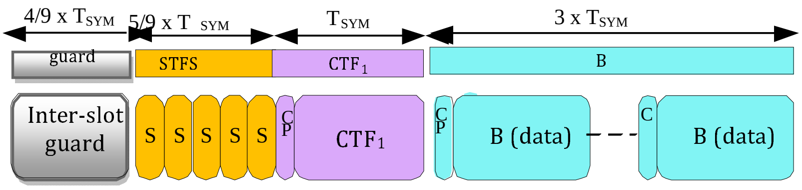


Figure 48: Open continuation half packet (OH)

NOTE 1: There are no PHY header (A0) nor A1 fields.

NOTE 2: However a MAC header field is assumed to exist in the first B symbol. This header may indicate the MAC packet number and the end/continuation of the burst.

NOTE 3: MCS for the B field and MIMO configuration are assumed to be the same previously set by the header of a last previous I packet.

10.2.3.3 Use of packet formats

The following rules summarize the use and allowed combinations of packets in a burst.

Assuming that an Rx is listening in a given slotxcarrier (see note 1):

1) the Rx should be able to receive I packet in any case;

2) the Rx should be able to receive a C packet only immediately after another I, O or C packet (notes 2, 3);

3) the Rx should be able to receive an O packet only if there has been a previous transmission of an I, O or C packet on the same carrier during the last N_{MAXO} slots (note 4);

4) there is a limit for the maximum length of a burst (N_{MAXB}) (note 5);

5) half-slots, when used in a multi-slot burst, should be placed at the end of the burst and there should be only one.

NOTE 1: This is the scanning strategy problem, a different topic.

NOTE 2: A concatenation of an I or O packet immediately followed by one or several C packets is named a "burst".

NOTE 3: The case of erroneous reception of a packet in a burst has to be taken into account. For example, in a burst of 3 packets, if packet 1 is received, packet 2 is corrupted, packet 3 should be received.

NOTE 4: The value of N_{MAXO} (the separation between the end of a burst and an open continuation packet) is left for further study but it should not be more than 23 slots in any case.

NOTE 5: The value of N_{MAXB} (the maximum length of a burst) is left for further study but it should not be more than 23 slots in any case.

10.2.3.4 Alternative design (L-S approach)

10.2.3.4.1 General

The alternative design, also known as "L-S approach" is documented as a second design option. This is a redesigned version of the "old" approach of long/short packets. It is recognized that this approach has advantages at coding level (due to common coding for the whole burst of B symbols), however at the expenses of several fundamental or implementation issues.

The L-S format design approach incorporates the optimizations in preambles done for the I-C-O format. It also implements the quantification of the burst length to half-slot units. The main difference between one and the other approach is that channel coding (of the B symbols) is done as a whole in the L-S format and individually (slot by slot) in the I-C-O approach. Because of that, the length of the burst has to be known before starting transmission and can be coded in the A0 field (L format) or in a way not yet decided in the S format.

10.2.3.4.2 New long (L) packet

This format is to be used in any initial transmission of RAC, WLAN and ULE services. It codes a single burst of variable length, processed with a common coding schema. The length of the burst has to be known before starting transmission and may be coded in the A0 field. The theoretical minimum length of the burst is a half slot (however full slots is assumed to be the minimum length in most of the cases) and there will be a maximum length, not yet decided. Total length is quantized in half slot units.

Structure:

NOTE 1: Time reference is the beginning of the inter-slot space.

4/9 T_{SYM} = inter-slot guard

14/9 T_{SYM} = STF

1 T_{SYM} = CTF

1 T_{SYM} = A0 (formerly HF)

0-1 T_{SYM} = CTFM = MIMO training CTF.

NOTE 2: Not always included. Used only if MIMO. One symbol is assumed to be enough for all MIMO cases (using alternate subcarriers for each space stream).

NOTE 3: The position of this field is open. It can be before the A1 or after the A1. In the first case, A1 may use MIMO.

0-3 T_{SYM} = A1.

NOTE 4: Not always included. Used only as option in some packet types. The number of A1 symbols is assumed to be within 0 and 3.

N2 T_{SYM} = B

NOTE 5: The number of B symbols that is necessary to complete the desired length of the packet.

EXAMPLE 1: For a 2 full slot length packet with no A1 symbols and no MIMO, B = 16.

EXAMPLE 2: For a 3 1/2 full slot length packet with 1 A1 symbols and no MIMO, B = 30.

EXAMPLE 3: For a 4 1/2 full slot length packet with 3 A1 symbols and MIMO CTFM, B = 37.

Packet diagram:

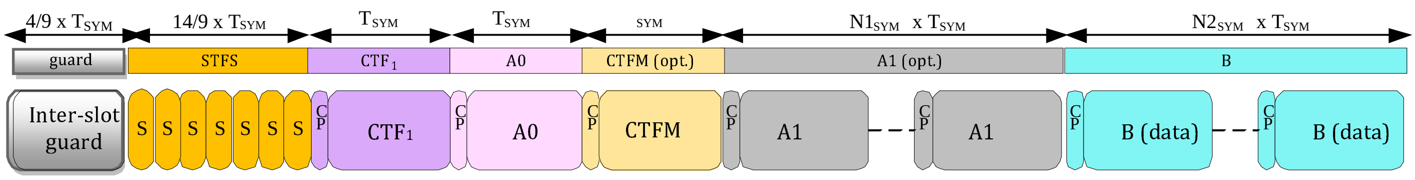


Figure 49: New long (L) packet

10.2.3.4.3 New Short (S) packet

This packet is to be used when there has been a previous transmission on the same carrier and within a maximum limit after the end of previous L or S packet. This limit is for further study (but it will be less than one frame in any case).

The length of the burst has to be known before starting transmission and should be coded at the beginning of the packet in a way not yet decided. Some potential ideas are:

- Using the CTF symbol.
- Using the STF symbol.
- Adding a header symbol (as "old" design).
- Use a fixed length identical to the previous L packet. If this has to be changed, an L packet should be used.

As with O packets, S packets do not allow to change the MCS or the MIMO setting that should be as defined by the previous L packet.

Structure:

NOTE 1: Time reference is the beginning of the inter-slot space

4/9 T_{SYM} = inter-slot guard

5/9 T_{SYM} = STF

1 T_{SYM} = CTFM

NOTE 2: With MIMO tracking capability. It is assumed to be enough even with MIMO by comparison with HF format).

N3 T_{SYM} = B

NOTE 3: The number of B symbols id the necessary to complete the desired length of the packet.

EXAMPLE 1: For a full slot length packet, B = 8

EXAMPLE 2: For a 2 full slot length packet, B = 18

EXAMPLE 3: For a 4 1/2 full slot length packet, B = 43

Packet diagram:

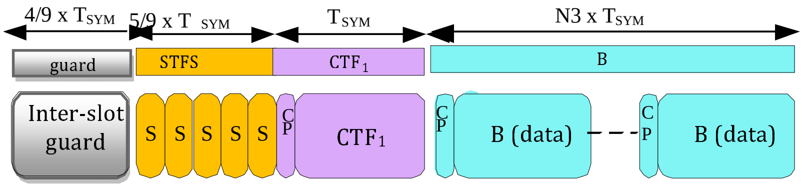


Figure 50: New Short (S) packet

NOTE 4: There are no A0 nor A1 fields.

NOTE 5: There is no PHY header.

NOTE 6: MCS for the B field and MIMO configuration are assumed to be the same previously set by the header of a last previous L packet.

10.3 Formats for scheduled services

10.3.1 Discussion and principles

10.3.1.1 General

ETSI TR 103 514 [i.26] defined several candidate packet formats for scheduled services. The general common format is defined in clause 6.3.2.2.2.1 and the list of candidate packet types is defined in clauses 6.3.2.2.2.2 (full slot formats) and 6.3.2.2.2.3 (half slot formats) of ETSI TR 103 514 [i.26]. All of them are named "HE" (for High Efficiency), however only formats 1 and 2 for full slots and the only format for half slot are "true High Efficiency".

HE formats were created during ETSI TR 103 514 [i.26] work. The level of maturity of these formats is different and several important pending issues were left for further study at the completion of ETSI TR 103 514 [i.26]. These pending issues impact moistly to the "true High Efficiency" format (types 1 and 2 less in full slot list and type HS1 for half slots). The main issues were related to the feasibility of the use of a reduced CP of 2,3 µs and the feasibility of using those formats in single-carrier channels.

On the other hand the formats with 4,3 µs CP are considered far more mature due to the reuse of a basic design already proven (at simulation level) in the standard packet formats.

Therefore priority will be given to the completion of the design of the 4,3 µs CP (despite their relative inefficiency).

As second priority, some proposals will also be given for the "true High Efficiency" formats, however additional physical level work (simulations) is required.

10.3.1.2 Packet variants defined by ETSI TR 103 514

The following variants were defined in ETSI TR 103 514 [i.26]:

Table 15: Full-slot HE packet variants in ETSI TR 103 514 [i.26]

Variant	Total symbols	Data symbols	Min BW	Pilot CP		Data CP		Inter-slot guard time (GF)		Notes
			(MHz)	(µs)	units	(µs)	units	(µs)	units	
1	10	9	3,456	2,3	4	2,3	4	23,1	40	1, 6
2	10	9	3,456	9,3	16	2,3	4	16,2	28	6
3	9	8	1,728	18,5	32	4,6	8	27,8	48	6
4	10	9	1,728	4,6	8	4,6	8	0	0	2, 3, 6
5	8	7	1,728	37,0	64	9,3	16	18,5	32	6
6	9	8	1,728	9,3	16	9,3	16	0	0	2, 3, 6
A1	10	9	1,728	4,6	8	3,5	6	10,4	18	6
A2	9	8	1,728	18,5	32	4,6	8	27,8	48	6
A2a	10	9	1,728	4,6	8	4,6	8	0	0	2, 3, 5, 6
A3	9	7	1,728	6,9	12	6,9	12	20,8	36	4, 6, 7

NOTE 1: This type is listed since it was the first HE type proposed. Based on later studies, it may probably be removed and replaced by types 2 or A1 in all cases.
NOTE 2: These types are designed for multi-slot transmissions using a burst of consecutive slots over the same carrier.
NOTE 3: When transmitting over multiple consecutive slots, it is assumed that a Reference Symbol (RPF) has to be inserted in each slot, however this is subject to further investigation and optimization in later stages.
NOTE 4: This type shares the data symbol CP with types B A1, B A2 and BA3 proposed for beacon and Random Access Channel (RAC) bearers. See clause 6.3.2.2.4.
NOTE 5: This type will start and will end 27,8 µs before the time reference This ensures continuation with a previous type 2 or 2A slot and also a Guard interval of 27,8 µs at the end.
NOTE 6: Times are also given in "units". Each unit equals to 0,5787037 µs, which corresponds to 1 sample at 1,728 MHz or two samples at 3,456 MHz. Durations in "units" are exact values while the figures in µs have been rounded to the closest decimal.
NOTE 7: In this configuration the reference symbol (RPF) is duplicated over the two first symbol intervals. Therefore 7 data symbols can be transmitted.

Table 16: Half-slot HE packet variants in ETSI TR 103 514 [i.26]

Variant	Total symbols	Data symbols	Min BW	Pilot CP		Data CP		Inter-slot guard time (GF)		Notes
			(MHz)	(µs)	units	(µs)	units	(µs)	units	
HS1	5	4	3,456	2,3	4	2,3	4	11,6	20	See note
NOTE: Times are also given in "units". Each unit equals to 0,5787037 µs, which corresponds to 1 sample at 1,728 MHz or two samples at 3,456 MHz. Durations in "units" are exact values while the figures in µs have been rounded to the closest decimal.										

10.3.1.3 Additional issues are ideas already identified

During TC DECT discussions, the convenience of unifying the formats with the existing packet-mode formats, when possible, was identified as a mechanism to simplify implementations. It was also noted the need of solving the format for initial transmissions under scheduled channels (not solved in ETSI TR 103 514 [i.26]). Both issues are addressed in the solution that will be proposed for 4,3 µs CP packets.

10.3.2 Formats with 4,3 µs CP

10.3.2.1 General

The formats 3 and 4 (see Table 15) cover the cases of operation with the nominal CP of 4,3 µs.

Format type 3 is the initial packet of a burst and includes an inter-slot guard of 27,8 µs plus additional time for the CP in the first symbol (18,5 µs). These extra allowances may be administered for different physical layer uses if needed, without impacting the MAC design. For example, a small STF (Synchronization training field) can be inserted at the expenses of the extra CP in 1st symbol and/or the inter-slot guard.

The operation of the CTF symbol and the format in general is very similar to the open continuation packet (type O) already defined for packet-mode operation. This packet carries 8 data symbols.

Format type 4 is the continuation format when using 4,3 µs CP, and is very similar to the immediate continuation format (type C) already defined for packet-mode operation.

10.3.2.2 Proposal of re-design

Based on previous discussion, this study proposes the unification of formats with existing formats for packet mode operation. This implies adding a small STF of 5/9 after inter-slot guard in the previously named type 3 format that will become identical to the Open continuation (type O) format.

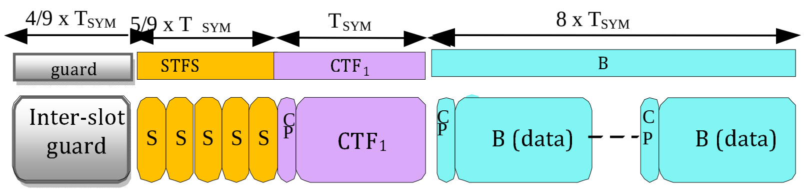
The case of initial transmissions in scheduled service carriers will be solved using the initial transmission format (type I) of packet mode services.

In addition to that, it is proposed to allow the use of half-slot formats in scheduled transmissions with nominal (4,3 µs) CP. Therefore, a total of 6 types of slot formats for scheduled services will exist, all of them identical to the packet mode formats.

Slot formats definitions are given in clause 10.2.3 and are not repeated here. The convention of considering the inter-slot guard to happen at the beginning of the slot is also followed here.

10.3.2.3 Use of the slot types - full slot and multiples of full slot transmissions

- For single slot full-slot scheduled transmissions over scheduled channels, the O type format given in clause 10.2.3.1.4 will be used.



4/9 T_{SYM} = inter-slot guard = 18,52 µs.
5/9 T_{SYM} = STF.
1 T_{SYM} = CTFm.
(with MIMO tracking capability. it is assumed to be enough even with MIMO by comparison with HF format).
8 T_{SYM} = B.

Figure 51: Packet for single slot full-slot scheduled transmissions

- For the special case of initialization of the transmission, the Initial (I) format defined in clause 10.2.3.1.2 will be used once, or after first acknowledgement, then O type will be used.

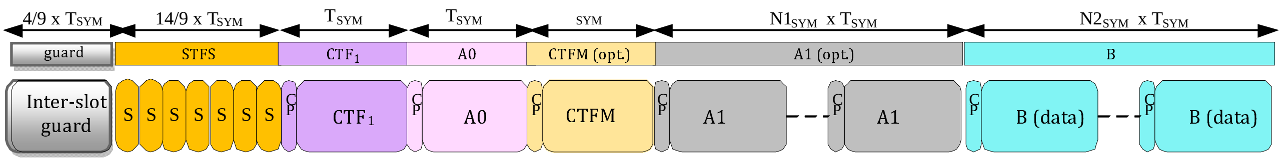


Figure 52: Packet for initialization of full-slot scheduled transmissions

- In case of scheduled transmissions over two or more slots, then the C type defined in clause 10.2.3.1.3 will be used.

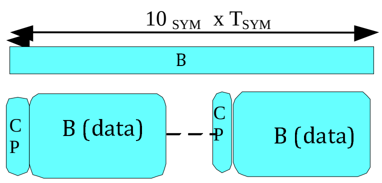


Figure 53: Packet for additional slot in multi-slot scheduled transmissions

10.3.2.4 Use of the slot types - half slots and odd multiples of half slot transmissions

It is proposed to also allow the use of the half-slot variants defined in clause 10.2.3.2 to create half-slot transmissions and multi-slot burst with size not multiple of full slot. This will be done as follows:

- Single slot half-slot scheduled transmissions are allowed. They will be typically used for acknowledgement or signalling channels. When used the format will be identical to the "Open continuation packet - half slot (OH)" defined in clause 10.2.3.2.4, except in the first transmission.
- For the special case of initialization of such transmissions, the "Initial packet - Half slot (IH)" defined in clause 10.2.3.2.2 will be used once, or after first acknowledgement, then OH type will be used.
- For the case of multi-slot bursts with size not multiple of full slot, the packet type "Immediate continuation packet - half slot" (CH) will be used in last position of the burst to match the desired burst size in half-slot units.

The rules for various combinations given in clause 10.2.3.3 is applicable with the difference that the combinations in the first column (bursts starting by I or IH) will only be used in case of initialization of the transmission.

10.3.2.5 Alternative design (L-S approach)

The alternative design described in clause 10.2.3.4.1 can also be used in scheduled transmissions. With this approach all "normal transmission" packets would be S format (see clause 10.2.3.4.3) of different sizes always multiple of half slot. The initialization packet would be "L" format.

The main practical difference between both approaches is that multi-slot packets would be a single unit for the purposes of channel coding and ARQ in the L-S approach and multiple units in the I-C-O approach. This has some advantages in terms of coding at the expenses of several drawbacks for ARQ and codec processing delay.

Note that both approaches converge in the case of using only single slots as bearers for the scheduled services.

10.3.2.6 Conclusion

With these rules a relatively simple and convenient solution is provided for scheduled transmissions with 4,3 µs CP, that now includes half slot granularity. It is foreseen that this solution will be implemented in the first release of normative standards.

10.3.3 Formats with reduced CP ("true High Efficiency formats")

10.3.3.1 Discussion and general

The use of reduced CP variant was agreed as design target for "true High Efficiency" scheduled transmissions, including for efficient transmissions of U-plane data over half slots. A design target of 9 usable data symbols in full slots (single slots) and 4 usable data symbols in half slots was established. Formats 1 and 2 from Table 15 are based on this design target.

Unfortunately, there were several doubts and unresolved topics that could not be closed at the closure of ETSI TR 103 514 [i.26]. Perhaps the most interesting was the feasibility of transmissions over single carriers with different views in the expert panel. Another, but related topic, was the feasibility of proper pulse conformation (pulse shaping) and the impact of the windowing in the remaining effective CP. All these issues should be tested and resolved with additional simulation before using these slot types.

The limitation of not being able to operate over single-carriers is considered unacceptable from MAC point of view. Because it would cause the need of finding convenient pairs of carriers free for use in the proper position of the grid and this cannot be guaranteed, and when possible, introduces constrains to other devices. Also, except for certain high bit rate services, it goes in opposite direction to the high efficiency, initial design goal of the format. Therefore the 2,3 µs CP types (types 1 and 2 and HS1) will not be pursed until further simulation is available.

10.3.3.2 New proposals

Since 2,3 µs CP may be too short, the design proposal would be slightly increasing the CP to compensate for the time consumed by the pulse shaping at the expenses of the inter-slot guard (still conservative) used by type 1.

The following new options are given in Table 17.

Table 17: Additional options for full-slot HE packet variants

Variant	Total symbols	Data symbols	Min BW	Pilot CP		Data CP		Inter-slot guard time (GF)		Notes
			(MHz)	(µs)	units	(µs)	units	(µs)	units	
A1a	10	9	1,728	3,5	6	3,5	6	11,5	20	
A1b	10	9	1,728	2,9	5	2,9	5	17,33	30	
A1c	10	9	1,728	4,6	8	2,9	5	15,6	27	

Type A1b seems the most promising since uses an inter-slot guard near identical to 4,3 µs packets. Type A1c may be used if additional CP or needed in the first symbol. Type A1a will need to be used if the 2,9 µs CP is found not feasible after simulation.

For Half Slot variants, no other alternative to existing format HS1 can be proposed due to the critical inter-slot guard time.

10.3.3.3 Conclusion

Nevertheless, only the 4,3 μs CP formats are considered as presumed feasible at this stage. Other formats may require additional physical layer work and may not be ready for the first technology release.

In any case, from MAC perspective, the design of procedures for setting and handling scheduled services will take into account the future addition of alternative packet formats and will reserve the expected coding capability.

Annex A: MCS Parameters

A.1 General

Table A.1 to Table A.42 contain rate-dependent parameters for supported bandwidths and number of spatial streams (N_{SS}). Data rates figures represent rates within the Data Field of the packet in kbps.

The figures are based on the Long Preample packet (as given in ETSI TR 103 514 [i.26] and in figure 12 of clause 7.9.2) without optimizations. The several optimizations described in clause 10 of the present document may improve these figures.

A.2 MCS parameters for 0,864 MHz

Table A.1: MCSs for 0,864 MHz, N_{SS} = 1

MCS	Modulation	R	N_{BPSC}	N_{SD}	N_{SP}	N_{CBPS}	N_{DBPS}	Data rate
0	BPSK	1/2	1	20	4	20	10	240
1	QPSK	1/2	2	20	4	40	20	480
2	QPSK	3/4	2	20	4	40	30	720
3	16-QAM	1/2	4	20	4	80	40	960
4	16-QAM	3/4	4	20	4	80	60	1 440
5	64-QAM	2/3	6	20	4	120	80	1 920
6	64-QAM	3/4	6	20	4	120	90	2 160
7	64-QAM	5/6	6	20	4	120	100	2 400
8	256-QAM	3/4	8	20	4	160	120	2 880
9	256-QAM	5/6	8	20	4	160	-	-
10	1024-QAM	3/4	10	20	4	200	150	3 600
11	1024-QAM	5/6	10	20	4	200	-	-
12	BPSK	1/4	1	20	4	20	5	120

Table A.2: MCSs for 0,864 MHz, N_{SS} = 2

MCS	Modulation	R	N_{BPSC}	N_{SD}	N_{SP}	N_{CBPS}	N_{DBPS}	Data rate
0	BPSK	1/2	1	20	4	40	20	480
1	QPSK	1/2	2	20	4	80	40	960
2	QPSK	3/4	2	20	4	80	60	1 440
3	16-QAM	1/2	4	20	4	160	80	1 920
4	16-QAM	3/4	4	20	4	160	120	2 880
5	64-QAM	2/3	6	20	4	240	160	3 840
6	64-QAM	3/4	6	20	4	240	180	4 320
7	64-QAM	5/6	6	20	4	240	200	4 800
8	256-QAM	3/4	8	20	4	320	240	5 760
9	256-QAM	5/6	8	20	4	320	-	-
10	1024-QAM	3/4	10	20	4	400	300	7 200
11	1024-QAM	5/6	10	20	4	400	-	-

Table A.3: MCSs for 0,864 MHz, N_{SS} = 3

MCS	Modulation	R	N_{BPSC}	N_{SD}	N_{SP}	N_{CBPS}	N_{DBPS}	Data rate
0	BPSK	1/2	1	20	4	60	30	720
1	QPSK	1/2	2	20	4	120	60	1 440
2	QPSK	3/4	2	20	4	120	90	2 160
3	16-QAM	1/2	4	20	4	240	120	2 880
4	16-QAM	3/4	4	20	4	240	180	4 320
5	64-QAM	2/3	6	20	4	360	240	5 760
6	64-QAM	3/4	6	20	4	360	270	6 480
7	64-QAM	5/6	6	20	4	360	300	7 200
8	256-QAM	3/4	8	20	4	480	360	8 640
9	256-QAM	5/6	8	20	4	480	400	9 600
10	1024-QAM	3/4	10	20	4	600	450	10 800
11	1024-QAM	5/6	10	20	4	600	500	12 000

Table A.4: MCSs for 0,864 MHz, N_{SS} = 4

MCS	Modulation	R	N_{BPSC}	N_{SD}	N_{SP}	N_{CBPS}	N_{DBPS}	Data rate
0	BPSK	1/2	1	20	4	80	40	960
1	QPSK	1/2	2	20	4	160	80	1 920
2	QPSK	3/4	2	20	4	160	120	2 880
3	16-QAM	1/2	4	20	4	320	160	3 840
4	16-QAM	3/4	4	20	4	320	240	5 760
5	64-QAM	2/3	6	20	4	480	320	7 680
6	64-QAM	3/4	6	20	4	480	360	8 640
7	64-QAM	5/6	6	20	4	480	400	9 600
8	256-QAM	3/4	8	20	4	640	480	11 520
9	256-QAM	5/6	8	20	4	640	-	-
10	1024-QAM	3/4	10	20	4	800	600	14 400
11	1024-QAM	5/6	10	20	4	800	-	-

Table A.5: MCSs for 0,864 MHz, N_{SS} = 5

MCS	Modulation	R	N_{BPSC}	N_{SD}	N_{SP}	N_{CBPS}	N_{DBPS}	Data rate
0	BPSK	1/2	1	20	4	100	50	1 200
1	QPSK	1/2	2	20	4	200	100	2 400
2	QPSK	3/4	2	20	4	200	150	3 600
3	16-QAM	1/2	4	20	4	400	200	4 800
4	16-QAM	3/4	4	20	4	400	300	7 200
5	64-QAM	2/3	6	20	4	600	400	9 600
6	64-QAM	3/4	6	20	4	600	450	10 800
7	64-QAM	5/6	6	20	4	600	500	12 000
8	256-QAM	3/4	8	20	4	800	600	14 400
9	256-QAM	5/6	8	20	4	800	-	-
10	1024-QAM	3/4	10	20	4	1 000	750	18 000
11	1024-QAM	5/6	10	20	4	1 000	-	-

Table A.6: MCSs for 0,864 MHz, N_{SS} = 6

MCS	Modulation	R	N_{BPSC}	N_{SD}	N_{SP}	N_{CBPS}	N_{DBPS}	Data rate
0	BPSK	1/2	1	20	4	120	60	1 440
1	QPSK	1/2	2	20	4	240	120	2 880
2	QPSK	3/4	2	20	4	240	180	4 320
3	16-QAM	1/2	4	20	4	480	240	5 760
4	16-QAM	3/4	4	20	4	480	360	8 640
5	64-QAM	2/3	6	20	4	720	480	11 520
6	64-QAM	3/4	6	20	4	720	540	12 960
7	64-QAM	5/6	6	20	4	720	600	14 400
8	256-QAM	3/4	8	20	4	960	720	17 280
9	256-QAM	5/6	8	20	4	960	800	19 200
10	1024-QAM	3/4	10	20	4	1 200	900	21 600
11	1024-QAM	5/6	10	20	4	1 200	1 000	24 000

A.3 MCS parameters for 1,728 MHz

Table A.7: MCSs for 1,728 MHz, N_{SS} = 1

MCS	Modulation	R	N_{BPSC}	N_{SD}	N_{SP}	N_{CBPS}	N_{DBPS}	Data rate
0	BPSK	1/2	1	52	4	52	26	624
1	QPSK	1/2	2	52	4	104	52	1 248
2	QPSK	3/4	2	52	4	104	78	1 872
3	16-QAM	1/2	4	52	4	208	104	2 496
4	16-QAM	3/4	4	52	4	208	156	3 744
5	64-QAM	2/3	6	52	4	312	208	4 992
6	64-QAM	3/4	6	52	4	312	234	5 616
7	64-QAM	5/6	6	52	4	312	260	6 240
8	256-QAM	3/4	8	52	4	416	312	7 488
9	256-QAM	5/6	8	52	4	416	-	-
10	1024-QAM	3/4	10	52	4	520	390	9 360
11	1024-QAM	5/6	10	52	4	520	-	-

Table A.8: MCSs for 1,728 MHz, N_{SS} = 2

MCS	Modulation	R	N_{BPSC}	N_{SD}	N_{SP}	N_{CBPS}	N_{DBPS}	Data rate
0	BPSK	1/2	1	52	4	104	52	1 248
1	QPSK	1/2	2	52	4	208	104	2 496
2	QPSK	3/4	2	52	4	208	156	3 744
3	16-QAM	1/2	4	52	4	416	208	4 992
4	16-QAM	3/4	4	52	4	416	312	7 488
5	64-QAM	2/3	6	52	4	624	416	9 984
6	64-QAM	3/4	6	52	4	624	468	11 232
7	64-QAM	5/6	6	52	4	624	520	12 480
8	256-QAM	3/4	8	52	4	832	624	14 976
9	256-QAM	5/6	8	52	4	832	-	-
10	1024-QAM	3/4	10	52	4	1 040	780	18 720
11	1024-QAM	5/6	10	52	4	1 040	-	-

Table A.9: MCSs for 1,728 MHz, N_{SS} = 3

MCS	Modulation	R	N_{BPSC}	N_{SD}	N_{SP}	N_{CBPS}	N_{DBPS}	Data rate
0	BPSK	1/2	1	52	4	156	78	1 872
1	QPSK	1/2	2	52	4	312	156	3 744
2	QPSK	3/4	2	52	4	312	234	5 616
3	16-QAM	1/2	4	52	4	624	312	7 488
4	16-QAM	3/4	4	52	4	624	468	11 232
5	64-QAM	2/3	6	52	4	936	624	14 976
6	64-QAM	3/4	6	52	4	936	702	16 848
7	64-QAM	5/6	6	52	4	936	780	18 720
8	256-QAM	3/4	8	52	4	1 248	936	22 464
9	256-QAM	5/6	8	52	4	1 248	1 040	24 960
10	1024-QAM	3/4	10	52	4	1 560	1 170	28 080
11	1024-QAM	5/6	10	52	4	1 560	1 300	31 200

Table A.10: MCSs for 1,728 MHz, N_{SS} = 4

MCS	Modulation	R	N_{BPSC}	N_{SD}	N_{SP}	N_{CBPS}	N_{DBPS}	Data rate
0	BPSK	1/2	1	52	4	208	104	2 496
1	QPSK	1/2	2	52	4	416	208	4 992
2	QPSK	3/4	2	52	4	416	312	7 488
3	16-QAM	1/2	4	52	4	832	416	9 984
4	16-QAM	3/4	4	52	4	832	624	14 976
5	64-QAM	2/3	6	52	4	1 248	832	19 968
6	64-QAM	3/4	6	52	4	1 248	936	22 464
7	64-QAM	5/6	6	52	4	1 248	1 040	24 960
8	256-QAM	3/4	8	52	4	1 664	1 248	29 952
9	256-QAM	5/6	8	52	4	1 664	-	-
10	1024-QAM	3/4	10	52	4	2 080	1 560	37 440
11	1024-QAM	5/6	10	52	4	2 080	-	-

Table A.11: MCSs for 1,728 MHz, N_{SS} = 5

MCS	Modulation	R	N_{BPSC}	N_{SD}	N_{SP}	N_{CBPS}	N_{DBPS}	Data rate
0	BPSK	1/2	1	52	4	260	130	3 120
1	QPSK	1/2	2	52	4	520	260	6 240
2	QPSK	3/4	2	52	4	520	390	9 360
3	16-QAM	1/2	4	52	4	1 040	520	12 480
4	16-QAM	3/4	4	52	4	1 040	780	18 720
5	64-QAM	2/3	6	52	4	1 560	1 040	24 960
6	64-QAM	3/4	6	52	4	1 560	1 170	28 080
7	64-QAM	5/6	6	52	4	1 560	1 300	31 200
8	256-QAM	3/4	8	52	4	2 080	1 560	37 440
9	256-QAM	5/6	8	52	4	2 080	-	-
10	1024-QAM	3/4	10	52	4	2 600	1 950	46 800
11	1024-QAM	5/6	10	52	4	2 600	-	-

Table A.12: MCSs for 1,728 MHz, N_{SS} = 6

MCS	Modulation	R	N_{BPSC}	N_{SD}	N_{SP}	N_{CBPS}	N_{DBPS}	Data rate
0	BPSK	1/2	1	52	4	312	156	3 744
1	QPSK	1/2	2	52	4	624	312	7 488
2	QPSK	3/4	2	52	4	624	468	11 232
3	16-QAM	1/2	4	52	4	1 248	624	14 976
4	16-QAM	3/4	4	52	4	1 248	936	22 464
5	64-QAM	2/3	6	52	4	1 872	1 248	29 952
6	64-QAM	3/4	6	52	4	1 872	1 404	33 696
7	64-QAM	5/6	6	52	4	1 872	1 560	37 440
8	256-QAM	3/4	8	52	4	2 496	1 872	44 928
9	256-QAM	5/6	8	52	4	2 496	2 080	49 920
10	1024-QAM	3/4	10	52	4	3 120	2 340	56 160
11	1024-QAM	5/6	10	52	4	3 120	2 600	62 400

A.4 MCS parameters for 3,456 MHz

Table A.13: MCSs for 3,456 MHz, N_{SS} = 1

MCS	Modulation	R	N_{BPSC}	N_{SD}	N_{SP}	N_{CBPS}	N_{DBPS}	Data rate
0	BPSK	1/2	1	108	6	108	54	1 296
1	QPSK	1/2	2	108	6	216	108	2 592
2	QPSK	3/4	2	108	6	216	162	3 888
3	16-QAM	1/2	4	108	6	432	216	5 184
4	16-QAM	3/4	4	108	6	432	324	7 776
5	64-QAM	2/3	6	108	6	648	432	10 368
6	64-QAM	3/4	6	108	6	648	486	11 664
7	64-QAM	5/6	6	108	6	648	540	12 960
8	256-QAM	3/4	8	108	6	864	648	15 552
9	256-QAM	5/6	8	108	6	864	720	17 280
10	1024-QAM	3/4	10	108	6	1 080	810	19 440
11	1024-QAM	5/6	10	108	6	1 080	900	21 600

Table A.14: MCSs for 3,456 MHz, N_{SS} = 2

MCS	Modulation	R	N_{BPSC}	N_{SD}	N_{SP}	N_{CBPS}	N_{DBPS}	Data rate
0	BPSK	1/2	1	108	6	216	108	2 592
1	QPSK	1/2	2	108	6	432	216	5 184
2	QPSK	3/4	2	108	6	432	324	7 776
3	16-QAM	1/2	4	108	6	864	432	10 368
4	16-QAM	3/4	4	108	6	864	648	15 552
5	64-QAM	2/3	6	108	6	1 296	864	20 736
6	64-QAM	3/4	6	108	6	1 296	972	23 328
7	64-QAM	5/6	6	108	6	1 296	1 080	25 920
8	256-QAM	3/4	8	108	6	1 728	1 296	31 104
9	256-QAM	5/6	8	108	6	1 728	1 440	34 560
10	1024-QAM	3/4	10	108	6	2 160	1 620	38 880
11	1024-QAM	5/6	10	108	6	2 160	1 800	43 200

Table A.15: MCSs for 3,456 MHz, N_{SS} = 3

MCS	Modulation	R	N_{BPSC}	N_{SD}	N_{SP}	N_{CBPS}	N_{DBPS}	Data rate
0	BPSK	1/2	1	108	6	324	162	3 888
1	QPSK	1/2	2	108	6	648	324	7 776
2	QPSK	3/4	2	108	6	648	486	11 664
3	16-QAM	1/2	4	108	6	1 296	648	15 552
4	16-QAM	3/4	4	108	6	1 296	972	23 328
5	64-QAM	2/3	6	108	6	1 944	1 296	31 104
6	64-QAM	3/4	6	108	6	1 944	1 458	34 992
7	64-QAM	5/6	6	108	6	1 944	1 620	38 880
8	256-QAM	3/4	8	108	6	2 592	1 944	46 656
9	256-QAM	5/6	8	108	6	2 592	2 160	51 840
10	1024-QAM	3/4	10	108	6	3 240	2 430	58 320
11	1024-QAM	5/6	10	108	6	3 240	2 700	64 800

Table A.16: MCSs for 3,456 MHz, N_{SS} = 4

MCS	Modulation	R	N_{BPSC}	N_{SD}	N_{SP}	N_{CBPS}	N_{DBPS}	Data rate
0	BPSK	1/2	1	108	6	432	216	5 184
1	QPSK	1/2	2	108	6	864	432	10 368
2	QPSK	3/4	2	108	6	864	648	15 552
3	16-QAM	1/2	4	108	6	1 728	864	20 736
4	16-QAM	3/4	4	108	6	1 728	1 296	31 104
5	64-QAM	2/3	6	108	6	2 592	1 728	41 472
6	64-QAM	3/4	6	108	6	2 592	1 944	46 656
7	64-QAM	5/6	6	108	6	2 592	2 160	51 840
8	256-QAM	3/4	8	108	6	3 456	2 592	62 208
9	256-QAM	5/6	8	108	6	3 456	2 880	69 120
10	1024-QAM	3/4	10	108	6	4 320	3 240	77 760
11	1024-QAM	5/6	10	108	6	4 320	3 600	86 400

Table A.17: MCSs for 3,456 MHz, N_{SS} = 5

MCS	Modulation	R	N_{BPSC}	N_{SD}	N_{SP}	N_{CBPS}	N_{DBPS}	Data rate
0	BPSK	1/2	1	108	6	540	270	6 480
1	QPSK	1/2	2	108	6	1 080	540	12 960
2	QPSK	3/4	2	108	6	1 080	810	19 440
3	16-QAM	1/2	4	108	6	2 160	1 080	25 920
4	16-QAM	3/4	4	108	6	2 160	1 620	38 880
5	64-QAM	2/3	6	108	6	3 240	2 160	51 840
6	64-QAM	3/4	6	108	6	3 240	2 430	58 320
7	64-QAM	5/6	6	108	6	3 240	2 700	64 800
8	256-QAM	3/4	8	108	6	4 320	3 240	77 760
9	256-QAM	5/6	8	108	6	4 320	3 600	86 400
10	1024-QAM	3/4	10	108	6	5 400	4 050	97 200
11	1024-QAM	5/6	10	108	6	5 400	4 500	108 000

Table A.18: MCSs for 3,456 MHz, N_{SS} = 6

MCS	Modulation	R	N_{BPSC}	N_{SD}	N_{SP}	N_{CBPS}	N_{DBPS}	Data rate
0	BPSK	1/2	1	108	6	648	324	7 776
1	QPSK	1/2	2	108	6	1 296	648	15 552
2	QPSK	3/4	2	108	6	1 296	972	23 328
3	16-QAM	1/2	4	108	6	2 592	1 296	31 104
4	16-QAM	3/4	4	108	6	2 592	1 944	46 656
5	64-QAM	2/3	6	108	6	3 888	2 592	62 208
6	64-QAM	3/4	6	108	6	3 888	2 916	69 984
7	64-QAM	5/6	6	108	6	3 888	3 240	77 760
8	256-QAM	3/4	8	108	6	5 184	3 888	93 312
9	256-QAM	5/6	8	108	6	5 184	4 320	103 680
10	1024-QAM	3/4	10	108	6	6 480	4 860	116 640
11	1024-QAM	5/6	10	108	6	6 480	5 400	129 600

A.5 MCS parameters for 6,912 MHz

Table A.19: MCSs for 6,912 MHz, N_{SS} = 1

MCS	Modulation	R	N_{BPSC}	N_{SD}	N_{SP}	N_{CBPS}	N_{DBPS}	Data rate
0	BPSK	1/2	1	234	8	234	117	2 808
1	QPSK	1/2	2	234	8	468	234	5 616
2	QPSK	3/4	2	234	8	468	351	8 424
3	16-QAM	1/2	4	234	8	936	468	11 232
4	16-QAM	3/4	4	234	8	936	702	16 848
5	64-QAM	2/3	6	234	8	1 404	936	22 464
6	64-QAM	3/4	6	234	8	1 404	1 053	25 272
7	64-QAM	5/6	6	234	8	1 404	1 170	28 080
8	256-QAM	3/4	8	234	8	1 872	1 404	33 696
9	256-QAM	5/6	8	234	8	1 872	1 560	37 440
10	1024-QAM	3/4	10	234	8	2 340	1 755	42 120
11	1024-QAM	5/6	10	234	8	2 340	1 950	46 800

Table A.20: MCSs for 6,912 MHz, N_{SS} = 2

MCS	Modulation	R	N_{BPSC}	N_{SD}	N_{SP}	N_{CBPS}	N_{DBPS}	Data rate
0	BPSK	1/2	1	234	8	468	234	5 616
1	QPSK	1/2	2	234	8	936	468	11 232
2	QPSK	3/4	2	234	8	936	702	16 848
3	16-QAM	1/2	4	234	8	1 872	936	22 464
4	16-QAM	3/4	4	234	8	1 872	1 404	33 696
5	64-QAM	2/3	6	234	8	2 808	1 872	44 928
6	64-QAM	3/4	6	234	8	2 808	2 106	50 544
7	64-QAM	5/6	6	234	8	2 808	2 340	56 160
8	256-QAM	3/4	8	234	8	3 744	2 808	67 392
9	256-QAM	5/6	8	234	8	3 744	3 120	74 880
10	1024-QAM	3/4	10	234	8	4 680	3 510	84 240
11	1024-QAM	5/6	10	234	8	4 680	3 900	93 600

Table A.21: MCSs for 6,912 MHz, N_{SS} = 3

MCS	Modulation	R	N_{BPSC}	N_{SD}	N_{SP}	N_{CBPS}	N_{DBPS}	Data rate
0	BPSK	1/2	1	234	8	702	351	8 424
1	QPSK	1/2	2	234	8	1 404	702	16 848
2	QPSK	3/4	2	234	8	1 404	1 053	25 272
3	16-QAM	1/2	4	234	8	2 808	1 404	33 696
4	16-QAM	3/4	4	234	8	2 808	2 106	50 544
5	64-QAM	2/3	6	234	8	4 212	2 808	67 392
6	64-QAM	3/4	6	234	8	4 212	3 159	75 816
7	64-QAM	5/6	6	234	8	4 212	3 510	84 240
8	256-QAM	3/4	8	234	8	5 616	4 212	101 088
9	256-QAM	5/6	8	234	8	5 616	4 680	112 320
10	1024-QAM	3/4	10	234	8	7 020	5 265	126 360
11	1024-QAM	5/6	10	234	8	7 020	5 850	140 400

Table A.22: MCSs for 6,912 MHz, N_{SS} = 4

MCS	Modulation	R	N_{BPSC}	N_{SD}	N_{SP}	N_{CBPS}	N_{DBPS}	Data rate
0	BPSK	1/2	1	234	8	936	468	11 232
1	QPSK	1/2	2	234	8	1 872	936	22 464
2	QPSK	3/4	2	234	8	1 872	1 404	33 696
3	16-QAM	1/2	4	234	8	3 744	1 872	44 928
4	16-QAM	3/4	4	234	8	3 744	2 808	67 392
5	64-QAM	2/3	6	234	8	5 616	3 744	89 856
6	64-QAM	3/4	6	234	8	5 616	4 212	101 088
7	64-QAM	5/6	6	234	8	5 616	4 680	112 320
8	256-QAM	3/4	8	234	8	7 488	5 616	134 784
9	256-QAM	5/6	8	234	8	7 488	6 240	149 760
10	1024-QAM	3/4	10	234	8	9 360	7 020	168 480
11	1024-QAM	5/6	10	234	8	9 360	7 800	187 200

Table A.23: MCSs for 6,912 MHz, N_{SS} = 5

MCS	Modulation	R	N_{BPSC}	N_{SD}	N_{SP}	N_{CBPS}	N_{DBPS}	Data rate
0	BPSK	1/2	1	234	8	1 170	585	14 040
1	QPSK	1/2	2	234	8	2 340	1 170	28 080
2	QPSK	3/4	2	234	8	2 340	1 755	42 120
3	16-QAM	1/2	4	234	8	4 680	2 340	56 160
4	16-QAM	3/4	4	234	8	4 680	3 510	84 240
5	64-QAM	2/3	6	234	8	7 020	4 680	112 320
6	64-QAM	3/4	6	234	8	7 020	5 265	126 360
7	64-QAM	5/6	6	234	8	7 020	5 850	140 400
8	256-QAM	3/4	8	234	8	9 360	7 020	168 480
9	256-QAM	5/6	8	234	8	9 360	7 800	187 200
10	1024-QAM	3/4	10	234	8	11 700	8 775	210 600
11	1024-QAM	5/6	10	234	8	11 700	9 750	234 000

Table A.24: MCSs for 6,912 MHz, N_{SS} = 6

MCS	Modulation	R	N_{BPSC}	N_{SD}	N_{SP}	N_{CBPS}	N_{DBPS}	Data rate
0	BPSK	1/2	1	234	8	1 404	702	16 848
1	QPSK	1/2	2	234	8	2 808	1 404	33 696
2	QPSK	3/4	2	234	8	2 808	2 106	50 544
3	16-QAM	1/2	4	234	8	5 616	2 808	67 392
4	16-QAM	3/4	4	234	8	5 616	4 212	101 088
5	64-QAM	2/3	6	234	8	8 424	5 616	134 784
6	64-QAM	3/4	6	234	8	8 424	6 318	151 632
7	64-QAM	5/6	6	234	8	8 424	7 020	168 480
8	256-QAM	3/4	8	234	8	11 232	8 424	202 176
9	256-QAM	5/6	8	234	8	11 232	9 360	224 640
10	1024-QAM	3/4	10	234	8	14 040	10 530	252 720
11	1024-QAM	5/6	10	234	8	14 040	11 700	280 800

A.6 MCS parameters for 13,824 MHz

Table A.25: MCSs for 13,824 MHz, N_{SS} = 1

MCS	Modulation	R	N_{BPSC}	N_{SD}	N_{SP}	N_{CBPS}	N_{DBPS}	Data rate
0	BPSK	1/2	1	468	16	468	234	5 616
1	QPSK	1/2	2	468	16	936	468	11 232
2	QPSK	3/4	2	468	16	936	702	16 848
3	16-QAM	1/2	4	468	16	1 872	936	22 464
4	16-QAM	3/4	4	468	16	1 872	1 404	33 696
5	64-QAM	2/3	6	468	16	2 808	1 872	44 928
6	64-QAM	3/4	6	468	16	2 808	2 106	50 544
7	64-QAM	5/6	6	468	16	2 808	2 340	56 160
8	256-QAM	3/4	8	468	16	3 744	2 808	67 392
9	256-QAM	5/6	8	468	16	3 744	3 120	74 880
10	1024-QAM	3/4	10	468	16	4 680	3 510	84 240
11	1024-QAM	5/6	10	468	16	4 680	3 900	93 600

Table A.26: MCSs for 13,824 MHz, N_{SS} = 2

MCS	Modulation	R	N_{BPSC}	N_{SD}	N_{SP}	N_{CBPS}	N_{DBPS}	Data rate
0	BPSK	1/2	1	468	16	936	468	11 232
1	QPSK	1/2	2	468	16	1 872	936	22 464
2	QPSK	3/4	2	468	16	1 872	1 404	33 696
3	16-QAM	1/2	4	468	16	3 744	1 872	44 928
4	16-QAM	3/4	4	468	16	3 744	2 808	67 392
5	64-QAM	2/3	6	468	16	5 616	3 744	89 856
6	64-QAM	3/4	6	468	16	5 616	4 212	101 088
7	64-QAM	5/6	6	468	16	5 616	4 680	112 320
8	256-QAM	3/4	8	468	16	7 488	5 616	134 784
9	256-QAM	5/6	8	468	16	7 488	6 240	149 760
10	1024-QAM	3/4	10	468	16	9 360	7 020	168 480
11	1024-QAM	5/6	10	468	16	9 360	7 800	187 200

Table A.27: MCSs for 13,824 MHz, N_{SS} = 3

MCS	Modulation	R	N_{BPSC}	N_{SD}	N_{SP}	N_{CBPS}	N_{DBPS}	Data rate
0	BPSK	1/2	1	468	16	1 404	702	16 848
1	QPSK	1/2	2	468	16	2 808	1 404	33 696
2	QPSK	3/4	2	468	16	2 808	2 106	50 544
3	16-QAM	1/2	4	468	16	5 616	2 808	67 392
4	16-QAM	3/4	4	468	16	5 616	4 212	101 088
5	64-QAM	2/3	6	468	16	8 424	5 616	134 784
6	64-QAM	3/4	6	468	16	8 424	6 318	151 632
7	64-QAM	5/6	6	468	16	8 424	7 020	168 480
8	256-QAM	3/4	8	468	16	11 232	8 424	202 176
9	256-QAM	5/6	8	468	16	11 232	9 360	224 640
10	1024-QAM	3/4	10	468	16	14 040	10 530	252 720
11	1024-QAM	5/6	10	468	16	14 040	11 700	280 800

Table A.28: MCSs for 13,824 MHz, N_{SS} = 4

MCS	Modulation	R	N_{BPSC}	N_{SD}	N_{SP}	N_{CBPS}	N_{DBPS}	Data rate
0	BPSK	1/2	1	468	16	1 872	936	22 464
1	QPSK	1/2	2	468	16	3 744	1 872	44 928
2	QPSK	3/4	2	468	16	3 744	2 808	67 392
3	16-QAM	1/2	4	468	16	7 488	3 744	89 856
4	16-QAM	3/4	4	468	16	7 488	5 616	134 784
5	64-QAM	2/3	6	468	16	11 232	7 488	179 712
6	64-QAM	3/4	6	468	16	11 232	8 424	202 176
7	64-QAM	5/6	6	468	16	11 232	9 360	224 640
8	256-QAM	3/4	8	468	16	14 976	11 232	269 568
9	256-QAM	5/6	8	468	16	14 976	12 480	299 520
10	1024-QAM	3/4	10	468	16	18 720	14 040	336 960
11	1024-QAM	5/6	10	468	16	18 720	15 600	374 400

Table A.29: MCSs for 13,824 MHz, N_{SS} = 5

MCS	Modulation	R	N_{BPSC}	N_{SD}	N_{SP}	N_{CBPS}	N_{DBPS}	Data rate
0	BPSK	1/2	1	468	16	2 340	1 170	28 080
1	QPSK	1/2	2	468	16	4 680	2 340	56 160
2	QPSK	3/4	2	468	16	4 680	3 510	84 240
3	16-QAM	1/2	4	468	16	9 360	4 680	112 320
4	16-QAM	3/4	4	468	16	9 360	7 020	168 480
5	64-QAM	2/3	6	468	16	14 040	9 360	224 640
6	64-QAM	3/4	6	468	16	14 040	10 530	252 720
7	64-QAM	5/6	6	468	16	14 040	11 700	280 800
8	256-QAM	3/4	8	468	16	18 720	14 040	336 960
9	256-QAM	5/6	8	468	16	18 720	15 600	374 400
10	1024-QAM	3/4	10	468	16	23 400	17 550	421 200
11	1024-QAM	5/6	10	468	16	23 400	19 500	468 000

Table A.30: MCSs for 13,824 MHz, N_{SS} = 6

MCS	Modulation	R	N_{BPSC}	N_{SD}	N_{SP}	N_{CBPS}	N_{DBPS}	Data rate
0	BPSK	1/2	1	468	16	2 808	1 404	33 696
1	QPSK	1/2	2	468	16	5 616	2 808	67 392
2	QPSK	3/4	2	468	16	5 616	4 212	101 088
3	16-QAM	1/2	4	468	16	11 232	5 616	134 784
4	16-QAM	3/4	4	468	16	11 232	8 424	202 176
5	64-QAM	2/3	6	468	16	16 848	11 232	269 568
6	64-QAM	3/4	6	468	16	16 848	12 636	303 264
7	64-QAM	5/6	6	468	16	16 848	14 040	336 960
8	256-QAM	3/4	8	468	16	22 464	16 848	404 352
9	256-QAM	5/6	8	468	16	22 464	18 720	449 280
10	1024-QAM	3/4	10	468	16	28 080	21 060	505 440
11	1024-QAM	5/6	10	468	16	28 080	23 400	561 600

A.7 MCS parameters for 20,736 MHz

Table A.31: MCSs for 20,736 MHz, N_{SS} = 1

MCS	Modulation	R	N_{BPSC}	N_{SD}	N_{SP}	N_{CBPS}	N_{DBPS}	Data rate
0	BPSK	1/2	1	636	22	636	318	7 632
1	QPSK	1/2	2	636	22	1 272	636	15 264
2	QPSK	3/4	2	636	22	1 272	954	22 896
3	16-QAM	1/2	4	636	22	2 544	1 272	30 528
4	16-QAM	3/4	4	636	22	2 544	1 908	45 792
5	64-QAM	2/3	6	636	22	3 816	2 544	61 056
6	64-QAM	3/4	6	636	22	3 816	2 862	68 688
7	64-QAM	5/6	6	636	22	3 816	3 180	76 320
8	256-QAM	3/4	8	636	22	5 088	3 816	91 584
9	256-QAM	5/6	8	636	22	5 088	4 240	101 760
10	1024-QAM	3/4	10	636	22	6 360	4 770	114 480
11	1024-QAM	5/6	10	636	22	6 360	5 300	127 200

Table A.32: MCSs for 20,736 MHz, N_{SS} = 2

MCS	Modulation	R	N_{BPSC}	N_{SD}	N_{SP}	N_{CBPS}	N_{DBPS}	Data rate
0	BPSK	1/2	1	636	22	1 272	636	15 264
1	QPSK	1/2	2	636	22	2 544	1 272	30 528
2	QPSK	3/4	2	636	22	2 544	1 908	45 792
3	16-QAM	1/2	4	636	22	5 088	2 544	61 056
4	16-QAM	3/4	4	636	22	5 088	3 816	91 584
5	64-QAM	2/3	6	636	22	7 632	5 088	122 112
6	64-QAM	3/4	6	636	22	7 632	5 724	137 376
7	64-QAM	5/6	6	636	22	7 632	6 360	152 640
8	256-QAM	3/4	8	636	22	10 176	7 632	183 168
9	256-QAM	5/6	8	636	22	10 176	8 480	203 520
10	1024-QAM	3/4	10	636	22	12 720	9 540	228 960
11	1024-QAM	5/6	10	636	22	12 720	10 600	254 400

Table A.33: MCSs for 20,736 MHz, N_{SS} = 3

MCS	Modulation	R	N_{BPSC}	N_{SD}	N_{SP}	N_{CBPS}	N_{DBPS}	Data rate
0	BPSK	1/2	1	636	22	1 908	954	22 896
1	QPSK	1/2	2	636	22	3 816	1 908	45 792
2	QPSK	3/4	2	636	22	3 816	2 862	68 688
3	16-QAM	1/2	4	636	22	7 632	3 816	91 584
4	16-QAM	3/4	4	636	22	7 632	5 724	137 376
5	64-QAM	2/3	6	636	22	11 448	7 632	183 168
6	64-QAM	3/4	6	636	22	11 448	8 586	206 064
7	64-QAM	5/6	6	636	22	11 448	9 540	228 960
8	256-QAM	3/4	8	636	22	15 264	11 448	274 752
9	256-QAM	5/6	8	636	22	15 264	12 720	305 280
10	1024-QAM	3/4	10	636	22	19 080	14 310	343 440
11	1024-QAM	5/6	10	636	22	19 080	15 900	381 600

Table A.34: MCSs for 20,736 MHz, N_{SS} = 4

MCS	Modulation	R	N_{BPSC}	N_{SD}	N_{SP}	N_{CBPS}	N_{DBPS}	Data rate
0	BPSK	1/2	1	636	22	2 544	1 272	30 528
1	QPSK	1/2	2	636	22	5 088	2 544	61 056
2	QPSK	3/4	2	636	22	5 088	3 816	91 584
3	16-QAM	1/2	4	636	22	10 176	5 088	122 112
4	16-QAM	3/4	4	636	22	10 176	7 632	183 168
5	64-QAM	2/3	6	636	22	15 264	10 176	244 224
6	64-QAM	3/4	6	636	22	15 264	11 448	274 752
7	64-QAM	5/6	6	636	22	15 264	12 720	305 280
8	256-QAM	3/4	8	636	22	20 352	15 264	366 336
9	256-QAM	5/6	8	636	22	20 352	16 960	407 040
10	1024-QAM	3/4	10	636	22	25 440	19 080	457 920
11	1024-QAM	5/6	10	636	22	25 440	21 200	508 800

Table A.35: MCSs for 20,736 MHz, N_{SS} = 5

MCS	Modulation	R	N_{BPSC}	N_{SD}	N_{SP}	N_{CBPS}	N_{DBPS}	Data rate
0	BPSK	1/2	1	636	22	3 180	1 590	38 160
1	QPSK	1/2	2	636	22	6 360	3 180	76 320
2	QPSK	3/4	2	636	22	6 360	4 770	114 480
3	16-QAM	1/2	4	636	22	12 720	6 360	152 640
4	16-QAM	3/4	4	636	22	12 720	9 540	228 960
5	64-QAM	2/3	6	636	22	19 080	12 720	305 280
6	64-QAM	3/4	6	636	22	19 080	14 310	343 440
7	64-QAM	5/6	6	636	22	19 080	15 900	381 600
8	256-QAM	3/4	8	636	22	25 440	19 080	457 920
9	256-QAM	5/6	8	636	22	25 440	21 200	508 800
10	1024-QAM	3/4	10	636	22	31 800	23 850	572 400
11	1024-QAM	5/6	10	636	22	31 800	26 500	636 000

Table A.36: MCSs for 20,736 MHz, N_{SS} = 6

MCS	Modulation	R	N_{BPSC}	N_{SD}	N_{SP}	N_{CBPS}	N_{DBPS}	Data rate
0	BPSK	1/2	1	636	22	3 816	1 908	45 792
1	QPSK	1/2	2	636	22	7 632	3 816	91 584
2	QPSK	3/4	2	636	22	7 632	5 724	137 376
3	16-QAM	1/2	4	636	22	15 264	7 632	183 168
4	16-QAM	3/4	4	636	22	15 264	11 448	274 752
5	64-QAM	2/3	6	636	22	22 896	15 264	366 336
6	64-QAM	3/4	6	636	22	22 896	17 172	412 128
7	64-QAM	5/6	6	636	22	22 896	19 080	457 920
8	256-QAM	3/4	8	636	22	30 528	22 896	549 504
9	256-QAM	5/6	8	636	22	30 528	25 440	610 560
10	1024-QAM	3/4	10	636	22	38 160	28 620	686 880
11	1024-QAM	5/6	10	636	22	38 160	31 800	763 200

A.8 MCS parameters for 27,648 MHz

Table A.37: MCSs for 27,648 MHz, N_{SS} = 1

MCS	Modulation	R	N_{BPSC}	N_{SD}	N_{SP}	N_{CBPS}	N_{DBPS}	Data rate
0	BPSK	1/2	1	936	32	936	468	11 232
1	QPSK	1/2	2	936	32	1 872	936	22 464
2	QPSK	3/4	2	936	32	1 872	1 404	33 696
3	16-QAM	1/2	4	936	32	3 744	1 872	44 928
4	16-QAM	3/4	4	936	32	3 744	2 808	67 392
5	64-QAM	2/3	6	936	32	5 616	3 744	89 856
6	64-QAM	3/4	6	936	32	5 616	4 212	101 088
7	64-QAM	5/6	6	936	32	5 616	4 680	112 320
8	256-QAM	3/4	8	936	32	7 488	5 616	134 784
9	256-QAM	5/6	8	936	32	7 488	6 240	149 760
10	1024-QAM	3/4	10	936	32	9 360	7 020	168 480
11	1024-QAM	5/6	10	936	32	9 360	7 800	187 200

Table A.38: MCSs for 27,648 MHz, N_{SS} = 2

MCS	Modulation	R	N_{BPSC}	N_{SD}	N_{SP}	N_{CBPS}	N_{DBPS}	Data rate
0	BPSK	1/2	1	936	32	1 872	936	22 464
1	QPSK	1/2	2	936	32	3 744	1 872	44 928
2	QPSK	3/4	2	936	32	3 744	2 808	67 392
3	16-QAM	1/2	4	936	32	7 488	3 744	89 856
4	16-QAM	3/4	4	936	32	7 488	5 616	134 784
5	64-QAM	2/3	6	936	32	11 232	7 488	179 712
6	64-QAM	3/4	6	936	32	11 232	8 424	202 176
7	64-QAM	5/6	6	936	32	11 232	9 360	224 640
8	256-QAM	3/4	8	936	32	14 976	11 232	269 568
9	256-QAM	5/6	8	936	32	14 976	12 480	299 520
10	1024-QAM	3/4	10	936	32	18 720	14 040	336 960
11	1024-QAM	5/6	10	936	32	18 720	15 600	374 400

Table A.39: MCSs for 27,648 MHz, N_{SS} = 3

MCS	Modulation	R	N_{BPSC}	N_{SD}	N_{SP}	N_{CBPS}	N_{DBPS}	Data rate
0	BPSK	1/2	1	936	32	2 808	1 404	33 696
1	QPSK	1/2	2	936	32	5 616	2 808	67 392
2	QPSK	3/4	2	936	32	5 616	4 212	101 088
3	16-QAM	1/2	4	936	32	11 232	5 616	134 784
4	16-QAM	3/4	4	936	32	11 232	8 424	202 176
5	64-QAM	2/3	6	936	32	16 848	11 232	269 568
6	64-QAM	3/4	6	936	32	16 848	12 636	303 264
7	64-QAM	5/6	6	936	32	16 848	14 040	336 960
8	256-QAM	3/4	8	936	32	22 464	16 848	404 352
9	256-QAM	5/6	8	936	32	22 464	18 720	449 280
10	1024-QAM	3/4	10	936	32	28 080	21 060	505 440
11	1024-QAM	5/6	10	936	32	28 080	23 400	561 600

Table A.40: MCSs for 27,648 MHz, N_{SS} = 4

MCS	Modulation	R	N_{BPSC}	N_{SD}	N_{SP}	N_{CBPS}	N_{DBPS}	Data rate
0	BPSK	1/2	1	936	32	3 744	1 872	44 928
1	QPSK	1/2	2	936	32	7 488	3 744	89 856
2	QPSK	3/4	2	936	32	7 488	5 616	134 784
3	16-QAM	1/2	4	936	32	14 976	7 488	179 712
4	16-QAM	3/4	4	936	32	14 976	11 232	269 568
5	64-QAM	2/3	6	936	32	22 464	14 976	359 424
6	64-QAM	3/4	6	936	32	22 464	16 848	404 352
7	64-QAM	5/6	6	936	32	22 464	18 720	449 280
8	256-QAM	3/4	8	936	32	29 952	22 464	539 136
9	256-QAM	5/6	8	936	32	29 952	24 960	599 040
10	1024-QAM	3/4	10	936	32	37 440	28 080	673 920
11	1024-QAM	5/6	10	936	32	37 440	31 200	748 800

Table A.41: MCSs for 27,648 MHz, N_{SS} = 5

MCS	Modulation	R	N_{BPSC}	N_{SD}	N_{SP}	N_{CBPS}	N_{DBPS}	Data rate
0	BPSK	1/2	1	936	32	4 680	2 340	56 160
1	QPSK	1/2	2	936	32	9 360	4 680	112 320
2	QPSK	3/4	2	936	32	9 360	7 020	168 480
3	16-QAM	1/2	4	936	32	18 720	9 360	224 640
4	16-QAM	3/4	4	936	32	18 720	14 040	336 960
5	64-QAM	2/3	6	936	32	28 080	18 720	449 280
6	64-QAM	3/4	6	936	32	28 080	21 060	505 440
7	64-QAM	5/6	6	936	32	28 080	23 400	561 600
8	256-QAM	3/4	8	936	32	37 440	28 080	673 920
9	256-QAM	5/6	8	936	32	37 440	31 200	748 800
10	1024-QAM	3/4	10	936	32	46 800	35 100	842 400
11	1024-QAM	5/6	10	936	32	46 800	39 000	936 000

Table A.42: MCSs for 27,648 MHz, N_{SS} = 6

MCS	Modulation	R	N_{BPSC}	N_{SD}	N_{SP}	N_{CBPS}	N_{DBPS}	Data rate
0	BPSK	1/2	1	936	32	5 616	2 808	67 392
1	QPSK	1/2	2	936	32	11 232	5 616	134 784
2	QPSK	3/4	2	936	32	11 232	8 424	202 176
3	16-QAM	1/2	4	936	32	22 464	11 232	269 568
4	16-QAM	3/4	4	936	32	22 464	16 848	404 352
5	64-QAM	2/3	6	936	32	33 696	22 464	539 136
6	64-QAM	3/4	6	936	32	33 696	25 272	606 528
7	64-QAM	5/6	6	936	32	33 696	28 080	673 920
8	256-QAM	3/4	8	936	32	44 928	33 696	808 704
9	256-QAM	5/6	8	936	32	44 928	37 440	898 560
10	1024-QAM	3/4	10	936	32	56 160	42 120	1 010 880
11	1024-QAM	5/6	10	936	32	56 160	46 800	1 123 200

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