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TECHNICAL REPORT

Digital Enhanced Cordless Telecommunications (DECT); DECT evolution technical study; Requirements and technical analysis for the further evolution of DECT and DECT ULE

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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).

Modal verbs terminology

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Executive summary

The present document contains the outcome of a series of studies identified by ETSI TC DECT and required for the short and mid-term evolution of DECT and ULE technologies. The present document is primary addressed to TC DECT and DECT industry communities and, as well, to other participants from new industry sectors that may be considering using DECT technology for new applications.

1 Scope

The present document describes the outcome of a series of studies identified by ETSI TC DECT and required for the short and mid-term evolution of DECT and ULE technologies. The outcome of the present document will be used for planning the further evolution of technology and the immediate technology roadmap during the next years.

The outcome of the present document will allow addressing new applications and markets for ULE and DECT technologies in the mid-term, and will contribute to the ETSI effort on Internet of Things (IoT).

It is not the scope of the present document the description of DECT and ULE technologies. The core part of the DECT technology is described in the DECT common interface standard (ETSI EN 300 175) [i.1] to [i.8] plus a series of profiles, such as the GAP [i.9], DPRS [i.12], WRS [i.10] or New Generation DECT [i.14] to [i.18]. Further regulatory details of the Radio interface are given in ETSI EN 301 406 [i.24], ETSI EN 301 908-10 [i.37] and ETSI EN 300 176-1 [i.11]. A summary overview of DECT technology can be found in ETSI EN 300 175-1 [i.1].

The ULE technology is described in ETSI TS 102 939 series [i.19] and [i.20].

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 EN 300 444: "Digital Enhanced Cordless Telecommunications (DECT); Generic Access Profile (GAP)".

[i.10] ETSI EN 300 700: "Digital Enhanced Cordless Telecommunications (DECT); Wireless Relay Station (WRS)".

[i.11] ETSI EN 300 176-1: "Digital Enhanced Cordless Telecommunications (DECT); Test specification; part 1: radio".

[i.12] ETSI EN 301 649: "Digital Enhanced Cordless Telecommunications (DECT); DECT Packet Radio Service (DPRS)".

[i.13] IETF RFC 8105: "Transmission of IPv6 Packets over Digital Enhanced Cordless Telecommunications (DECT) Ultra Low Energy (ULE)".

[i.14] ETSI TS 102 527-1: "Digital Enhanced Cordless Telecommunications (DECT); New Generation DECT; Part 1: Wideband Speech".

[i.15] ETSI TS 102 527-2: "Digital Enhanced Cordless Telecommunications (DECT); New Generation DECT; Part 2: Support of transparent IP packet data".

[i.16] ETSI TS 102 527-3: "Digital Enhanced Cordless Telecommunications (DECT); New Generation DECT; Part 3: Extended wideband speech services".

[i.17] ETSI TS 102 527-4: "Digital Enhanced Cordless Telecommunications (DECT); New Generation DECT; Part 4: Light Data Services; Software Update Over The Air (SUOTA), content downloading and HTTP based applications".

[i.18] 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.19] 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.20] 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.21] IEEE 802.11-2012™: "IEEE Standard for Information technology -- Telecommunications and information exchange between systems Local and metropolitan area networks -- Specific requirements -- Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specifications".

[i.22] IEEE 802.16-2012™: "IEEE Standard for Information technology--Broadband Wireless Metropolitan Area Networks (MANs)--IEEE Standard for Air Interface for Broadband Wireless Access Systems".

[i.23] ARIB STD T-95: "OFDMA/TDMA TDD Broadband Wireless Access System (XGP)".

[i.24] ETSI EN 301 406: "Digital Enhanced Cordless Telecommunications (DECT); Harmonised Standard covering the essential requirements of article 3.2 of the Directive 2014/53/EU".

[i.25] ETSI TS 136 300: "LTE; Evolved Universal Terrestrial Radio Access (E-UTRA) and Evolved Universal Terrestrial Radio Access Network (E-UTRAN); Overall description; Stage 2 (3GPP TS 36.300)".

[i.26] IETF RFC 4291: "IP Version 6 Addressing Architecture".

[i.27] IETF RFC 4861: "Neighbor Discovery for IP version 6 (IPv6)".

[i.28] IETF RFC 4862: "IPv6 Stateless Address Autoconfiguration".

[i.29] IETF RFC 4944: "Transmission of IPv6 Packets over IEEE 802.15.4 Networks".

[i.30] IETF RFC 6282: "Compression Format for IPv6 Datagrams over IEEE 802.15.4-Based Networks".

[i.31] IETF RFC 6775: "Neighbor Discovery Optimization for IPv6 over Low-Power Wireless Personal Area Networks (6LoWPANs)".

[i.32] IETF RFC 7136: "Significance of IPv6 Interface Identifiers".

[i.33] IETF RFC 3610: "Counter with CBC-MAC (CCM)".

[i.34] IETF RFC 4903: "Multi-Link Subnet Issues".

[i.35] IETF RFC 8065: "Privacy Considerations for IPv6 Adaptation- Layer Mechanisms".

[i.36] IETF RFC 3315: "Dynamic Host Configuration Protocol for IPv6 (DHCPv6)".

[i.37] 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.38] IEEE 802.15.4™: "IEEE Standard for Local and metropolitan area networks -- Part 15.4: Low-Rate Wireless Personal Area Networks (LR-WPANs)".

3 Definitions, symbols and abbreviations

3.1 Definitions

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

algorithm: mathematical process or function that transforms an input into an output

antenna diversity: diversity implies that the Radio Fixed Part (RFP) for each bearer independently can select different antenna properties such as gain, polarization, coverage patterns and other features that may affect the practical coverage

NOTE: A typical example is space diversity, provided by two vertically polarized antennas separated by 10 cm to 20 cm.

expedited (messages, procedures, operations): MAC C/O operations (messages, procedures, operations) intended for ultra-fast setup and release of bearers, allowing in most cases reduction in the number of messages and early or late U-plane transmission compared to regular procedures

expedited connections: advanced connections able to use the expedited messages of the advanced connection control part 2 set and their associate procedures for bearer setup and release

eXtended Global Platform: wireless technology deployed mostly in Japan using the micro-cell and TDD/OFDMA/SC-FDM technology

guard space: nominal interval between the end of a radio transmission in a given slot and the start of a radio transmission in the next successive slot

half slot: one 48th of a TDMA frame which is used to support one physical channel

Home Automation Network: network that connects all sensors and actors in a house or apartment, providing interoperability for devices of different vendors and typically has a connection to the Internet

NOTE: The Home Automation Network is used for various applications, from Home Automation and Security to Climate Control and Energy Management.

3.2 Symbols and abbreviations

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

6CO	6LoWPAN Context Option
6LBR	6LoWPAN Border Router
6LN	6LoWPAN Node
6LoWPAN	IPv6 over Low power Wireless Personal Area Networks
ACK	(positive) ACKnowledgement
ADPCM	Adaptive Differential Pulse Code Modulation
AES	Advanced Encryption Standard
ARIB	Association of Radio Industries and Businesses (Japan)
ARO	Address Registration Option
ARQ	Automatic Repeat reQuest (also Automatic Repeat Query)
BER	Bit Error Ratio
BPSK	Binary Phase Shift Keying
B_S	Slow Broadcast channel
C	higher layer control Channel (see C_S and C_F)
C/L	ConnectionLess mode
C/O	Connection Oriented mode
CBC	Connectionless Bearer Control
CBC-MAC	Cipher Block Chaining Message Authentication Code
CCM	Counter with CBC-MAC
CGA	Cryptographically Generated Address
CI	Cell Identity
CI	Common Interface (standard)
CODEC	COder-DECoder
CP	Cyclic Prefix (OFDM)
C-plane	Control plane
CRC	Cyclic Redundancy Check
DBPSK	Differential Binary Phase Shift Keying
DC	Direct Current
DECT	Digital Enhanced Cordless Telecommunications
DES	Data Encryption Standard
DFT	Discrete Fourier Transform
DHCPv6	Dynamic Host Configuration Protocol for IPv6
DLC	Data Link Control Layer
DPRS	DECT Packet Radio Service

NOTE: See ETSI EN 301 649 [i.12].

DQPSK	Differential Quaternary Phase Shift Keying
DSAA2	DECT Standard Authentication Algorithm #2
DSC	DECT Standard Cipher (algorithm)
DSC2	DECT Standard Cipher #2 (algorithm)
DSP	Digital Signal Processing
DSSS	Direct Sequence Spread Spectrum
E type	B-field multiplexer mode when the slot carries signalling only (channels C_F, G_F and M)
E+U type	B-field multiplexer mode when the slot carries U-plane data (channel I_{PF}) AND signalling (channels G_F and M)
FDD	Frequency Division Duplex
FDMA	Frequency Division Multiple Access
FEC	Forward Error Correction
FFT	Fast Fourier Transform
FMID	Fixed part MAC IDentity
FP	DECT Fixed Part
FT	Fixed radio Termination
GAP	Generic Access Profile
G_{FA}	higher layer information control channel (slow) (a logical channel to the MAC layer)
GFSK	Gaussian Frequency Shift Keying
GI	Guard Interval (OFDM)

GMSK	Gaussian Minimum Shift Keying
GSM	Global System for Mobile communications
HBA	Hash-Based Address
HLM	High Level Modulation
HSDPA	High Speed Downlink Packet Access
HTTP	Hypertext Transfer Protocol
I	higher layer Information channel (see I_N and I_P) in general
IE	Information Element
IID	Interface Identifier (IPv6)
IID	Internet IDentity
I_N	higher layer Information channel unprotected (in general, any variant)
I_{NX}	A possible new name for the I_{PX} service
I_P	higher layer Information channel protected (in general, any variant)
IP	Internet Protocol
IPEI	International Portable Equipment Identity
I_{PQ}	higher layer Information channel (protected) with single subfield format
IPv6	Internet Protocol Version 6
I_{PX}	higher layer Information channel, encoded protected, minimum delay operation
IV	Initialization Vector
IWU	InterWorking Unit
LA	Location Area
LAN	Local Area Network
LL-ULE	Low Latency ULE
LO	Local Oscillator
LS	Least Squares (channel estimation)
M	MAC control channel in general (on A-tail or B fields)
MAC (CCM)	Message Authentication Code
MAC	Medium Access Control layer
MAP	bit MAPpings
MBC	Multi-Bearer Control
MCS	Multi-Channel Set
ME	Management Entity
MIMO	Multiple Input Multiple Output
MM	Mobility Management
MMSE	Minimum Mean-Square Error (channel estimation)
MTU	Maximum Transmission Unit
MUX	time MUltipleXor
N	identities channel
NBMA	Non-Broadcast Multi-Access
N_{FFT}	FFT size
NG-PHS	Next Generation PHS
NLOS	No Line Of Sight
NR	Normal-Reverse
N_S	Split identities channel on B-field for ULE
N_T	identities information channel or one message in such channel
NWK	NetWorK layer
OFDM	Orthogonal Frequency Division Multiplex
OFDMA	Orthogonal Frequency Division Multiple Access
P	Paging channel
PAPR	Peak-to-Average Power Ratio
PBX	Private Branch eXchange
PCM	Pulse Code Modulation
PDU	Protocol Data Unit
PHL	PHysical Layer
PHS	Personal Handyphone System
PHY	PHYsical
PMID	Portable part MAC IDentity (MAC layer)
PP	DECT Portable Part
PSK	Phase Shift Keying
PT	Portable radio Termination

PVC	Permanent Virtual Circuit
Q	system information channel
QAM	Quadrature Amplitude Modulation
QPSK	Quadrature Phase Shift Keying
RF	Radio Frequency
RFP	Radio Fixed Part
RFPI	Radio Fixed Part Identity
RX	Receiver
SC-FDM	Single Carrier Frequency Division Multiplexing
SC-FDMA	Single Carrier Frequency Division Multiple Access
STD	Standard
STF	Special Task Force
SUOTA	Software Update Over The Air
TBC	Traffic Bearer Control
TCP	Transmission Control Protocol
TDD	Time Division Duplex
TDMA	Time Division Multiple Access
T_{FFT}	Transform period (FFT)
T_{SYM}	Symbol interval
U type	B-field multiplexer mode when the slot carries U-plane data only (channels I_N or I_P)
UDP	User Datagram Protocol
UEAR	Ultra-expedited_access_request (ULE)
UEARR	Ultra-expedited_access_request_ready_for_release (ULE)
UEBC	Ultra-expedited_continuation of transmission (ULE)
UER	Ultra-expedited_release (ULE)
UERR	Ultra-expedited ready for release (ULE)
ULE	Ultra Low Energy
UMTS	Universal Mobile Telecommunication System
U-plane	User plane
URLLC	Ultra Reliable Low Latency Communications
UTRAN	UMTS Terrestrial Radio Access Network
Wi-Fi	IEEE 802.11 [i.21] family of standards
WiMAX	IEEE 802.16 [i.22] family of standards
WLL	Wireless Local loop
WRS	Wireless Relay Station
XGP	eXtended Global Platform
XOR	eXclusive OR

4 Overview

4.1 Scope of the present document

The present document contains a list of technical studies required for the short term, mid-term and long term evolution of DECT technology. The selection of studies was chosen by TC DECT in years 2016 and beginning of 2017 as response to several technology and industry needs.

The outcome of the present document will be used for planning the further evolution of technology and the immediate technology roadmap during the next years.

The studies covered by the present document address four main technical and business areas:

- DECT ULE (Ultra Low Energy) variants with High Reliability and Low Latency intended primarily for industry automation scenarios.
- Ultra-Reliable Low Latency Communications (URLLC) high bit rate multibearer solutions intended primarily for the professional audio industry.
- Introduction to the long term evolution of DECT based on OFDM.

- DECT ULE (Ultra Low Energy); Implementation of the new IETF RFC 8105 [i.13] "Transmission of IPv6 Packets over Digital Enhanced Cordless Telecommunications (DECT) Ultra Low Energy (ULE)".

4.2 List of technical studies covered by the present document

As result of TC DECT priorities, driven by several industry needs, the technical studies covered by the present document are the following:

1) **Low Latency ULE:** This is an original proposal intended for ULE applications. It is described in clause 5. In short, it is a new technical approach, reusing some elements of "conventional" ULE, but with a new MAC strategy and a new approach for channel setups. The approach proves that delays in the range of 1 ms to 2 ms are possible in high performance systems.

2) **URLLC C/O streaming solutions:** This is a high bitrate streaming technology with Ultra reliability and Low Latency intended for professional audio/video applications such as studio microphones. The work was based on requirements provided by the professional audio industry and is based on an original solution proposed by the present document. In short, high performance, high reliability, low latency radio microphones operating as far as 24 bit × 96 kHz (high definition audio) can be built on top of DECT technology. Several critical areas are identified as for further study, i.e. the channel coding and the implementation of 256 QAM.

3) **URLLC C/L multicast multibearer streaming:** This another variation of the high bitrate, ultra-reliable, low latency, streaming technology, but now applied to downstream multicast traffics. The typical use case are studio and consumer (hi-end) headphones. Again, the study proves that such technology can be build based on DECT, even for the most demanding cases. Significant extra work will be needed, at standard and at implementation levels.

4) **HLM (High Level Modulation):** This is a clause analysing the status of current DECT standard and identifying what is covered, what should be added, the limitations of current procedures and what areas would require further study.

5) **Long Term evolution based on OFDM:** This is an introductory clause of a very complex technical topic that will have to be further developed during the next 5 years. It can be considered as a kick-off of DECT-2020 future technology. STF has recommended the evolution towards OFDM/OFDMA/SC-FDMA with several technical parameters and options proposed an analysed in some detail.

6) **Implementation of the IETF RFC 8105 [i.13]: "Transmission of IPv6 Packets over Digital Enhanced Cordless Telecommunications (DECT) Ultra Low Energy (ULE)":** contains an introduction to this RFC and a set of recommendations on DECT standardization for improving the support of ULE devices based on the RFC.

7) **Other improvements:** Contains other miscellaneous improvements to DECT technology.

5 Low Latency Machine-to-Machine communications

5.1 Low Latency ULE

The purpose of this clause is analysing the options for creating a low latency technology for Machine-to-Machine communications (M2M) targeted at ≤ 1 ms one-way delay and with as low power consumption as possible. Nevertheless, latency target should be prioritized over consumption.

This clause describes the technical solution proposed by STF 518 for Low latency ULE. In this context ULE is understood as a M2M technology optimized for low throughput communications with *as low power consumption as possible* for the intended specification. Now, *Low Latency* is prioritized with a target for one way latency in the range of 1 ms. The technology cannot be as *"Low Energy"* as regular ULE due to physical (spectrum access) considerations but it is still energy efficient.

5.2 Design objectives

The design objectives can be summarized as follows:

- One millisecond one-way latency target (≤ 1 ms).
- Optimized for low throughput and low traffic.
- Asynchronous operation.
- As low power consumption as possible: nevertheless, latency target should be prioritized over consumption.
- Reliable U-plane with error detection, error protection, and acknowledgement. Reliability is prioritized over any other consideration except security.
- State-of-the art security. If possible with the same CCM encryption used in ULE. Security is prioritized over any other design consideration.
- Efficient use of the spectrum, as possible.
- Minimum disturbance to existing systems.

Taking into account the efficiency requirements, the foreseen market needs, the available spectrum and technology possibilities the delay objective is understood as an average target. Therefore an asynchronous packet mode technology is the design target.

This means that the 1 ms delay target should be achieved in most of the cases, but extra delay is allowed in case of collisions, spectrum congestion, etc. provided that the probability is enough low.

This also means that the technology may not suitable for some critical applications (such as real time control of a drone). For such applications a strategy of synchronous mode with reserved bandwidth is recommended.

NOTE: Another study has been made for the opposite case: low latency with high data throughput in synchronous mode. See clause 6.

5.3 Physical layer, spectrum and implementation considerations

Extended DECT/IMT-2000 spectrum 1 880 MHz to 1 920 MHz is recommended. Nevertheless the solution proposed in this analysis will also work using only existing 1 880 MHz to 1 900 MHz spectrum.

Operation and latency are improved with extended spectrum due to the larger number of available carrier x slot options for setting-up bearers.

Implementations are assumed to be "modern", based on signal processing, wideband radios, and able to process the entire band at the same time. Limitations such as single-channel radios, slow hopping, etc., are not to be taken into account since they collide with the design objective of reducing latency.

5.4 Basic principles of the solution

5.4.1 Modulation

Taken into account the low throughput objectives, it seems that standard GFSK modulation may be used in most cases. However, High Level Modulation (HLM) may also be introduced if needed.

5.4.2 No scan sequence limitations in RFP

In order to provide immediate access to the channels without any need to wait for Primary (or further) scan sequences for bearer setup, the concept of "scan sequence" is avoided. The RFP will allow setups at any time, in any frequency and in any slot. In practice this means that the RFP implementation should be continuously listening for possible setups in all carriers. This requires a wideband implementations based on signal processing or a multi-transceiver implementation.

This decision is taken to protect the design objective of minimum latency.

5.4.3 Absolute fast setup capability in PPs

By the same reason, (to provide immediate access to the channels without any need to wait for paging or scan sequences), the PP are assumed to have the capability to operate in "absolute setup detection". This means a new ultra-fast setup mode where the PP is able to listen for setup, when required, in any frequency and in any slot. In practice, this also means that the PP implementation should be based on signal processing and wideband radios or, at least, multiple transceivers.

In the case of the PP, it is foreseen that this capability may be active permanently, or only when required. I.e. after an initial setup attempt by the PP towards the RFP. The exact behaviour depends on the specific PP type and intended application. As with ULE, it may be assumed that there may exist different types of PPs (actuators, sensors, etc.) with different operation cycles.

This decision is taken to protect the design objective of minimum latency.

5.4.4 Slot type

Standard full slots may be used. The delay advantage by using half slots (0,2 ms) does not justify the loss in bit-rate capability. Therefore, standard full slots are proposed.

Nevertheless, the topic of which slot should be used for sending the dummy bearer is open. Half slots may be an option there.

5.4.5 No limitations on slot direction in the frame

In order to achieve the design objectives, it is proposed that *any* slot in the frame may be used in *any* direction, at any time.

The reason of this design decision is not compromising the delay objective.

This solution, despite being completely different to the one used in cordless telephony, it is not fundamentally incompatible with DECT technology. The basic DECT low MAC layer design in ETSI EN 300 175-3 [i.3] (frame structure, inter-slot spaces, etc.) is ready to support this operation. On the other hand, new messages with new semantics may need to be created. It may also impact dummy bearer fields and operation.

5.4.6 Basic decision: connection-oriented vs. connection-less bearers

Current C/O bearers (duplex bearers separated 5 ms) cannot provide the delay objective. Therefore, they cannot be used.

In another clause of the present document (clause 6) an alternative approach for low latency based on multi-bearer and still using existing bearers is proposed. However, it is considered as not suitable for the intended application because of the very different traffic objectives, and also because of the need of guarantee low roundtrip-delay. Low delay roundtrip operations are fundamental to allow low latency asynchronous operation since they are used in the bearer setup process.

Therefore, a C/L approach where each simplex bearer is independent of any other running in opposite direction is proposed. This, combined with the total frame flexibility, and with the *"any-carrier any-slot*" setup capability, should allow the intended delay objective.

It is therefore proposed that the elementary bearer to be used in the solution would be the *simplex bearer*. Simplex bearers should be set up independently and may be placed in any slot position and in any direction, as required by the latency objective.

5.4.7 Basic decision: mixed MAC C/L / C/O approach

5.4.7.1 Proposal of mixed C/L / C/O approach

However, a complete C/L solution will be very weak in terms of reliability. Mutual acknowledgement of the communication is required to respect the reliability objective. Furthermore, from the point of view of U-plane, a C/O approach simplifies greatly the design and also reduces the need of extra overhead (identifiers, etc.).

Therefore, it is proposed a mixed approach where:

- Elementary bearer is the *simplex* bearer used always in one-way C/L mode. This means C/L "low-MAC" mode.
- However, since most operations will be in fact bilateral, the semantic of the MAC messages may assume some kind of C/O operation. Similar sequences as those used in ULE are proposed.

From the point of view of the U-plane, it will operate in C/O mode since it provides several advantages.

- Therefore, a C/O U-plane transported over a C/L set of bearers will be used.

NOTE: There is no fundamental contradiction in this combination. It is used, i.e. in the Internet.

5.4.7.2 Q1/Q2 bit reporting

An important impact of the mixed approach is the reporting of MAC quality (Q1/Q2) bits that would be specific for the new operation mode and different to standard duplex bearer reporting. This has to be specifically designed (it would be something new) but it is seems as relatively straightforward if it is restricted to single bearer operation. In practice it means that the Q1/Q2 reporting in each new *simplex* bearer would not indicate the quality of its opposite bearer in a duplex pair (12 time-slots ago) but something different.

5.4.8 U-plane model: MAC, DLC and NWK

Accepting the MAC U-plane in "standard" C/O mode, and that mutual MAC ACK (Q1/Q2) is solved, then, the rest of the U-plane design may be as in ULE. This includes the DLC, NWK layer and the also the CCM security (implemented at DLC layer).

The NWK layer model will be C/O with full MM (as in ULE).

5.4.9 C-plane NWK layer and security

The same security used in ULE will be used (CCM with AES-128). Therefore, security level will be as good as in "regular" ULE and the same encryption and authentication NWK procedures will be used.

For NWK layer C-plane procedures the same approach used in ULE will be used. An ancillary regular connection using duplex bearers will be set when needed, while the LL-ULE connection will be reserved for U-plane low latency data.

5.4.10 Possible MAC messages

The new simplex bearers will be controlled by a new set of messages. The following messages are proposed. They may be implemented using Advanced Connection Control Part 2 set, where there are enough available codes.

Table 1: Advanced Connection Control Part 2 (table 7.45a of ETSI EN 300 175-3 [i.3])

Command				Advanced connection control part 2 messages
0	0	0	0	expedited_access_request
0	0	0	1	expedited_access_request_ready_for_release
0	0	1	0	Null or G_{FA} channel transmission
0	1	0	0	Ultra-expedited_access_request (UEAR)
0	1	0	1	Ultra-expedited_access_request_ready_for_release (UEARR)
0	1	1	0	Ultra-expedited ready for release (UERR)
0	1	1	1	Ultra-expedited_release (UER)
1	0	0	0	Ultra-expedited_continuation of transmission (UEBC)
				}
All others				} reserved
				}
1	1	1	0	ready for release with G_{FA} transmission
1	1	1	1	expedited_release with G_{FA} transmission
NOTE: The codes for bearer_confirm is re-used from the regular advanced connection control set.				

The semantic of the message sequences may be reused in most cases from "standard" ULE. Nevertheless, the following differences are identified:

1) Due to the planned "simplex" flexible mode, the concept of "connected state" never exists.

2) By the same reasons, it is believed that UERR and UER should contain full PMID/FMID addresses.

3) Therefore, another mechanism is needed for inserting release reasons and/or DLC GFA packets. This may be based on B-field signalling.

4) It is believed that all packets should contain a MAC message with FMID/PMID addresses. In case of multiple bursts, a new message "continuation of transmission (UEBC)" is proposed.

5) Termination of transmissions of long bursts and/or bi-directional transfers require further investigation (due to the DLC insertion).

5.4.11 Channel selection

New channel selection rules are needed due to the planned use of the carrier/slot grid prioritizing latency over other considerations. It is understood that both peers do their respective channel selection. In latency critical nodes, this has to be done continuously "in advance" in order to avoid any extra delay caused by it.

RFP will always use "total fast setup" (see clause 5.4.3) for setting the bearers in FT → PT direction.

5.4.12 Example of sequence 1: short burst transmission PT => FT

Figure 1 shows the basic sequence for the simplest operation: PT → FT transmission of a single burst ("burst" in ULE terms), with low latency and success case.

Triggering event

slot	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23
Dummy carrier 1	↓																							
Carrier 4				⇧																				
Carrier 6					⇩																			

Figure 1: Example 1: Short burst PT => FT transmission

Triggering event: It happens at the beginning of slot 2 (but too late to use slot 2).

PT channel selection: If the target is getting the minimum latency, then the PT has to be pre-selecting channels in advance. Under the assumption that it has preselected slot 3, carrier 4 as the closest transmission possibility.

Transmission PT → FT in slot 3: The PT starts transmission with a UEARR message, and inserting in the same slot the U-plane packet (with format, CCM tail, etc.).

Delay: assuming no collision nor transmission errors, the RX decodes and validates the packet at the end of slot 3. Therefore, the total latency is 0,83 ms (2 / 24 × 10 ms). This assumes a worst case position of the triggering event and slot position.

Therefore, the latency of the technology is 0,83 ms.

Note that there is no "slot direction concept" in the new technology (see clause 5.4.5). Any slot with a free frequency may be used.

FT reply: The FT replies doing an independent channel selection for setting the first available slot, that in the example is slot 4, carrier 6. UER Release message. This closes the MAC operation.

Insertion of DLC GFA: This requires further study but due to the unpaired simplex bearers use, it seems more robust not using the A-field in order to use it for full addresses. In the example, the B-field of the slot 4 may be used. The reason for release may also be inserted there.

5.4.13 Further considerations for short burst transmissions

5.4.13.1 Error cases

In case or error in either PT or FT transmissions, the PT will not receive the reply in slot 4. In such a case the PT should repeat the attempt, ideally in slot 5. The proper algorithms for handling the case should be designed. They will be different from ULE but the same principles for collision prevention apply. To reduce delay, they should try to introduce the random spread in the carrier dimension (taken advantage of the multi-carrier capability of the radio) and only if this is not possible, they will use the slot dimension. In most cases, frequency is enough and slot 5 may be used.

Error cases increases latency, but this is unavoidable and may happen in near any radio technology. Probability of the case should be low enough, and if the case arises, the extra delay should be the minimum.

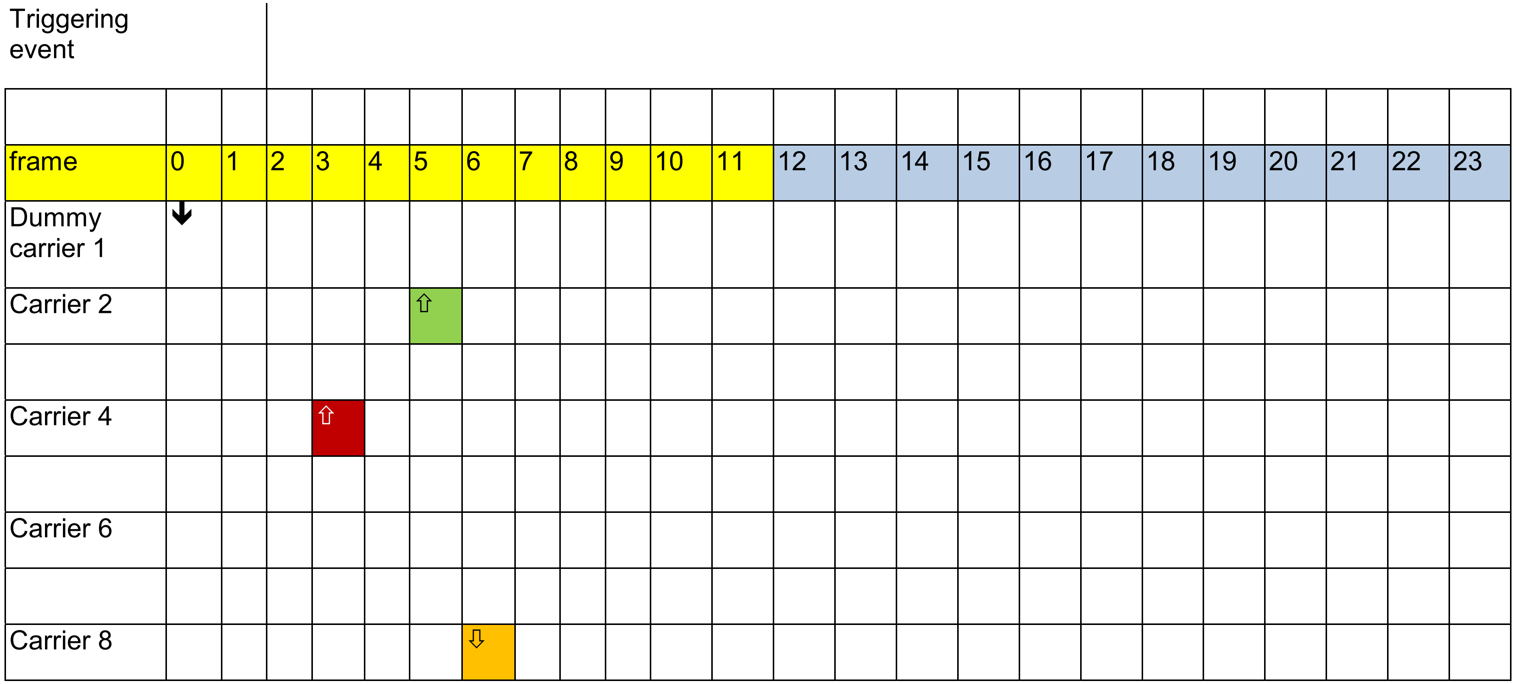

Figure 2: Example 1 with error or collision in slot 3

The delay for the transmission with error and retransmission is 1,66 ms.

5.4.13.2 Effect of the dummy bearer

An issue with difficult solution is the effect of the dummy bearer. This mostly impacts upstream transmissions (downstream transmissions are less critical). In the previous example, the dummy was set on slot 1. However, if the dummy would be on slot 3, then additional limitations may apply.

The problem is not fundamental but an implementation one. Since multi-carrier radios may be used, in theory the PT may send the packet on slot 3 in a different carrier from the dummy. The problem is the near impossible implementation of the radio at FT side.

frame	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23
Dummy carrier 1				↓																				
Carrier 2																								
Carrier 4				⇧																				

Figure 3: Collision with the dummy bearer

For the opposite case (FT → PT transmissions) the issue does not exist. The FT may send the packet AND the dummy on the same slot (at different carriers). The PT may receive both without major issue.

frame	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23
Dummy carrier 1				↓																				
Carrier 4				⇩																				
Carrier 6					⇧																			

Figure 4: No collision with the dummy bearer

5.4.13.3 Solutions to the dummy bearer issue

The most obvious solution for the dummy bearer issue, is accepting 0,41 ms of additional delay if the case happen. This would mean using slot 4 instead of slot 3.

Triggering event

frame	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23
Dummy carrier 1				↓																				
Carrier 4					⇧																			
Carrier 6						⇩																		

Figure 5: Obvious solution to the collision with the dummy bearer issue

A more creative idea, possible only if special dummies are used would be the following:

a) Use a half slot for the dummy. Use the second half of the full slot for it.

b) If the FT detects an initial transmission from a PT in the slot, then it disables temperately the dummy (to allow decoding the received burst).

c) Then the FT may receive the packet.

slot	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23
Dummy carrier 1				/↓																				
Carrier 4				⇧																				
Carrier 6					⇩																			

Figure 6: Optimized solution to the collision with the dummy bearer issue

5.4.14 Example of sequence 2: multi burst transmission PT => FT

There are potentially several strategies for implementing multi burst transmissions. In absence of specific requirements for the case, a sequence that mimics ULE operation has been chosen. It seems flexible enough for supporting multiple cases, however, it is not the ultimate in terms of optimization if high data rates in asymmetric mode are the expected traffic pattern. As already said, any further analysis requires specific requirements.

The following example shows a one way PT → FT transmission of a burst of 5 packets. Success case.

Triggering event

frame	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23
Dummy carrier 1	↓									⇧			⇩											
Carrier 2				⇧								⇧												
Carrier 3											⇩													
Carrier 4					⇩																			
Carrier 4														⇧?										
Carrier 5							⇩	⇧																
Carrier 6																								
Carrier 7						⇧			⇩															
....																								

Figure 7: Example: one way PT → FT transmission of a burst of 5 packets. Success case

Triggering event: at the beginning of slot 2.

PT channel selection: as for the single burst case, but repeated independently for all slots.

Messages:

Slot 3. PT → FT UEAR (Ultra-expedited_access_request); B-field carries U-plane data.

Slot 4. FT → PT UERR (Ultra-expedited_ ready for release); B-field empty or in E-format (carrying i.e. G_{FA}). Note the different carrier due to independent channel selection.

Slot 5. PT → FT UEBC (Ultra-expedited_continuation of transmission); B-field carries U-plane data. Note the different carrier due to independent channel selection.

Slot 6. FT → PT UERR (Ultra-expedited_ ready for release); B-field empty or in E-format (carrying i.e. G_{FA}).

Slot 7. PT → FT UEBC (Ultra-expedited_continuation of transmission); B-field carries U-plane data.

Slot 8. FT → PT UERR (Ultra-expedited_ ready for release); B-field empty or in E-format (carrying i.e. G_{FA}).

Slot 9. PT → FT UEBC (Ultra-expedited_continuation of transmission); B-field carries U-plane data.

Slot 10. FT → PT UERR (Ultra-expedited_ ready for release); B-field empty or in E-format (carrying i.e. G_{FA}).

Slot 11. PT → FT UEBC (Ultra-expedited_ready for release); B-field carries U-plane data. This is the last segment of the burst. Therefore, PT inserts the "ready for release".

Slot 12. FT → PT UER (Ultra-expedited_release); B-field empty or in E-format (carrying i.e. G_{FA}).

Slot 13 (t.b.d.). PT → FT UER (Ultra-expedited_ release); B-field empty or in E-format (carrying i.e. G_{FA}). This requires further analysis. For the time being a "release" is inserted to mimic as much as possible ULE behaviour. However, it may probably be removed.

General comment

As already said, there are potentially other strategies for implementing multi burst transmissions. However, it would be necessary to analyse specific requirements and expected traffic patterns to justify a different solution.

6 Ultra Reliable Low Latency circuit mode C/O streaming applications

6.1 Overview

The purpose of this analysis is investigating the possibilities of current DECT technology and proposing certain improvements in order to better addressing Low Latency Audio Streaming applications using C/O service. Examples of such applications are radio microphones, headsets, etc.

6.2 Investigation of the possibilities of current DECT technology

6.2.1 General

This part of the analysis investigates the possibilities of current DECT for such applications. The analysis does not enter in the limitations of current chipsets that may be too focused on cordless telephony applications.

By existing DECT technology, it is meant the current DECT base standard [i.1] to [i.8] V2.7.1. Some fixings/clarifications may be needed in some never used features (i.e. HLM). In addition to that, application profile(s) covering the new applications would be needed.

This analysis presents figures of data rate and delays achievable with current DECT technology and with some proposed "minor" enhancements.

6.2.2 Latency in single-bearer DECT transmissions

The latency of a DECT single bearer transmissions ranges from 9,6 ms to around 15 ms depending on several conditions.

The typical figure of 10 ms assumes that the sampling interval may be adjusted to the optimal point (what is normally true in telephony) and that transmission may start slightly before capturing the samples for whole packet, what is normally true in PCM (and other codecs). Therefore, 10 ms is given as typical figure for the transmission delay.

The delay may be reduced by 0,4 ms to 9,6 ms if the Rx side operates with early delivery of the samples without waiting for the complete reception of the packet and without observing the Z field. This is mentioned here because it may be interesting for the intended application. This is possible if the codec is PCM or PCM-like.

For complex codecs and for any case when there is a block-processing of the whole packet (for instance a FEC or an interleaving schema), the delay is 10,42 ms. 10 ms come from the sampling interval and 0,42 ms (rounded to 0,4) for the transmission time.

Refer to clause 6.2.6 for a detailed explanation on when the 9,6 ms, 10 ms or the 10,4 ms delay apply.

In some applications, it is not possible changing the sampling interval according to changes in slot position due to handover, etc. In these cases, if seamless transmission is desired, 5 ms extra latency may apply. This extra delay can be avoided, by establishing limitations to the handover space.

In case of two-way transmissions, there is a 5 ms gain in the roundtrip delay due to the fixed difference in positions between the slots for both ways: the total delay is 15 ms / 15,4 ms (instead of 20 ms / 20,8 ms).

6.2.3 Latency in multi-bearer (symmetric) DECT transmissions

The previous analysis is only true in case of single bearer transmissions. In multi-bearer transmissions, the delay is reduced. The optimal case is achieved when two slot pairs in the edge of the frame are used (i.e. slots 0,11 and 12, 23). In such a case the delay is 4,6 ms / 5 ms / 5,4 ms. See example 2 in figure 2 for the configuration.

In previous figures 1 to 7, 0,4 ms comes from the transmission time of a DECT slot (10 ms / 24 ms) and the 5 ms comes from the sampling time. See clause 6.2.6 for this discussion.

In symmetric transmissions it is not possible going down from 4,6 ms / 5 ms / 5,4 ms because of the frame interval allocated to the opposite direction. Therefore, this is the latency limit with current DECT frame structure for symmetric transmissions.

6.2.4 Introducing flexibility in slot positions

It would be possible going below this limit if the concept of complete flexibility in slot positions is introduced. However, this would mean a significant change over current standard.

If flexibility in slot positions is introduced, the best delay that can be achieved is 0,4 ms / 0,8 ms / 1,2 ms. This would happen in a multi-bearer transmission of 12 + 12 full slots with completely alternate slot positions. See example 3 in figure 2 for the configuration. Note that this configuration is an extension to the technology and is not possible today.

Refer to clause 6.2.6 for the discussion when the ±0,4 ms should be considered.

The detailed calculation of the 0,4 ms / 0,8 ms / 1,3 ms is done as follows:

- sampling time: 10 / 24 = 0,42 × 2 = 0,84 ms;
- transmission time = 0,42 ms (this can be avoided by starting Tx before creating the whole packet, what is possible in PCM);
- total: 0,42 ms / 0,84 ms / 1,26 ms.

As already said, this is a significant change over existing standard that would require changing many basic principles and protocol elements (messages) at MAC and NWK layers. Nevertheless, it is considered feasible in the mid-term.

6.2.5 Latency in multi-bearer (asymmetric) DECT transmissions

In case of asymmetric connections, the situation is better because existing standard already allows placing the asymmetric bearers (double simplex bearers) in any position of the frame. Therefore, it is already possible designing a system with 0,4 ms / 0,8 ms / 1,3 ms delay. This may be done i.e. for an asymmetric configuration of 1 + 13 slots with the slots configured as shown in example 4 of figure 2. The figure is shown for uplink transmission; however, the downlink case is equivalent.

A 0,4 ms / 0,8 ms / 1,3 ms configuration as shown in example 4 of figure 2 requires a near fixed slot positions. However, some flexibility is possible if the number of slots is increased. Frequency handover is always possible, but its optimal implementation introduces the need of some extra provisions in the standard.

A greater flexibility for handover or a reduction in the number of bearers may be achieved if 0,4 extra ms of worst-case delay (equivalent to one slot) are accepted. This means a total delay of 0,8 ms / 1,3 ms / 1,7 ms. Implementation of 0,8 ms / 1,3 ms / 1,7 ms delay systems is relatively easy in the short term with current standard. The flexibility may also be used to reduce the number of slots required in the forward direction, what be of interest if the bitrate is large enough, i.e. by using HLM. See examples 5 and 6 in figure 2 that show two cases of 0,8 ms / 1,3 ms / 1,7 ms delay configurations.

6.2.6 Sampling references for PCM-like and other codecs and impact on the delay.

This clause provides a detailed explanation on when the 0,4 ms additional delay should be accounted and when not. This is normally not relevant for telephony but becomes relevant when the intention is obtaining minimum delay.

Assuming the example 4 of figure 9 (0,4 ms / 0,8 ms / 1,3 ms), when there is an usable slot every 2, and using the sample transmitted over slot 3 as example.

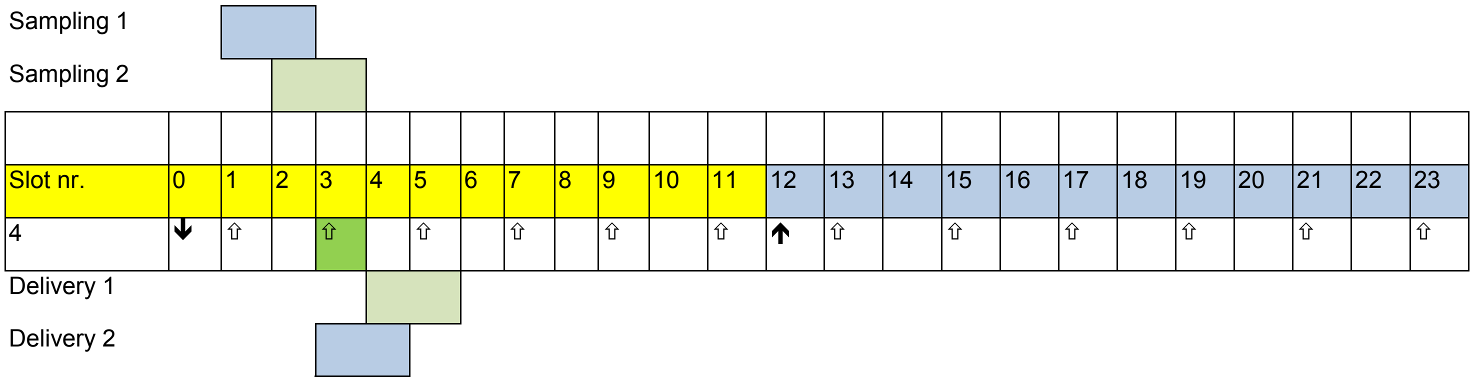

Figure 8: Detailed explanation of delay

The packet should contain 0,8 ms of audio samples in all cases, (1/12 of a DECT frame).

Sampling 1 means establishing the sampling reference at the beginning of the slot. This means that the whole packet of samples is available before start of transmission. This allows the use of any codec and block processing of the whole packet.(i.e. FEC, interleaving, etc.). This mode may be required by some codecs.

Delivery 1 means waiting until the complete reception of the packet before delivering it. Again, it allows block processing of the whole packet.(i.e. FEC, interleaving), and also observation of CRC (i.e. for muting purposes).

Sampling 2 means establishing the sampling reference at the end of the slot (in fact a few bits before). This means that the whole packet of samples is not available before start of transmission. However, they will be available at the proper time to allow complete transmission. This is possible if the coded is PCM or derived from PCM and there is no block processing or interleaving. The exact value of the "few bits before" depends on the codec type, codec word size and if CRCs have to be added.

Delivery 2 means start delivering of samples just "a few bits" after start of reception of the packet without waiting until the complete reception of the whole packet. This does not allow any block processing.(i.e. FEC, interleaving), and does not allow observation of CRC (if used) before delivering the samples (i.e. for muting). However, this is possible if the codec is PCM and latency has to be prioritized. The exact value of the "a few bits" depends on codec type, codec word size and size of preambles, separation space and A-field in the slot. CRCs (if used) may be observed and used for handover purposes.

The combination of sampling 1 and delivery 1 gives the worst case figure (1,3 ms in the example 4 of figure 9). However, this configuration allows the use of any codec and full flexibility for block processing of the whole slot. It allows use of FEC and/or interleaving.

However, if the codec is PCM or other sample based, it would be possible synchronizing the sampling at the end of the packet. This is "sampling 2" and allows saving 0,4 ms of delay.

At the receiver side, if the codec is PCM and there is no intention to use/observe any CRC bits (see note 1), then configuration "delivery 2" may be used. This saves 0,4 additional ms. The combination of delivery 2 with sampling 2 gives the minimum delay figure (0,4 ms in the example 4 of figure 9).

NOTE 1: regarding sampling 2, the remark on observation use of CRC bits (which may prevent muting/) refers to the real time use of the stream. It is anyway still possible observing the CRCs and marking the errors and samples for post processing. This makes possible the application of muting and/or redundancy in a later stage. It is also possible using them for handover decisions.

In the case of a microphone, it means that the artist/operator listening to the real-time stream received from the microphone may hear the error (i.e. a "click" produced by an erroneous packet), but this may be corrected in the recorded master.

NOTE 2: The same principles may apply to some types of FEC, that may be usable for correcting data in a non-real-time processing. i.e. a FEC that only adds redundancy data at the end of the packet may be used with sampling 2 and delivery 2 (the size of redundancy field impacts the delay). Real-time outflow would not benefit from FEC , but stored data may do it.

6.2.7 Examples

Legend

0	Slot "normally" FT → PT and slot number
0	Slot "normally" PT → FT and slot number
🡇	Duplex bearer downlink
🡅	Duplex bearer uplink
⇧	Double-simplex uplink
⇩	Double-simplex downlink

case	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23
1	↓												↑											
2	↓											↓	↑											↑
3	↓	↑	↓	↑	↓	↑	↓	↑	↓	↑	↓	↑	↓	↑	↓	↑	↓	↑	↓	↑	↓	↑	↓	↑
4	↓	⇧		⇧		⇧		⇧		⇧		⇧	↑	⇧		⇧		⇧		⇧		⇧		⇧
5	↓		⇧			⇧			⇧			⇧	↑		⇧			⇧			⇧			⇧
6	↓		⇧		⇧		⇧		⇧			⇧	↑		⇧		⇧		⇧		⇧			⇧
7	↓⇧	⇧	⇧	⇧	⇧	⇧	⇧	⇧	⇧	⇧	⇧	⇧	↑⇧	⇧	⇧	⇧	⇧	⇧	⇧	⇧	⇧	⇧	⇧	⇧

Figure 9: Compilation of examples

6.2.8 Data rates and data rate considerations

6.2.8.1 Using full slots

Table 2 shows the gross data rates in the forward direction (I_N mode in kbit/s) for configurations 1 + 13 and 1 + 23 with the different modulation options:

Full slot

Table 2: Data rates for different configurations

Conf.	GFSK	QPSK	8PSK	16QAM	64 QAM	Delay (ms)
x	1	2	3	4	6	
1 + 1	32	64	96	128	192	9,6 / 10 / 10,4
1 + 13	416	832	1 248	1 664	2 496	0,4 / 0,8 / 1,3
1 + 23	736	1 472	2 208	2 944	4 416	0,4 / 0,8 / 1,3

Gross data rates will be reduced due to the error detection and error protection code overheads. Current overheads used by the IP/IPQ service and IPX (turbocodec) have not been taken into account since it is unclear if they will be re-used in the intended application or an independent protection scheme will be used.

6.2.8.2 Reference values for audio applications

- 16 bits at 48 kbit/s = 768 kbit/s
- 20 bits at 96 kbit/s = 1 920 kbit/s
- 24 bits at 96 kbit/s = 2 496 kbit/s
- 24 bits at 192 kbit/s = 4 608 kbit/s

6.2.8.3 Using double slots

Some improvement in data rate may be achieved by suing double slots, what in practice means using the space between two slots plus one of the A-fields. The theoretic data rates are shown below:

Double slot:

Table 3: Data rates for different configurations using double slots

Conf.	GFSK	QPSK	8PSK	16QAM	64 QAM	Delay (ms)
x	1	2	3	4	6	
1d + 1d	80	160	240	320	480	9,2 /10 / 10,8
1d + 7d	560	1 120	1 680	2 240	3 360	0,4 / 1,3 / 2,1
1d + 11d	880	1 760	2 640	3 520	5 280	0,4 / 1,3 / 2,1

As previously, these are gross data rates at transparent IN service and should be reduced by the error detection and error protection code overheads. Current overheads used by the IP/IPQ service and IPX (turbo-codec) have not been taken into account since it is unclear if they will be re-used in the intended application or an independent protection scheme will be used.

6.3 Further improvements to DECT technology

6.3.1 General

The scope of this part of the analysis is focused on investigating several possible improvements to existing DECT technology. Potentially, many technology improvements are possible. The present document proposes and analyses the following ideas:

- Use of A-field space in double-simplex slots.
- Use of A-field preamble and Z-field space in double-simplex slots.
- Use of bits from the inter-slot space.
- Introduction of 256 QAM.
- Mixing full and double slots.
- Optimized slot structures.

6.3.2 Use of A-field space in double-simplex slots

Since double-simplex slots does not need to exchange signalling (other than at setup and at release), it may be envisioned using their A-field space for carrying user data. The current uses of A-field at setup and at release would need to be redesigned to use the duplex bearer. B-field signalling may be used over the duplex if needed.

In principle all fields in the A-field of double-simplex bearers may be reallocated for carrying user data. The required signalling information may be passed using the duplex bearer.

By using all A-field bits for U-plane data, **64 additional bits** may be recovered. This equals to 6,4 additional kbit/s per slot. With this improvement the gross data rates are as follows:

Table 4: Data rates for different configurations with A-field improvement

Configuration	**GFSK**	**QPSK**	**8PSK**	**16QAM**	**64 QAM**	**Delay (ms)**
x	1	2	3	4	6	
1 + 1	38,4	76,8	115,2	153,6	230,4	9,6 / 10 / 10,4
1 + (12 + 1)	460,8 + 32	921,6 + 64	1 382,4 + 96	1 843,2 + 128	2 764,8 + 192	0,4 / 0,8 / 1,3
1 + (22 + 1)	844,8 + 32	1 689,6 + 64	2 534,4 + 96	3 379,2 + 128	5 068,8 + 192	0,4 / 0,8 / 1,3
NOTE: This table assumes the conservative option of the duplex slot not being used for U-plane data.						

6.3.3 Use of A-field preamble and Z-field space in double-simplex slots

The use of multiple synchronization words in a multibearer transmission is in theory not strictly necessary. A single synchronization word in i.e. the duplex bearer should be enough since the receiver may always use this reference for the other bearers. The information on the presence and carrier of the double simplex may be transmitted using the duplex bearer.

Z-field bits info may also be used. CRC info to be used for quality control may be send over the duplex or using some CRC inside the U-plane stream.

This allows the recovery of 32 + 8 = 40 additional bits.

The gross data rates would be as follows:

Table 5: Data rates for different configurations with A-field and preamble improvement

Conf.	GFSK	QPSK	8PSK	16QAM	64 QAM	Delay (ms)
x	1	2	3	4	6	
1 + 1	42,4	84,8	127,2	169,6	254,4	9,6 / 10 / 10,4
1 + (12 + 1)	508,8 + 32	1 017,6 + 64	1 526,4 + 96	2 035,2 + 128	3 052,8 + 192	0,4 / 0,8 / 1,3
1 + (22 + 1)	935,8 + 32	1 871,6 + 64	2 807,4 + 96	3 743,2 + 128	5 614,8 + 192	0,4 / 0,8 / 1,3

6.3.4 Use of bits from the inter-slot space

In professional products where cost constrains are not as critical as in telephony, and if the range of the scenarios is restricted to a few hundreds of meters (no WLL scenarios required), it is believed that some extra bits may be recovered from the inter-slot space.

In DECT, the slot space equals to 480 symbols intervals. This means that assuming previous improvements, there are still 56 bits of inter-slot space. This value may be reduced. This would allow further data rate improvements.

Exact data depends on the intended coverage range and implementation constrains.

Assuming that 24 of the 56 symbols intervals available can be reallocated to U-plane data (still leaving 32 bits), the gross data rates would be as follows:

Table 6: Data rates for different configurations with A-field, preamble and inter-slot space improvement

Conf.	GFSK	QPSK	8PSK	16QAM	64 QAM	Delay (ms)
x	1	2	3	4	6	
1 + 1 (duplex)	32 + 32	64 + 64	96 + 96	128 + 128	192 + 192	9,6 / 10 / 10,4
Δ Each simplex	44,8	89,6	134,4	179,2	268,8	
1 + (2 + 1)	89,6 + 32	179,2 + 64	268,8 + 96	358,4 + 128	537,6 + 192	4,6 / 5 / 5,4
1 + (4 + 1)	179,2 + 32	358,4 + 64	537,6 + 96	716,8 + 128	1 075,2 + 192	2,1 / 2,5 / 2,9
1 + (6 + 1)	268,8 + 32	537,6 + 64	806,4 + 96	1 075,2 + 128	1 612,8 + 192	1,3 / 1,7 / 2,1
1 + (12 + 1)	537,6 + 32	1 075,2 + 64	1 612,8 + 96	2 150,4 + 128	3 225,6 + 192	0,4 / 0,8 / 1,3
1 + (22 + 1)	985,6 + 32	1 971,2 + 64	2 956,8 + 96	3 942,4 + 128	5 913,6 + 192	0,4 / 0,8 / 1,3

6.3.5 Introduction of 256 QAM

Another obvious mechanism to increase data rate is introducing 256 QAM. This modulation is used by many other technologies. With 256 QAM data rates would be as follows. Table 7 assumes improvements of clauses 6.3.2, 6.3.3 and 6.3.4 (only 24 bits taken from the inter-slot space).

Table 7: Data rates for different configurations with A-field, preamble and inter-slot space improvement adding 256 QAM

Conf.	GFSK	QPSK	8PSK	16QAM	64 QAM	256 QAM	Delay (ms)
x	1	2	3	4	6	8	
1 + 1 (duplex)	32 + 32	64 + 64	96 + 96	128 + 128	192 + 192	256 + 256	9,6 / 10 / 10,4
Δ Each simplex	44,8	89,6	134,4	179,2	268,8	358,4	
1 + (2 + 1)	89,6 + 32	179,2 + 64	268,8 + 96	358,4 + 128	537,6 + 192	716,8 + 256	4,6 / 5 / 5,4
1 + (4 + 1)	179,2 + 32	358,4 + 64	537,6 + 96	716,8 + 128	1 075,2 + 192	1 433,6 + 256	2,1 / 2,5 / 2.9
1 + (6 + 1)	268,8 + 32	537,6 + 64	806,4 + 96	1 075,2 + 128	1 612,8 + 192	2 150,4 + 256	1,3 / 1,7 / 2,1
1 + (12 + 1)	537,6 + 32	1 075,2 + 64	1 612,8 + 96	2 150,4 + 128	3 225,6 + 192	4 300,8 + 256	0,4 / 0,8 / 1,3
1 + (22 + 1)	985,6 + 32	1 971,2 + 64	2 956,8 + 96	3 942,4 + 128	5 913,6 + 192	7 884,8 + 256	0,4 / 0,8 / 1,3

High level modulations such as 64 QAM or 256 QAM would need FEC. This has to be taken into account in the bitrate calculation. Shown figures are gross bitrates.

6.3.6 Sliding collision detection

In the options described in clauses 6.3.3 and 6.3.4 it may be argued that the removal of the synchronization word may introduce difficulties in the detection of sliding collision cases. The following strategies are proposed for handling the case:

1) Existing systems detect the sliding collisions based on existing mechanisms. No issue is identified here.

2) New systems implementing the described streaming applications detect the collision based on:

   a) For duplex bearers, using the existing mechanisms.

   b) For the new double-simplex bearers using any of the following strategies:

      i) Z-field CRC bits are calculated as a CRC of the n first AND n last bytes of each packet.

NOTE: This strategy does not allow identifying if the sliding collisions happens at the beginning or at the end of the packet.

      ii) Z-field CRC bits are alternatively calculated (in even/odd frames) from the n first and from the n last bytes of each packet.

      iii) there are two groups of Z-field CRC bits: the first group is located at some place close to the beginning of the slot and is calculated from the first n bytes of each packet; second group is located at the end of the slot and is calculated from the last n bytes of each packet.

6.3.7 Mixing full and double slots

Another way for improvement in slot usage efficiency may be the creation of schemes mixing full and double slots. The target combination would be a full-slot duplex bearer combined with double slot double-simplex bearers. An example is shown in figure 10 and table 8. This also reduces the delay due to the single slot interval in the backward direction. The figures for this example would be as follows (without considering any of the previous improvement from clauses 6.3.2, 6.3.3 and 6.3.4).

Example 8

case	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23
8	↓	⇧			⇧			⇧			⇧		↑	⇧			⇧			⇧			⇧	

Figure 10: Example 8: mixing full and double slots

Table 8: Data rates for example 8

Conf.	GFSK	QPSK	8PSK	16 QAM	64 QAM	256 QAM	Delay (ms)
x	1	2	3	4	6	8	
1f + (1f + 8d)	640	1 280	1 920	2 560	3 840	5 120	0,4 / 1,3 / 2,1

For sake of simplicity, the forward full slot has not being taken into account.

As previously, these are gross data rates at transparent IN service and should be reduced by the error.

6.3.8 Optimized slot structures

The ultimate improvement in performance would be achieved by designing new slot structures optimized for the intended use.

The design should take into account the requirements of:

- required delay;
- required data rate;
- protection;
- maximum modulation level usable.

7 Ultra Reliable Low Latency multicast multi-bearer streaming applications

7.1 Overview

The purpose of this clause is presenting a technical proposal for a URLLC (Ultra Reliable Low Latency Communications) multicast multi-bearer service in DECT. The target application of the new service will be high reliability professional and consumer unidirectional streaming applications. Typical example: studio and consumer (including high-end) headphones.

7.2 Basic principles

Taken into account the bitrate needs of the required service and the required timeframe, it is proposed the use of standard DECT PHY layer with High Level modulation. The option 256 QAM and potentially new channel protection schemes (coding) will need to be added to the standard.

Three mechanisms are proposed in this contribution to achieve the high reliability requirement: channel coding, antenna diversity and optional redundant transmission over two different frequencies.

Full slots with some modifications are proposed as fundamental slot structure. They will be used on a multi-bearer C/L downlink arrangement. Full slots have been chosen over double slots due to the low delay requirement.

This contribution proposes new optimized bearer formats with enhanced U-plane capacity. They will be used for data transmission in combination with HLM and channel coding.

The proposal is based on separating U-plane transmission from C-plane transmission on two different logical and physical channels.

This is based on the fact that C-plane does not have the same requirements of bitrate and delay as U-plane. The approach will also allow the use of different modulations for both channels (i.e. GFSK for C-plane and HLM for U-plane).

The control channel, that is only moderately time critical (10 ms delay) will be transmitted using part of the B field of the standard C/L bearers. In near all practical cases, this will done over the dummy bearer. The amount of B-sub-fields required will be, in principle, one subfield. This dummy bearer (or C/L) will be the only bearer transmitting a synchronization word and an A-field.

The U-plane will consist on a number of "new" double simplex bearers in downstream direction and placed on approximatively equidistant positions in order to contribute to the low latency objective. Several configurations are proposed. Some tolerance in the exact "equidistant" slot position is also foreseen as an option (at the expenses of slightly higher delay) to help implementation (i.e. redundant configurations with single radios) and making possible slot handover. See dedicated clause 7.5.3 for details.

Due to the high bitrate requirements, the traffic bearers will be a new type of double simplex bearers. They will be called "double simplexAB" bearers. These bearers inherit all improvements in efficiency previously described in clause 6.3. In short, they will not carry synchronization word, nor A-field and some bits will be taken from the inter-slot space to gain additional bitrate. The preliminary design (based on all optimizations described in clause 6.3) will use 448 symbols per slot, leaving 32 as inter-slot space. Further optimization (by taken extra inter-slot bits) is probably possible.

This number of symbols multiply by the bit/symbol provided by the modulation and by the total number of slots gives the gross bit-rate figure of each configuration that is shown in clause 7.4.1. Note that this is a gross figure and that error detection and error correction will use part of this bitrate.

The U-plane bearers will use HLM with some type of codec (the existing Turbocoding or other specifically designed for the application). Each "double simplexAB" bearer may go on different carriers as standard DECT allows, and potentially a frequency handover may be possible when needed. Slot handover is only possible if the slot tolerance concept is used. See clause 7.5.3. Clause 7.8 gives some introductory ideas to deal with the topic of quality control, what is not trivial in a multi-bearer unidirectional configuration.

To ensure the maximum reliability, simultaneous transmission over two bearers at different carriers is proposed as an option. This transmission may be completely redundant (1 + 1), partially redundant, based on interpolating the sampling rate, or based on a combined coding. The two bearers may be in the same slot (what forces the use of dual receivers at the PP) or, if slot tolerance is allowed, (see clause 7.5.3) they may potentially be at different slots. The first option may be used in professional audio products where latency is more critical and the extra cost is not an issue, while the second option may be of interest for consumer products:

- By partially redundant, it is meant protecting only part of the dataflow. I.e. in a 20 bit or 24 bit PCM transmission PCM, it is more important investing on protecting the initial bits of the sample, while the final bits are probably less critical and may not require the same level of protection or redundancy.
- By based on sampling, it is meant that if a product is designed (and marketed) as supporting 96 ksamples/s transmission, one possible option would be splitting it into two streams at 48 ksamples/s transmitted over two frequencies. The proper algorithms at Rx may interpolate the missed sample in case of error and this would be hard to detect even to experienced listeners. This is described in clause 7.6.4.

Antenna diversity is also proposed as a valid protection mechanism. However, existing antenna diversity at FT side, as implemented in most DECT products, is not useful for a downlink C/L configuration. The effective usage of the space diversity requires dual (or multiple) antennas in the PP side.

7.3 Possible configurations

7.3.1 General

The following set of configurations is defined in order to address a wide range of bitrate and delay requirements. They will be analysed in more detail in further clauses.

conf	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23
0					↓																			
1	⇩				↓								⇩											
2	⇩				↓		⇩						⇩						⇩					
3			⇩		↓		⇩				⇩				⇩				⇩				⇩	
4	⇩			⇩	↓		⇩			⇩			⇩			⇩			⇩			⇩		
5		⇩		⇩	↓	⇩		⇩		⇩		⇩		⇩		⇩		⇩		⇩		⇩		⇩

Legend	
0-11	slot "normally" FT → PT and slot number
12-24	slot "normally" PT → FT and slot number
↓	DECT dummy bearer
⇩	Double-simplex downlink

Figure 11: Compilation of configurations

7.3.2 Comments to the configuration's table

Target configurations for professional audio scenarios are configurations 3 and 4 and perhaps 5. Configurations 1 and 2 may be of interest for consumer electronics. Configuration 5 is an extreme case with 12 bearers downstream. Potentially, there may be a configuration 6 with all bearers (24) downstream, but it introduces several problems. For the time being it is not considered.

The position of the bearers tries to show the concept of "equidistance". In this diagram the further concept of "tolerance" is not yet applied. The exact position of the bearers may change without impacting the properties. However, they should be fixed, or approximately fixed if slot tolerance is applied.

There will be at least one (or more) "regular" dummy or C/L bearer carrying the synchronization word and all C-plane channel used in the design. In normal cases this bearer will be the dummy, what means that no extra bearer is required for this function. The case of combination with ULE requires further analysis (if this case had sense from product perspective).

The dummy bearer may be also at any position, and, in this case, it may change position if needed according to normal DECT rules (bearer handover or replacement but keeping always at least one dummy on the air). Due to the importance of the dummy (since it carries the synchronization word), some additional signalling informing of changes in bearer position will be required.

7.3.3 Remarks on the dummy

One important remark: in the diagram the configurations are designed so the dummy bearer does not collide with any traffic bearer. However, this is not fundamental. In fact some cases such as configuration 5 with slot tolerance, or a hypothetical configuration 6 would require sending two bearers on the same slot. Since transmission is always downlink, this is not an issue: the only consequence is that dual radios are required.

7.3.4 The traffic bearers

The traffic bearers are a new type of double simplex bearers "double simplexAB". Double simplexAB bearers are U-plane only bearers: they will not carry synchronization word, nor A-field (header and tail) and some bits from the inter-slot space are "stolen" to gain additional bitrate.

These enhancement are possible by three reasons:

- There is no fundamental need for transmission of multiple synchronization words. A synchronization word is already transmitted on the "regular" bearer (the dummy). Since all bearers are produced by the same transmitter, they may have an exact timing. See next clause on practical implementation concerns.
- No signalling transmission over such bearers is required. The small amount of signalling required by the overall configuration will be packaged on the special control channel sent over the B field of the dummy. The position of the traffic bearers and any operation such as setup or handovers will be announced on this common control channel. This also removes issues related to identities mismatches: the traffic bearers and their position (slot/carriers) are in fact announced by the control channel, which is a normal DECT transmission and may carry NT signalling in the A-field.
- The design of the DECT inter-slot space was made considering a wide range of cell sizes and cases (such as WLL) and also very low cost implementations. Considering the limited cell size of professional audio systems (compared to WLL) and the improvements by better implementations, this study considers that the inter-slot space can be reduced. This study proposes leaving 32 bits as inter-slot space, which is still a conservative value. See clause 7.3.6 for the equivalence of this space in time.

7.3.5 Notes on the synchronization approach

Some manufacturers have noted some concerns on the clock stability required to receive a whole frame (1 152 symbol spaces) with a single synchronization word. After making some numbers this study concludes that it cannot be a technical issue, even for cost-constrained implementation. Except in some error cases.

Perhaps the only problematic case that would require some detailed calculations is the loss of the dummy on several frames. If this was an issue, it should be noted that the strategy of additional dummy or C/L bearers (with regular structure) in other position of the frame is always possible. This would be a "redundancy in the dummy".

In addition to that, if the FP system implements any other conventional C/O service (i.e. voice) the synchronization word can be used as redundancy.

Also, most chip manufacturers have noted that this solution is not possible with their current chipsets. This is noted, but considering that strong HLM (up to 256 QAM) is also proposed, the conclusion is that new chipsets will be needed in any case.

7.3.6 Notes on the inter-slot space

The equivalence in time of the proposed inter-slot space of 32 symbols is as follows:

Table 9: Inter-slot guard time and equivalence in meters

Inter-slot guard time in "classic" DECT bits/symbols	Equivalence in time (µs)	Equivalence in meters
32	27,77	8 333

The extra bits gained from the inter-slot space may be added at the beginning or at the end of the slot. This is not decided yet.

7.3.7 Notes on sliding collision detection

Some feedback has been raised on the detection of sliding collision and its possible use on antenna diversity:

- Sliding collision at the regular bearers may be detected using today's mechanism.
- Sliding collision in the new "double simplexAB" slots may be detected:
  - Sliding collision at the end, by adding 4 bits or 8 bits CRC calculated over the last octets (similar to existing mechanism).
  - Sliding collision at the beginning by adding 4 bits or 8 bits CRC calculated over the first octets. Such bits may be inserted at any convenient position, i.e. after the n octets.

Such CRCs may be deducted from the available bitrate of "stolen" again from the inter-slot space. E.g. one extra symbol may be used for this function.

Nevertheless, it should be pointed out that antenna diversity as used today (based on switching at the FT) is probably useless in the scenario of the application. In clause 7.6.3 other schemes of antenna diversity are proposed. In general, they require moving the antenna diversity to the PT.

7.4 Available bitrates

7.4.1 Available bitrates possible with the proposed configurations (1 + 1 redundancy not considered yet)

Assuming all previous improvements, and assuming the option of 256 QAM (not yet in the standard) the available bitrates possible for the different configurations are given in table 10.

Symbols per slot: 448

Table 10: Available bitrates for the different configurations

Conf.	slots	GFSK	QPSK	8PSK	16QAM	64 QAM	256 QAM
x		1	2	3	4	6	8
1 (2 ds)	1	89,6	179,2	268,8	358,4	537,6	716,8
2 (2 ds)	4	179,2	358,4	537,6	716,8	1 075,2	1 433,6
3 (3 ds)	6	268,8	537,6	806,4	1 075,2	1 612,8	2 150,4
4 (4 ds)	8	358,4	716,8	1 075,2	1 433,6	2 150,4	2 867,2
5 (6 ds)	12	537,6	1 075,2	1 612,8	2 150,4	3 225,6	4 300,8

These figures assume 448 symbols per slot. This number of symbols multiply by the bit/symbol provided by the modulation and by the total number of slots gives the gross bit-rate figure of each configuration. Note that this is a gross figure and that error detection and error correction will use part of the bitrate.

Redundancy 1 + 1 by using two frequencies (doubling the available bitrate) is not considered yet in the values given in table 10.

7.5 Delay

7.5.1 Delay calculation

The calculation of the delay is slightly complex. It would depend on:

- The configuration.
- The codec type (codec case A, B, C, see clause 7.5.2).
- The possible introduction of tolerances in slot position.

The overall table computing all possible configurations and codec cases is the following:

Table 11: Delays for the different configurations

	Codec case	**A**	**B**	**C**
Configuration	**slots**			
1 (1 ds)	2	4,583	5	5,417
2 (2 ds)	4	2,083	2,5	2,917
3 (3 ds)	6	1,253	1,67	2,087
4 (4 ds)	8	0,833	1,25	1,667
5 (6 ds)	12	0,416	0,833	1,250

As it can be shown, sub-millisecond transmission is in theory possible for configurations 4 and 5 if PCM-like codecs (codec case A) are used.

7.5.2 Codec cases and influence in delay

7.5.2.1 General

The three delay figures depend on the codec operation and are given for three cases, A, B and C. In this case, both the audio codec and the possible error correction code used to protect the transmission have influence.

7.5.2.2 Detailed explanation of the codec impact on the delay.

This clause provides a detailed explanation on when the delay cases A, B or C should be used. They depend on how the samples are inserted and extracted from each packet.

Figure 12 shows an example based on configuration 5 (6 × "double slotsAB"), The principles apply equally to other configurations just changing the primary figure of audio time per sample (0,833 ms for the shown configuration 5).

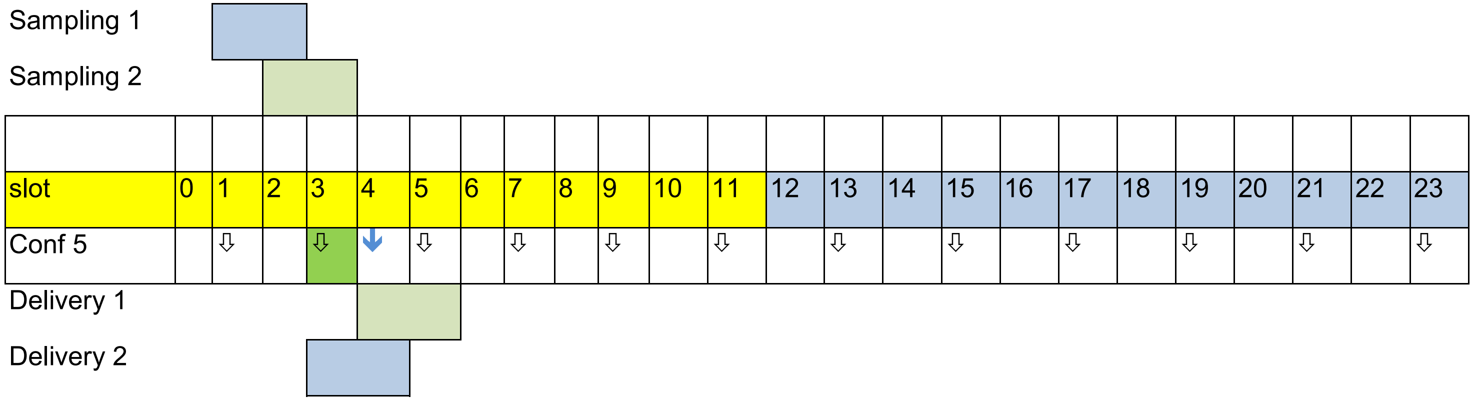

Figure 12: Detailed explanation of delay

Assuming configuration 5 (6 × "double slotsAB"), each packet should contain 0,833 ms of audio samples in all cases, (1/12 of a DECT frame):

- Sampling 1 means establishing the sampling reference at the beginning of the slot. This means that the whole packet of samples is available before start of transmission. This allows the use of any codec and block processing of the whole packet (i.e. FEC, interleaving, etc.). This mode may be required by some codecs.
- Delivery 1 means waiting until the complete reception of the packet before delivering it. Again, it allows block processing of the whole packet (i.e. FEC, interleaving), and also observation of CRC (i.e. for muting purposes).
- Sampling 2 means establishing the sampling reference at the end of the slot (in fact a few bits before). This means that the whole packet of samples does not need to be available before start of transmission. However, all samples will be available at the proper time to allow complete transmission. This is possible if the coded is PCM or derived from PCM and there is no block processing or interleaving. The exact value of the "few bits before" depends on the codec type, codec word size and if CRCs have to be added.
- Delivery 2 means start delivering of samples just "a few bits" after start of reception of the packet without waiting until the complete reception of the whole packet. This does not allow any block processing (i.e. FEC, interleaving), and does not allow observation of CRC (if used) before delivering the samples (i.e. for muting). However, this is possible if the codec is PCM and latency has to be prioritized. The exact value of the "a few bits" depends on codec type, codec word size and size of preambles, separation space and A-field in the slot. CRCs (if used) may be observed and used for handover purposes.

The combination of sampling 1 and delivery 1 gives the worst case figure case C (1,667 ms for the configuration 4 shown in the example). However, this configuration allows the use of any audio codec and full flexibility for block processing of the whole slot. It allows use of FEC and/or interleaving.

However if the audio codec is sampled based (such as PCM, and many PCM derived codecs) it would be possible synchronizing the sampling at the end of the packet. This is "sampling 2" and allows saving 0,4 ms of delay (case B).

At the receiver side, if the codec is sampled based and there is no intention to use/observe any FEC or CRC bits (see note 1), then configuration "delivery 2" may be used. This saves 0,4 additional ms. The combination of delivery 2 with sampling 2 gives the minimum delay figure case A. For configuration 4, case A gives 0,8333 ms delay.

NOTE 1: Regarding sampling 2, the remark on observation of CRC bits (which may prevent muting) refers to the real time use of the stream and CRCs placed at the end of the slot. Others strategies may be possible. It is anyway still possible observing the CRCs and using them for handover decisions.

NOTE 2: The same principles may apply to some types of FEC, that may be usable for correcting. A FEC that only adds redundancy data at the end of the packet may be used with sampling 2. However, delivery 1 should be used to benefit from the FEC. Therefore, this would be the case B of the tables.

Regarding the case of the codec already included in the DECT standard (Turbocoding, MAC service I_{PX}), it seems that this would be the case "C". This is consequence of an interleaving feature ("puncturing") that is intended for improving resiliency against short burst errors. It should be, however, relatively easy designing error correction codings operating in the case "B". On the other hand, going to case "A" would be more difficult and would require FEC processing based on smaller data "blocks" and with the FEC overhead distributed in the slot. This would give reduced FEC efficiency and bad protection against burst errors. However, it would be in theory possible going closer to case "A".

7.5.3 Introducing slot tolerance

The previous design is based on fixed slot positions. By different reasons, introducing some flexibility in the slot positions is desirable. There are several reasons for it:

- Flexibility in administration of the frame if there are other systems in the area.
- Easier implementation of handovers: the FP may observe the target slot and the handover may be implemented with single radios.
- Easier implementation of frequency redundancy 1 + 1 (see clause 7.6.2) that may be implemented with single radios.

Nevertheless, any flexibility adds extra delay. This is consequence of the need for buffering the signal to compensate for any movement in slot position (since all movements should be implemented "seamless").

Two options for slot flexibility have been considered: assuming one slot tolerance or two slots:

- One slot tolerance means that a bearer intended to be at i.e. slot 5 may also use slot 6 and may change between these two slots when required.
- Two slot tolerance means that a bearer intended to be at i.e. slot 5 may also use slots 4 or 6 and may change between these three slots when required.

The impact in the delay is shown in table 12.

Table 12: Delays for the different configurations with slot tolerance

tolerance		**fixed**			**+1 (1 slot)**			**± 1 (2 slot)**		
Codec case		**A**	**B**	**C**	**A**	**B**	**C**	**A**	**B**	**C**
Conf	slots									
1 (1 ds)	2	4,583	5	5,417	5	5,417	5,834	5,417	5,834	6,251
2 (2 ds)	4	2,083	2,5	2,917	2,5	2,917	3,334	2,917	3,334	3,751
3 (3 ds)	6	1,253	1,67	2,087	1,67	2,087	2,504	2,087	2,504	2,921
4 (4 ds)	8	0,833	1,25	1,667	1,25	1,667	2,084	1,667	2,084	2,501
5 (6 ds)	12	0,416	0,833	1,250	0,833	1,250	1,667	1,250	1,667	2,084

As it can be seen, any slot of tolerance adds 0,4 ms extra delay.

The best configuration is obviously no tolerance at all. But it forces dual radios to implement any handover (that should be a pure "frequency handover"), and also adds some complexity to the observation of the target slot x carrier (that cannot be done directly by the FP). Nevertheless, this is the preferred mechanism for professional products with ultra Low Latency requirements.

Tolerance ±1 (2 slot) may be the optimal combination for consumer products with less Latency requirements.

7.6 Redundancy

7.6.1 General

Due to the Ultra reliability requirements for this type of products, and to the expected error mechanism in a mobile NLOS device, two mechanisms are proposed to fight against loss of signal, including selective fading:

- Frequency redundant transmission.
- Antenna diversity with adaptive equalization at receiver.

Both mechanisms should be effective against selective fading. The first one is also effective against interference caused for other systems.

7.6.2 Frequency redundancy

This is the most expensive mechanism (in terms of spectrum) and probably the most effective: in short, the data flow is transmitted over two bearers at two frequencies at the same time. The transmission may be a mere replication or more elaborated schemes. See clause 7.6.4 for an own creative proposal that may be of interest in professional and consumer audio applications.

The double transmission may be:

- Exactly over the same slot, if no slot tolerance is introduced (see clause 7.5.3). In this case the mechanism has no influence in the delay and the delay is minimum. Dual radios should be used in any case, with triple radios required if seamless handover without destroying the redundancy is a requirement.
- Over different slots if the concept of slot tolerance (see clause 7.5.3) is introduced. This may allow implementation using single radios (at the expense of more delay). The redundancy does not add delay per se. Extra delay comes from the slot tolerance.

The channel coding in case of redundancy requires further study, as noted there are multiple possibilities, including combining both flows in the same coding schema.

7.6.3 Space redundancy (antenna diversity) and adaptive equalization

The antenna switch as defined in DECT is not of practical interest in the required scenario due to the downlink transmission combined with multiple (receivers) of the signal. Therefore, FT antenna diversity is not investigated further.

On the other hand, a new PT antenna diversity may be considered as an effective signal protection mechanism. This mechanism can work in the intended downlink-only transmission scenario, and can operate in multi-cast situations. It has the advantage of not having cost in terms of spectrum.

A well designed antenna redundancy mechanism can be implemented without additional standard support. It will be a pure implementation option implemented in the PT radio system. Two antennas separated a few centimetres may provide effective protection against fading considering the intended indoor use.

The implementation is assumed to be based on two receivers at the PT with the two signals combined using DSP. Typically the feature will be used in combination with adaptive equalization to fight against multipath. This scheme is well-known in many other technologies.

7.6.4 Original proposal on "optimized" use of frequency diversity in high performance systems

Taken into account the intended use of the technology in professional and high-end audio products and the slightly effect that the sampling rate really have, the following solution (oriented to professional or hi-end products) is proposed.

It is assumed that a data flow of 20 bits to 24 bits sampled at 96 kHz is required (high resolution audio quality). It is proposed to split the flow in two sub flows sampled at 48 kHz each. Both flows are transmitted over the two frequencies, with the proper codec protection, that typically will protect more the first bits of the sample, with reduced or no protection for bits after the 16.

In case of no errors, a flow sampled at 96 kHz may be received. Therefore, the maximum quality is achieved. The product can be marketed as operating at 96 ksample/s.

If there is an irreparable error in some of the flows, then the intermediate sample may be interpolated from the received 48 kHz flow. This is equivalent to receive a signal sampled at 48 kHz and later upsampled to 96 kHz. Taken into account that this only happens in sporadic frames, and that the 48 kHz / 96 kHz upsampling is already hardly detectable (it is not the purpose of this study entering in this debate), the missed sample will be probably undetectable even for the most experienced listeners.

Therefore with this approach, a product can be marketed as operating at 96 kHz sampling rate, adding frequency tolerance with no real spectrum cost.

7.6.5 Redundancy in the dummy or C/L bearer

7.6.5.1 Redundancy in the dummy or C/L bearer

The dummy bearer may also be subject to redundant transmission. Redundant transmission may be used to fight against errors in the dummy transmission itself. Note that the dummy carries the synchronization word, and several frames of dummy errors combined with poor clock tolerance may cause reception issues.

Currently the DECT standard would allow the insertion of an additional C/L bearer in any downlink frame position replicating the dummy content. However, this would consume extra slots.

7.6.5.2 Original idea

It is proposed as better alternative the use of the opposite position of the dummy (in the normally "uplink" area of the frame) to insert a redundant dummy. Such dummy may be in the same or in different frequency to the "primary" dummy and may be specially marked as "secondary" in the signalling mechanisms.

The "secondary" dummy may be removed temporally if needed (see clause 7.8.3.3.2).

This original idea requires further develop in the standard (it is not possible yet).

7.7 Case example

Considering the case of a high end or studio stereo sets with 20 bits resolution at 96 kHz and 1 + 1 frequency redundancy.

This product can be designed comfortably using configuration 4 (8 slots downstream) and 256 QAM by following the strategy proposed in clause 7.6.4.

The 96 kHz stream is split into two at 48 kHz and transmit each flow on each of the redundant frequencies. The bitrate for each stereo flow at 48 kHz (20 bit) is 2 × 960 kbit/s = 1 920 kbit/s. Configuration 4 with 256 QAM provides 2 867 kbit/s. Therefore, there is a margin of 2 867 / 1 920 = 1,5 to include powerful error correction coding schemas. This is in addition to the redundancy 1 + 1 already described.

It is possible in theory implementing such device with 64 QAM, but then the margin for the channel coding would be only 2 150 / 1 920. It is believed that 256 QAM with 1,5 ratio should provide better results.

The configuration 4 would spend only traffic 8 slots downstream. Therefore, 2/3 of the frame is still available for implementing additional headphones of other audio services.

Regarding the delay, the configuration 4 would provide in theory 0,833 ms if case A is implemented. However, this is not seen feasible due to the channel protection. Case B is considered feasible and is probably the right strategy for the application. This has to be taken into account in channel coding design. Case B design with all FEC redundancy placed at the end of the slot, will provide 1,25 ms delay.

In theory other approached based on splitting the slot in blocks and FEC processing each of them separately may be used to achieve delay figures intermediate between case A and case B. The 1 ms target would probably be achieved even with FEC protection. This would require further study.

7.8 Quality control: a problem approximation

7.8.1 General

Quality control is one of the major issues for any solution based on pure C/L downlink operation. The lack of immediate quality feedback is here mixed with additional issues result of the bitrate and latency requirements. The static or quasi-static slot structure and the frequency diversity, also contributed to the problem.

The solution to the topic is technically complex. This clause provides some basic ideas that may be further investigated.

7.8.2 Analysis of the problem

7.8.2.1 Basic overview

In short the problems can be split into three groups:

1) How the FP may control the quality n the received traffic channels by the different destinations.

2) How the FP may scan the carrier x slot map in order to find suitable candidates for a handover, if required.

3) The implementation of such handover when required.

Item 1 is common to any technical approach based on C/L downlink.

Item 2 is specific of the proposed solution and consequence of the extensive bearer usage and the static or quasi-static slot position.

7.8.2.2 The handover problem

In case of repeated quality problems the standard DECT response is a bearer handover. Such handover may be done changing the slot (and potentially the carrier) of just changing the carrier in the same slot position. The first case will be named "standard bearer handover" and to the last case "frequency handover". Frequency handover is normally avoided in DECT but it is required or convenient in the present solution. It is definitively required if slot tolerance is not introduced.

In DECT there is no fundamental restriction, other than the scanning problem (see next clause), to perform frequency handovers. However, it has implementation consequences: it requires a dual radio in both peers of the link.

Therefore, high performance systems intended to operate with the minimum delay would require dual radio in both FP and PP. If the devices also implement frequency redundancy, then triple radio would be required in both ends (FP and PP).

For cost-restricted devices, the obvious solution would be introducing slot tolerance. This would allow a solution to the handover problem with single radios. See clause 7.8.3.2 for possible solutions.

7.8.2.3 The scan problem

Another problem is the impossibility by the FP to scan channels in the same slot position of an active transmission. This is a fundamental problem and cannot be solved (in practical terms) using multiple radios.

If a FP has an active transmission of slot x, then it will not be able to scan any carrier in this slot due to the "blind" effect caused by the transmission. This problem cannot be solved using multiple radios. Solutions based on diplexers filters (as used in FDD systems) are practically impossible to implement due to the reduced frequency separation between Rx and Tx carriers.

7.8.3 Possible solution paths

7.8.3.1 General

Two solution paths are envisioned:

- Based on introducing slot tolerances.
- Based on collecting feedback from the PP.

Both solutions are not exclusive and both may be used at the same time.

7.8.3.2 Solutions based on slot tolerance

Introducing slot tolerance is the only mechanism that may allow the FP to scan channels and to build a channel map suitable for handovers. Handovers can then be regular bearer handovers or frequency handovers. The first one requires slot tolerance and the last one requires extra radios.

For systems using frequency redundancy, coordination in the handovers of both frequencies may be required to restore the ability of the FP to continue doing channel scanning.

EXAMPLE 1: In a system designed with tolerance +1, the FP may scan channels in the adjacent slot and be ready for a possible handover that may be implemented using a single radio.

EXAMPLE 2: In the same system (with slot tolerance +1), adding 1 + 1 frequency redundancy, the FP may only scan channels if both transmissions are in the same slot. Therefore, this has to be the "stable situation". However, the system can be implemented with only dual radio (and not three), since the handovers may be designed changing the slot. In case of handover, it is required to handover also the redundant transmission to restore the capability of the FP to do further channel scans.

In example 2, note that this may not be the optimal scheme. It would not protect against wideband interference but it is the only solution unless the mechanisms described in clause 7.8.3.3 are introduced.

7.8.3.3 Solutions based on collecting feedback from the PP

7.8.3.3.1 General

Since the FP cannot directly know the quality of the received transmissions, and since it has also limitations for channel scan, the most powerful solutions that can be envisioned consist on involving the PPs in the process.

This solution path represents a variation to the general strategy of a pure C/L design. **Some** capability to transfer data upstream (C-plane data in this case) from PPs to FP has to be added.

7.8.3.3.2 Proposed uplink transmission mechanism.

Several alternatives are possible and may be envisioned. However, due to the required use and the slot usage by other services, this study proposes the following solution:

- Upstream transmission will be done using a pure uplink C/L bearer placed in the opposite position (x + 12) to the dummy bearer.

- The transmission mechanism will be a pure polling system: the FP will interrogate the PPs (by means of a control message send on the downlink C/L channel) and only them, they will reply to the FP sending the requested information.
- The PP reply will be "synchronous" in specific frames mandated by the FP. This simplifies the mechanism removing the "collision" problem (at least internal collision).
- The carrier of such C/L uplink bearer may be typically the same f the dummy. However, it may also be different (then, additional information is required in the downlink "interrogation" message).
- If the idea described in clause 7.6.5.1 is used, the "redundant dummy" should be switched off when an uplink message has to be transmitted.

7.8.3.3.3 Impact on the downlink C-plane channel

The uplink transmission based on polling, would require additional messages in the downlink control channel. It is foreseen that this may require the use of additional B-field subfields. There are a total of four subfields usable by all control requirements, including the command of the uplink polling mechanism.

7.8.3.3.4 Information to be transmitted uplink

It is envisioned that the FP will typically request two types of information from the PPs:

- U-plane quality report: the PP will reply sending information about the received quality on the different bearers (including redundant carriers) during the previous frames. The FP will use this information to decide possible handovers.
- Information on background channel scanning: since the FP has limitations to build a map of unused channels (due to the "blind" effect), one strategy would be instructing some PPs to do that. The PP may use their radio (perhaps multiple radios to build this map). The FP may further request the PP to send upstream its view of the unused channel map. The FP may then update its map based on this information, including channels impossible to scan by the FP itself.

EXAMPLE: Consider the case of operation with no slot tolerance.

7.9 Authentication, subscription and security

This clause gives some ideas that can be further developed to address the issues of authentication, initialization of the service (subscription) and security in general:

- In order to handle authentication, subscription and security, a separate C/O channel (named service channel) will be used. This channel will be set and released when needed.
- The service channel will be used to authenticate the PPs, and potentially to mutually authenticate the FP.
- Authentication algorithm DAA2 will be used.
- Subscription to the multicast channel will be done using signalling operations over the service channel. These operations may be combined with authentication and key allocation operations.
- Encryption of the multicast flow will be done with DSC2 mechanisms. Two alternatives are possible:
    a) Encryption at DLC layer (CCM or CCM-like).
    b) Encryption at MAC layer (DSC2 like):
        - The second alternative seems to be more convenient to be implemented by hardware. It seems to be also more compatible with a synchronous data stream scenario.. However, variations of CCM may also be possible.
        - Due to the intended bitrates, hardware implementation seems mandatory.
        - By security reasons, different IV should be used by the different bearers in the same system.

- Some changes in current standard will be required to handle redundant bearers and to better generate the cipher streams in practical systems.
- It should be noted that the range of LBNs may be insufficient if the concepts of multiple radios and redundant transmission are introduced combined with massive multi-bearer situations.

- The keys may be exchanged using C/O channels (service channel) authenticated and protected with equivalent encryption (DSAA2 and DSC2):
  - Mechanisms similar to the already designed exchange of keys for CCM multicast may be used.
  - Allocation and retrieval procedures are possible.

7.10 Way forward / for further study

This study represents a first approximation to the topic of high reliability professional and consumer unidirectional streaming applications. The study presents a first sketch for a possible solution based on several ideas proposed here by first time. The study does not claim that this is the only possible solution and potentially other solutions may exist. Nevertheless, the study has the merit of proposing the first technical approach and therefore proving that the problem can be solved.

The topic would require a further phase of dedicated study specifically dedicated to URLLC high-bitrate streaming. This study may be implanted as a Technical Report and may be cover solutions based on multicast and solutions based on C/O bearers.

Some identified areas for this study are the following:

1) Complete design of the solution, including investigation and comparative analysis of alternative technical approaches.
2) Capture of specific requirements from industry for specific application scenarios, i.e. a set of definite bitrates, latency and reliability requirements for further developing. It is envisioned that there will be multiple cases with different market targets and bitrates.
3) Study and design of channel protection coding specifically designed for the application. It is assumed that it may be different from existing Turbocoding.
4) Study and design of channel coding schemas to be used in combination with frequency redundancy.
5) Introduction of 256 QAM in DECT or alternative modulations (i.e. Trellis) to further protect the signal and reducing peak-to-average power ratio.
6) Feasibility study of the proposed enhancements in slot designs covering topics such as receiver clock accuracy.
7) Design of the control channel with identification of the required repertory of messages.
8) Design of the uplink transmission mechanism, if used.
9) Investigation of other alternatives such as different C-plane transmission approaches.
10) Redundancy and protection of the C-plane bearer(s).
11) Design of the C/O "service channel"·operation.
12) Study of other audio professional applications, such as conference systems.

It should be noted that topics 3 and 4 and not trivial.

Several topics, such as the study and design of channel coding are technically complex and would require investigation of multiple technical alternatives.

The outcome of this second phase of research should be enough to allow the creation of normative standards in a third step. This phase may be completed by the end of year 2017.

The third step would be the creation of normative standards, This may be implemented by means of a major update of the DECT CI library plus a series of application profiles. For the time being, two application profiles are seen. Tentative titles may be as follows:

- DECT: DECT URLLC; Professional audio applications: part 1: radio microphones and point-to-point services.
- DECT: DECT URLLC; Professional audio applications: part 2: radio headphones and multicast services.

There may also be the option of creating the profiles by application target: one for professional audio, other for consumer audio. However, currently specifications are not mature enough for considering this division.

The normative stage may be conducted and completed during year 2018.

Video streaming, being technically feasible, would require different profiles. They may be under the same multipart deliverable or under a new one. Video applications may require further additions to the DECT CI library and specific studies of topics such as channel protection (coding) or variable bitrate transmission. Technical solutions based on variable bitrate should be investigated. It is not seen as a topic for the same timeframe.

8 High Level Modulation

8.1 General

This purpose of this clause is performing a first analysis of the status of High Level Modulation in DECT and identifying the required work to fully developing this important feature in DECT evolution.

Substantial work related to HLM is already done in current DECT CI, but substantial additional development is required to make HLM a workable practical solution in DECT. Two evident evolution paths are identified, both requiring further study:

1) HLM for packet mode and, in general, variable bit-rate communications. This applies to high speed Internet access and also to video communications with variable bit-rate codecs. In practice most scenarios except those identified as case 2.

2) HLM for high-data-rate synchronous communications with demanding latency requirements that prevents the use of packet mode techniques.

8.2 Overview of current DECT standard

8.2.1 Features already defined in DECT standard

The following features are already defined in DECT standard (ETSI EN 300 175 parts 1 [i.1] to 8 [i.8]).

- Definition of physical modulation modes up to 64 QAM:
  - Physical layer already defined in part 2.
- Channel encoding using turbocodes with variable rate "r":
  - There is one MAC mode called "IP_encoded_protected". However, this mode has some practical limitations that will be described later.
- Basic MAC and NWK control messages to change modulation mode:
  - However, transition procedures are not fully defined.
- There are some good practical ideas, such as mixing HLM in B-field while keeping the 2-level modulation for the A-field:
  - Convenient in order to send control messages if the radio conditions go worse and the HLM information cannot be received.

8.2.2 Features missing in DECT standard (general and packet mode transmission)

The following features are identified as missing in the standard:

- Adaptive modulation and procedures to control modulation changes (proposed in some parts of current CI, but not fully developed):
  - Adaptive means that modulation is dynamically changed according to radio link characteristics.
  - Fundamental to get a workable solution in packet mode and variable bit rate cases.
- DLC layer (LUx) is not designed and is unable to work in an adaptive modulation environment.
- Use of the encoded protected mode with DLC LU10:
  - Today not possible due to lack or error detection capability.
- Lack of MAC ARQ or Hybrid ARQ for encoded protected mode:
  - 3GPP uses both: encoding plus ARQ.
- The so called "I_P_encoded protected" seems to be in fact an "I_N_encoded protected":
  - Error detection capabilities weak and depending of "r". No error detection capabilities if r = 1.
- And some clear errors:
  - Lack of quality feedback for B-field (Q1/Q2 bits) (at least for I_N).
  - No error detection capabilities in Ip_encoded_protected when r=1:
    - => Incompatible with LU10.
  - Lack of definition of adaptive sequences and transitions when modulation a codec r ratio need to be changed:
    - => interoperability issues.
  - Weakness and lack of protection of the B-field E-mux mode:
    - It is transmitted at full rate but without encoding protection.
    - This has been in part corrected in last revision of the standard, for non encoded protected modes. Solution for encoded protected operation is not defined yet. Use in application profiles has to be defined.

8.2.3 Features missing in DECT standard for constant-bit rate low latency applications

Constant-bit rate low latency applications require a different analysis. The low latency makes practically impossible the use of ARQ mechanisms. The constant-bit rate makes hard any change on modulation. This in part simplifies part of the problem, but otherwise it puts extra requirements in the channel coding topic. Channel coding combined with frequency (and potentially space) diversity seems to be the only mechanisms that may be used in a very Low Latency scenarios, Current codec in DECT was not designed for this scenario. Therefore, new codecs would need to be specified and further developed into the standard. They should take into account the specific application requirements (bitrate, protection needs and latency). They may also integrate the protection by redundant transmission using frequency diversity.

In addition to that, it seems that 256 QAM transmission (or potentially other optimized schemas) should be incorporated into the standard. This modulation level is used in many other competing technologies and is highly convenient for implementing the very demanding bitrate requirements of several streaming applications such as professional audio.

8.2.4 Analysis of some limitations in current standard

The current design of HLM in the standard suffers from serious limitations that could jeopardize the practical usability. This study has identified the following limitations:

1) The basic design assumes a quasi-static operation with fixed modulation rate and codec rate ("r"), and lacks of mechanisms to quickly modify these parameters as response to real time changes in the propagation channel. The current mechanisms to do that, are too slow and heavy in signalling operations to be considered an option, especially in the case of multibearer connections. In addition to that, if the modulation and codec rate change involves a variation in the PDU size (what happens in near all cases), there is no way to handle correctly the transition al DLC level.

2) In multibearer connections, the design does not take into account the possibility of having different modulation and codec rates in the different bearers, or changing individually and dynamically these parameters bearer by bearer. This is a severe design drawback since the DECT design and the characteristics of the propagation channel in typical DECT scenarios may involve a completely different channel behaviour for each bearer (due to the different operation frequency).

3) The standard does not provide information about the adaptive sequences that should be travelled through by the modulation states and codec rates as response to changes in the channel (this is related to the lack of an adaptive modulation design).

4) The DLC layer cannot handle properly an operation with mixed PDU sizes or a dynamic modification in the PDU size, if there are segments pending for retransmission.

5) In multibearer connections: there is some ambiguity and limitations in the MAC C-plane control messages (attributes, bearer request) that make unclear the differentiation on attributes change for the whole connection (MBC) or just for the bearer (TBC).

6) The design of the B-field E-mode multiplexer is very weak in case of higher modulation levels, especially when combined with high codec protection schemas (low "r"). The standard mandates the transmission of the E-mode control messages with the same modulation, but without any coding protection (what is very weak for low "r" values).

7) In Encoded protection modulation, the current MAC service does not include internal error detection capability. It means that the error detection capability is limited in all cases, with no error detection capability at all when r =1. Since DECT DLC does not have own CRC and relies on MAC error detection capability, this means that DLC services like LU10 or LU2 cannot operate properly with the current IP_encoded_protected (IPX) service.

8) There is no combination of encoded protected mode with MAC ARQ (Hybrid ARQ). This combination is considered a good design paradigm and is used in many other modern radio transmission technologies (i.e. UMTS, HSDPA, DECT LU7, etc.). DECT uses this design paradigm, in LU7, but not in IP_encoded protected (IPX) service, what seems to be a contradiction.

8.2.5 Some identified possible solutions (list not exhaustive)

8.2.5.1 General

Due to the complexity, the topic, the analysis is split in three parts:

1) solution for no-encoded protected modes;

2) solution for encoded protected modes; and

3) other improvements.

Part 1) was partially implemented in the latest revision of the standard. Parts 2) and 3) are not implemented at all, are technically complex and would require further studies.

8.2.5.2 Proposal of improvements for no encoded protected modes

NOTE 1: It is assumed that 8PSK will be the highest modulation mode usable without channel coding (as stated in current standard).

The suggested design principles are the following:

- DLC layer: Design principle: the DLC will operate with a fixed DLC PDU size independently of the modulation rate. This is a change to current standard solution. This study proposes the use of a PDU size equal to the current DLC PDU size for Ipq service and 2-level modulation. All other modulation modes will transport multiple PDUs of the fixed size. This approach introduces no changes for 2-level modulation, but in turn, forces a new MAC design for all other modulation states. Nevertheless, it is considered the simpler approach.

Comment: this is one of the possible technical alternatives. The objective is decoupling the DLC implementation (LU10) and higher layers from the possible changes in modulation level as response to channel changes. This approach also solves the issue of multibearer connections with different modulation levels (what is sensible from radio point of view).

It may be assumed that the slot size is the same and constant for the whole connection (there are no mixed slots in a multibearer connection) otherwise, the solution would be far more complex:

- MAC U-plane: Two new MAC modes (services) are created with this principle (with and without ARQ). They are preliminarily named Ipz and Ipzr:
    - Ipz: identical to Ipq for 2-level modulation. For any other modulation, Ipz replicates the 2-level slot structure (payload + CRC) n times, being n the modulation level. Therefore each payload segment has always a fixed size (for a given slot), equal to the DLC PDU size.
    - Ipzr: is the error_correct variant of the Ipz.
- MAC C-plane: multiplexers:
    - Redesign of the E and E+U modes for HLM. The proposed design solution for no encoded protected mode is to "slow down" the modulation level when in E or E+U mode to increase reliability.

NOTE 2: Current standard already do that for the A-field (transmitted in BPSK in all cases except some special modes 4a/4b).

Table 13: Proposed "slow down" of the modulation level when in E or E+U mode to increase reliability

Modulation mode (from ETSI EN 300 175-2 [i.2])	S-field modulation	A-field modulation	B+Z-field modulation when in U-mode	B+Z (see note), field modulation when in E or E+U modes.
1a	GFSK	GFSK	GFSK	GFSK
1b	π/2-DBPSK	π /2-DBPSK	π /2-DBPSK	π /2-DBPSK
2	π /2-DBPSK	π /2-DBPSK	π /4-DQPSK	π /2-DBPSK
3	π /2-DBPSK	π /2-DBPSK	π /8-D8PSK	π /4-DQPSK
NOTE 1: The convenience of applying the reduction to the Z field or not, requires further study. NOTE 2: 16 QAM, 64 QAM modulation modes are not usable without encoded-protected operation. Analysis is moved to part C. NOTE 3: Modes 4a and 4b (if used) require further study. A possible solution is not using such modes in data transmission.				

- MAC-C-plane: messages:
  - The messages Attributes_T, Attributes_B and all other messages sending attributes (including the bearer setup messages) have to be thoroughly reviewed identifying and differentiating attributes set at connection level and at bearer level. There will be defined MAC procedures for individually change modulation states at bearer level (bearer by bearer), and for modification of attributes of the whole connection (connection level). This procedure is needed also for double simplex bearers, what imply transmission of the messages on different bearers that the ones to be modified. This may involve the split of the attributes message in two variants: connection attributes and bearer attributes, or to create a new message.
  - In the case of the message for changing modulation state and r value, it will be necessary the creation of a message with capability to change the state of multiple bearers at the same time (in a similar way to the current channel list message). This is necessary for instance for asymmetric connections where there is only on bearer able to transfer MAC messages (the duplex) and there may be the need to reduce modulation state in several double simplex bearers at the same time.
- MAC C-plane, ME and NWK C-plane: sequences:
  - In addition to the list of modulation states, this study proposes the introduction of the concept of "modulation sequences" defined by standard. Each "modulation sequence" will define a highest modulation state (i.e. 8 PSK), a lowest state (typically BPSK) and all intermediate states. In the case of no encoded protected operation, and assuming 8 PSK as highest state, the sequence is trivial (only one), but the concept has to be introduced here since for encoded-protected operation there are many possible combinations.
  - The importance of the sequences is that they allow both peers to know what is the transition initiated or expected by the other peer as response to channel changes.

8.2.5.3 Proposal of improvements for encoded protected modes

8.2.5.3.1 Analysis

NOTE: This topic is far more complex than the no-encoded protected modes and this is only a first draft. Further work will be needed.

The suggested design approach is based on the following:

MAC: complete redesign of the MAC encoded protected modes (I)

The first step is the redefinition of MAC services for encoded-protected operation. The current service (named Ipx after last standard revision) is in fact an "In encoded protected". This mode may be useful, for instance, for transmission of voice with coding protection, however, is not usable for data application where there is an stack on top of it with retransmission capabilities (i.e. LU10). The reason is lack of error detection capability of the residual error after the codec correction, or in case of r ratios close to 1, lack of error detection capability at all. The problem is evident choosing r=1. Note that in DECT the DLC does not have its own CRC, but relies on the CRC of the MAC layer.

The solution to the issue is introducing additional CRCs inside the payload of the codec.

The current mode IPX may be useful for certain applications (but not for packet data). The proposal is to keep it (changing the name to I_{NX}) and creating new modes with error detection capability.

Regarding the INX, there may be a debate about if it has to be minimum or normal delay (both cases are possible and have pros and cons depending on the application). Current Ipx was defined as minimum delay in last MAC review (previously there were ambiguity). However, this review may be the right time to re-discuss the topic. The best solution is allowing both modes:

- Current Ipx > Ixna (I_N encoded protected, minimum delay).
- New mode > Ixnb (I_N encoded protected, normal delay).
- New mode > Ixp (I_P encoded protected with residual error detection).

DLC design

- By analogy with the solution for no-encoded protected cases, the solution proposed by this study is operating the DLC with fixed PDU size. This automatically de-couples the physical and MAC layer problematic (selection of modulation state and r ratio) from higher layers: DLC always sends and receives PDUs of fixed size to the MAC.
- On the other hand, this makes more complex the design of the MAC: the MAC frame carries a variable number of segments (PDUs) in every bearer depending on the modulation state and codec r ratio.

Here there are two important design decisions:

- The size of the PDU (fixed size).
- If individual CRCs for each PDU are added, or just a CRC for the whole slot (when the slot carries several PDUs).

For the size of the PDU, a possible solution (by analogy with the no-encoded protected solution) is choosing it equal to the payload size of the slot with 2-level modulation and r=1. For other modulations a variable number of PDUs would be transported according to modulation or r ratio:

EXAMPLE 1: For QPSK, r=1 ⇒ 2 PDUs would be carried:

- For 16 QAM, r = 1 ⇒ 4 PDUs
- For 16 QAM r= 1/2 ⇒ 2 PDUs
- For 16 QAM r= 3/4 ⇒ 3 PDUs
- For 64 QAM r=1 ⇒ 6 PDUs
- For 64 QAM r=5/6 ⇒ 5 PDUs

The drawback of this solution is that there is no possibility to use r < 1 in BPSK mode.

Alternatives may be using a different PDU size (i.e. equal to ½ of the BPSK mode slot) or redesigning the DLC with a different paradigm.

Another drawback is the inefficiency if the r ratio does not match with an integer number of PDUs per slot. In practice this means that the codec properties have to be taken into account to design the elementary PDU size.

The selection of the elementary PDU size has to be completed in the next study stage.

MAC: redesign of the E and E+U multiplexers modes (encoded protected modes)

One of the weakest parts of the current DECT solution for encoded-protected mode is the solution given to the E multiplexer mode. Current standard forces the use of the same HLM modulation type used in U-mode slots but without any coding protection.

EXAMPLE 2: U-mode is using 64 QAM and codec with r ratio = 1 / 2. When the slot changes to E-mux mode it should use the same 64 QAM but without coding protection.

The result is extreme weakness of the C-plane transmission using B field (E-mux), which may suffer a very high error ratio.

There are two possible approaches:

- The most convenient solution would be protecting by the codec also the E or E+U modes. This would be a complete redesign of the E/U multiplexer for encoded services. This solution may be combined with reduction of the modulation mode and/or variation of the r ratio (increasing the protection). This solution is complex and would require many changes in the MAC standard. It would be also necessary to study the interleaving pattern of the turbo coding, in order to be sure it effectively protects control messages sent on a slot subfield.
- An alternative solution, less effective but far simpler, would be mandating a "slow down" of the modulation (changing to a more robust modulation) when the slot switches to E or E+U mode.

In the first solution, it will be necessary to redefine the different E and E+U type mux modes and formats for all HLM states and slots, taken into account that now the number of transported subfields would depend on the codec "r" ratio. A potential high number of modes and variants will occur.

The previous result is also influenced by the strategy on regard to the extra protection introduced for the E/E+U mode (reduction of modulation rate and/or reduction of "r" rate.

After completing previous stage, a list of HLM modes will be got, each one characterized by a main modulation mode (when in U-mux mode), main codec "r" rate, modulation mode when in E/E+U mux, and "r" rate when an E/E+U mux.

MAC: definition of states (for encoded protected modes)

This is a relatively complex task composed of several sub-tasks:

- The first part of the task is the identification of a finite number of states (characterized by a modulation mode and r ratio), conveniently chosen according to slot uses and codec correction properties.
- The second task is the assessment of the robustness of each state. This task requires building a simulation model with a channel error model.

Considering all HLM modes and possible r ratios, the number of potential states (initially very high if all "r" values are allowed) should be reduced to around 16 to 32 (depending on the strategy for PDU size) per slot type.

For each state, the format and parameters for the E mode slots are defined (complexity depending on the chosen solution).

With the current structure of the MAC standard, a table for E and E+U mux formats should be created for each state and slot type. This means around 48 to 96 tables assuming that only full, long and double slot formats are supported.

MAC: definition of sequences

The next step is the identification of sequences. Each sequence is characterized by a maximum state (highest throughput, minimal protection) and a lowest state (maximum protection. The sequences should be designed according to the valuation of the robustness and it is not evident that the minimum state (maximum robustness) is the BPSK, r=1.

The sequence will define at the end the strategy for changing modulation and/or r in case of variations in the radio conditions.

Furthermore, it is possible that the optimal sequences are different depending on the channel model used in the simulation (i.e. random errors vs. burst errors, ch. dominated by multipath vs. distance, etc.). Thus, several sequences may exist even for a given maximum modulation state.

Technology implications should be taken into account. I.e. the support of QAM modulation is more expensive in terms or radio components that only PSK.

At the end, a repertory of sequences would be created.

Example of sequence (example values, not really analysed with a simulation model).

Table 14: Example of sequence combing modulation order with codec ratio r

Sequence code: 1, PDU size= size of the slot with 2 level modulation				
State code	**Modulation in U mode (B +Z field)**	**codec "r" ratio**	**Modulation in E or E+U mode**	**codec "r" ratio**
1111	64 QAM	1	tbd	tbd
1110	64 QAM	5/6		
1101	64 QAM	4/6		
1100	64 QAM	3/6 (=1/2)		
1011	16 QAM	3/4		
1010	16 QAM	2/4		
1001	16 QAM	1/4		
1000	4 QAM	1/2		
0111	BPSK	1		

Comment to the example: this is an example to introduce the concept of sequence. However, it is not "real". For instance, it is unclear that BPSK with r=1 provides more robustness than QPSK with r=1/2.

The example also assumes than the basic PDU size is equal to the size of the slot with 2-level modulation. This is also an open topic:

MAC: Introduction of Hybrid retransmission

Hybrid retransmission is the name used in many radio technologies for the combination of coding protection plus MAC ARQ. Hybrid means that the result of several retransmissions, none of them completely perfect, may be used to build a correct reconstruction of the codec payload.

This concept is similar to what in DECT is called "selective retransmission", an existing optional feature of the DECT Ip_error_correct service. The only difference is the addition of codec protection.

Hybrid ARQ is used in many radio technologies: f.i HSDPA (UMTS), and in DECT itself, it is used in LU7.

This study proposes the addition of Hybrid protection to DECT by the creation of a Ip encoded protected + ARQ mode by combining the new Ip_encoded_protected mode with the existing MAC ARQ mechanism (Ip_error_correct). The basic design would be as follows:

- A new MAC service: Hybrid mode is created (encoded protected + error correct).
- The format of each slot is identical to the Ip_encoded_protected mode: same format and same possibilities of codec ratios.
- Generation and decodification of each slot is identical to the Ip_encoded_protected mode.
- On top of the codec protection, the channel is also protected by a MOD2 retransmission schema with the existing Ip_error_correct mechanism. The ARQ mechanisms operates if the codec has not been able to get a correct payload.
- Selective retransmission would be possible (as today). Several receptions would be used by decoder to get a correct slot, even if none of them has been fully correct.
- MAC packet lifetime and BA-field headers will be as existing iP-error_correct.

The hybrid retransmission provides protection to random errors (by the codec) and to burst errors (by the ARQ) and is generally accepted as a good protection paradigm.

Notes on retransmission schemes.

The introduction of MAC Hybrid retransmission is not in contradiction to the existence of DLC retransmission: both mechanisms operate at different levels.

8.2.5.3.2 Pros and cons of ARQ schemas

MAC (Hybrid):

- Fast mechanism (intended to retransmit a packet in next frame).
- Implementable by hardware.
- Combinable with codec design (Hybrid concept).
- Intended to operate with a finite lifetime (if retransmission does not success after x frames, then it will waive to retransmit the packet).

Higher layer retransmissions (DLC, TCP):

- Slower response time (no next frame).
- Larger lifetime (transmissions windows).
- Intended to be implemented in software.

- Not connectable with the codec hardware.

The combination of protection and retransmission schemas a best practice used in many radio transmission technologies.

EXAMPLE: The codec corrects the 90 % of the errors, the MAC ARQ (implemented by hardware) the 9 %, and the 1 % remaining is passed to the DLC.

8.2.5.4 Other improvements

8.2.5.4.1 Increasing of the modulation to 256 QAM

An additional option is to increase the maximum modulation level to 256 QAM instead of 64 QAM.

Rationale: the PHS (ARIB standard 28) already supports 256 QAM as highest state. This mode is also used in Next Generation PHS (OFDMA, ARIB standard T95). There is no reason why DECT is one step behind.

The introduction of 256 QAM in the ETSI EN 300 175-2 [i.2] is not a very complex task and provides automatically an increasing of 8/6 in all maximum rates supported by DECT. Taken into account that all HLM modes will be implemented using digital base band modems and I/Q modulators, the implementation cost of 256 QAM would be in practice zero or very small compared to 64 QAM.

The new modulation mode should be first created in ETSI EN 300 175-2 [i.2]. Once the ETSI EN 300 175-2 [i.2] is updated including the new modulation mode, all other parts of the base standard would need to be updated accordingly (MAC, DLC and NWK).

This change would have an impact in the radio test standards ETSI EN 300 176-1 [i.11], ETSI EN 301 406 [i.24] and ETSI EN 301 908-10 [i.37]. The impact is minimum, but the three standards would require an update to include the new mode.

9 Long term evolution of DECT

9.1 DECT OFDM evolution

9.1.1 Overview and technology positioning

9.1.1.1 Basic principles

OFDM has been identified as the technology approach for the long term evolution of DECT.

Basic technology principles:

- Operation over License exempt spectrum.
- State-of-the-art bandwidth efficiency:
  - OFDM for the downlink.
  - OFDMA and SC-FDM options for the uplink.
- Spatial multiplexing (MIMO).
- Better availability and quality of service than Wi-Fi [i.21].
- Better multipath protection than Wi-Fi.

9.1.1.2 Target application and scenarios

The following scenarios have been identified as targets:

- Medium range, high capacity, multimedia cellular system able to carry audio, video, and data communication.
- Main target scenario are private systems operating over unlicensed spectrum.
- Initial target segments:
    - Corporate communications, including industrial automation.
    - Including: all kind of corporate buildings, factories, airports, railways, harbours, government, etc.
    - Wireless local Loop.
- Public cellular system not envisioned as a target scenario.

9.1.1.3 Comparison with other technologies

9.1.1.3.1 Comparison with Wi-Fi

Compared to Wi-Fi [i.21], DECT OFDM will offer:

- Larger size of the cells (1 DECT advanced cell = x Wi-Fi cells).
- Different design of the MAC.
- Lower subcarrier spacing.
- Higher tolerance to multipath.
- Better quality of service.
- Better spectrum efficiency.
- MAC will be based on scheduled algorithm (as LTE, WiMAX) and not on contention (Wi-Fi).
- Better quality of the spectrum.
- More exclusive use of the spectrum:
    - On the other hand, it is not the intention of the technology to compete with Wi-Fi in maximum data rate achievable on small size cells.

9.1.1.3.2 Differences with LTE and WiMAX

Compared to WiMAX [i.22] and LTE [i.25], DECT OFDM will offer:

- Higher data rate for a given complexity (smaller FFT size).
- Less hardware complexity (FFT size) for a given bandwidth and data rate.
- Higher capacity and data rates. Sustainable in the long term.

At the expense of:

- Smaller size of the cells.
- Lower tolerance to multipath due to larger subcarrier spacing.

9.1.2 Basic specifications

The following basic specifications are assumed as working hypothesis:

- Technology will be based on OFDM plus MIMO:
  - OFDM in the downlink.
  - Option of OFDMA and SC-FDM in the uplink.
- The peak downlink data rate should be 1 Gbit/s.
- MAC should be based on scheduled algorithm (LTE, WiMAX) and not on contention (Wi-Fi).
- Technology will be able to operate over existing DECT band:
  - Technology will be back compatible with DECT.
- In addition to DECT band, technology may use other expansion bands:
  - For data rate calculation, it will be assumed that at least 20 additional MHz are available.
- Spectrum use model will be license exempt:
  - DECT operation model should be possible (customer buys and owns system, and get the right of use of the spectrum without any further authorization).
- Technology should not require radio planning (self-planning or Dynamic channel by the technology itself).
- Technology will be TDD.
- Time structure will be 10 ms TDD, with 24 basic slots (DECT frame structure).
- Frequency structure will be based on 1 728 MHz (DECT channel). Slightly variations as 1 725 MHz are acceptable.

9.1.3 Proposals for the physical layer

The following proposals of subcarrier spacing have been investigated:

- Subcarrier spacing: 108 kHz.
- Subcarrier spacing: 75 kHz.
- Subcarrier spacing: 54 kHz.
- Subcarrier spacing: 45 kHz.
- Subcarrier spacing: 37, 5 kHz (as in XGP / NG-PHS).
- Subcarrier spacing: 30 kHz and 32 kHz.
- Subcarrier spacing: 27 kHz.
- Subcarrier spacing: 15 kHz (as in LTE).

Within them, the 37,5 kHz, 27 kHz and 54 kHz proposals have been investigated in some detail. The other proposals will be listed at the end of the clause (clause 6.3.5) with their comparative differences.

9.2 The 37,5 kHz subcarrier spacing proposal

9.2.1 Overview

The value 37,5 kHz for the subcarrier spacing has been investigated in some detail. The origin of this figure comes from the Japanese work on XGP (eXtended Global Platform), formerly known as Next-Generation PHS [i.23]. The value is also a multiple (× 5) of the half-step value used in 3GPP LTE (7,5 kHz when the reduced subcarrier value option is used).

The use of exactly the 37,5 kHz value produces a mismatch with the existing DECT channelization. This mismatch may be avoided by using a value of 37,5652 kHz. However, after some calculations, it can be proven that the mismatch when using 37,5 kHz would be of only 3 kHz per DECT channel, leading to an accumulated drift of 15 kHz at the edge of the DECT band (assuming an exact match at the edge of DECT channel 5), or 12 kHz at the border between DECT channels 0 and 1 (or 8 and 9). This value is less than the subcarrier spacing and is irrelevant taking into account the roll off factor of current DECT channels.

This proposal develops the scenario for 37,5 kHz with a design that achieves a perfect matching with the current DECT time structure. Realistic and convenient values are got for all relevant parameters, such the CP ratio and the number of symbols per slot. The design considers the case of multiple slots (see clauses 9.2.5.1 and 9.2.5.2) and includes an extended CP option for operation under problematic propagation scenarios (as LTE does).

It should be remarked that the IMT-Advanced target data rate of 1 Gbit/s is achievable with a FFT size of only 1 024 (see table 15), what is an advantage compared to other IMT-advanced proposals.

9.2.2 Basic parameters for the frequency structure

Basic parameters:

- Subcarrier spacing: 37,5 kHz.
- 46 subcarriers per current DECT channel:
    - One DC and one guard subcarrier.
    - DC carrier is placed at the central frequency.
    - Guard subcarrier is placed at the edge between two carriers.
- Basic channel: 1,725 MHz:
    - 37,5 kHz × 46 = 1,725 MHz.
    - There is a mismatch of 3 kHz compared to existing DECT carrier structure (1,728 MHz).
    - Exact match with current DECT carrier grid at edge between current DECT channels 4 and 5 (1889,568 MHz).
    - Maximum accumulated mismatch at central frequencies is 13,5 kHz and happens at (current) channels 0 and 9.
    - Not an issue at all since it is within current tolerances.
    - Current LO accuracy specification in DECT "classic" is ± 50 kHz (see ETSI EN 300 175-2 [i.2], clause 4.1.2).

9.2.3 Channelization and matching with current DECT 1,728 MHz channels

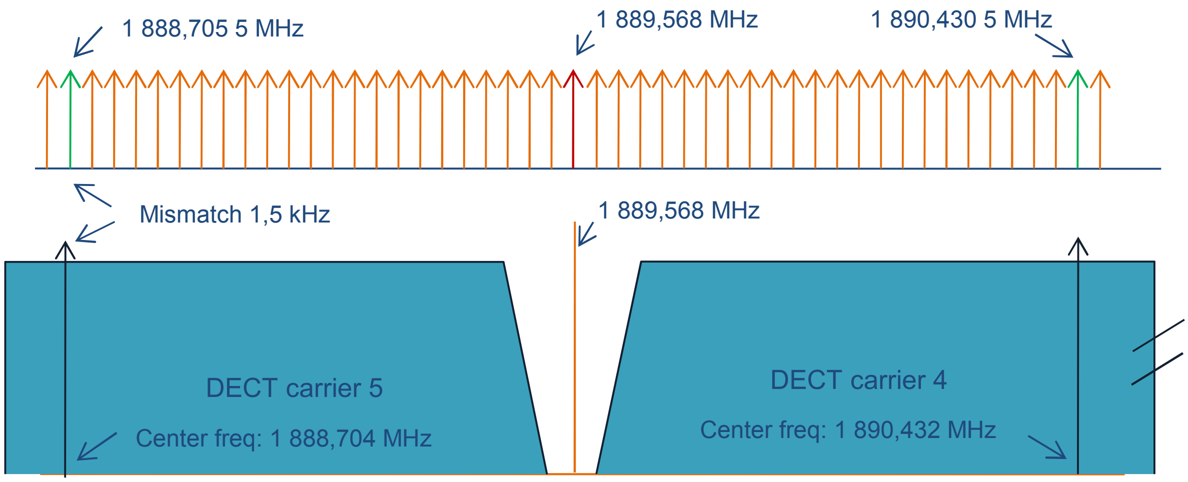

Figure 13: Explanation of carrier grid matching between DECT OFDM and DECT "classic"

See table A.1 for carrier positions in clause A.1.

9.2.4 37,5 kHz proposal basic parameters

Table 15: 37,5 kHz proposal basic parameters

Parameter	DECT-Advanced					
	Type 1	Type 2	Type 3	Type 4 (base DECT band)	Type 5 (DECT w/ extension band)	Type 6
Sampling frequency	2,400 MHz	4,800 MHz	9,600 MHz	19,200 MHz	38,400 MHz	76,800 MHz
Usable channel bandwidth with DECT channelization	1,728 MHz (1 DECT carrier)	3,456 MHz (2 DECT carriers)	8,640 MHz (5 DECT carriers)	17,280 MHz (10 DECT carriers)	38,016 MHz (22 DECT carriers)	76,032 MHz (44 DECT carriers)
Subcarrier spacing	37,5 kHz					
Subcarriers per DECT channel (1,728 MHz)	46 (44 + DC + guard)					
Total nr. of subcarriers	46	92	230	460	1 012	2 024
FFT	64	128	256	512	1 024	2 048
Transform period (T_{FFT})	$(37,5\ kHz)^{-1}$ = 26,67 µs					
Frame interval	10 ms					
Slots per frame	24					
Samples @ 2,4MHz per slot	1 000					
Slot interval (including guard spaces)	10 ms / 24 = 416, 67 µs					
Symbol per slot (including guard spaces), see note	Standard CP option		Extended CP option (t.b.d.)			
	14		13		12	
Samples @ 2,4 MHz per symbol	71		76		83	
Additional slot guard space (samples @ 2,4 MHz), note 1	6 (3 + 3)		12 (6 + 6)		4 (2 + 2)	
Samples @ 2,4 MHz per slot (+ slot guard space)	3 + 994 + 3		6 + 988 + 6		2 + 996 + 2	
Symbol interval (T_{SYM})	414,27 / 14 = 29,58 µs		411,67 / 13 = 31,67 µs		415,00 / 12 = 34,58 µs	
Guard interval (cyclic prefix)	29,58 - 26,67 = 2,91 µs		31,67 - 26,67 = 5,00 µs		34,58 - 26,67 = 7,91 µs	
Guard interval (CP) in samples @ 2,4 MHz	6		11		18	

Parameter	DECT-Advanced					
	Type 1	Type 2	Type 3	Type 4 (base DECT band)	Type 5 (DECT w/ extension band)	Type 6
Symbol w/out CP in samples @ 2,4 MHz	65		65		65	
Guard interval ratio	1 / 9,165		1 / 5,334		1 / 3,372	
Multipath tolerance in meters	873 m		1 500 m		2 373 m	
NOTE: An alternate option would be extending the CP of the first symbol using these samples.						

9.2.5 Slot Time Structure

9.2.5.1 Structure of single and multi-slot blocks

In DECT-advanced it is proposed that the DECT concept of double slot is generalized to any number of slots in sequence.

In a single slot connection, it is proposed that the space equivalent to ONE symbol is used as guard space and optionally to increase the GI ratio in the second symbol (first transmitted symbol). This space is assumed to be at the beginning of the slot. Therefore, the slot would carry 14 user symbols (with std. CP) and 13 or 12 with extended CP.

In multi-bearer connection all slots in sequence are considered an n^{th}-ple slot. Only the first slot would carry 13 symbols (std. CP) and all the others would carry 14 symbols.

If extended CP is used, then the first slot will carry 12 or 11 symbols and the subsequent slots will carry 13 or 12 symbols.

9.2.5.2 Inter-slot guard space

Based on previous work done by STF 518, it is proposed to set the guard space as equivalent to 1 OFDMA symbol. This is slightly less than current DECT, but seems to be large enough for all intended applications.

The proposed inter-slot guard space equals to:

Table 16: 37,5 kHz proposal slot structure

Inter-slot guard space (see note)	OFDM samples @ 2,4 MHz	guard time calculation	guard time (µs)	Equivalence in "classic" DECT bits/symbols	Equivalence in meters
Standard option (14 symbols/slot)	71 + 6	29,58 + 2,50 = 32,08 µs	32,08	36,96	9 625
Std option with double CP in the 2^{nd} symbol	71 - 6 + 6	29,58 - 2,50 + 2,50 = 29,58 µs	29,58	34,08	8 875
Extended CP option (13 symbols/slot)	76 + 12	31,67 + 5,00 = 36,67 µs	36,67	42,24	11 000
Extended CP option (13 symbols) with double CP in the 2^{nd} symbol	76 - 11 + 12	31,67 - 4,58 + 5,00 = 32,08 µs	32,08	36,96	9 625
Extended CP option (12 symbols/slot)	83 + 4	34,58 + 1,67 = 36,25 µs	36,25	41,76	10 875
Extended CP option (12 symbols) with double CP in the 2^{nd} symbol	83 - 18 + 4	34,58 - 7,50 + 1,67 = 28,75 µs	28,75	33,12	8 625
NOTE: The additional slot guard space is added to the inter-slot guard space. Another option for this additional slot guard space is using it to extend the CP of the first transmitted symbol. Figures will slightly change (this is not shown in the table).					

9.2.6 Control plane multiplexing (signalling)

9.2.6.1 General

Insertion of channels for DECT signalling requires further study. It is unclear if at this stage if keeping existing A-field / B-field format in OFDMA slots is the best approach.

Nevertheless, as this option would need to be studied in any case, the next clauses provide a possible design and bitrate calculations for a possible A-field / B-field format.

9.2.6.2 Possible C-plane multiplexing based on A-field / B-field mux schema

If DECT standard A-field / B-field mux schema is used, a simple proposal would be using the second symbol in an initial slot for creating an A-field in a similar way to standard DECT. This symbol may have a different (stronger) coding protection that the others. It may also be protected by an extended CP (already described).

With this format the initial slot will carry 1 symbol for A-field and 10 or 9 symbols for B-field data.

No symbol will be used for this use in subsequent slots in a multislot burst. Therefore, they will carry 12 or 11 symbols of data.

The capacity of this 2nd symbol is the following:

Table 17: 37,5 kHz proposal 2nd symbol capacity

64 QAM	6 x 44	264 bits
256 QAM	8 x 44	352 bits

Assuming 2 subcarriers not usable (used for guard space or DC).

As the capacity seems large enough for inserting current DECT control channels, it is proposed that part of the capacity in this symbol may be used for the beginning of the U-plane data, and in turn, part of the last symbol in the slot or in a multi-slot burst may be used for CRC. for U-plane.

The proper coding may be used for the second slot carrying signalling and it may be more conservative than the used for the U-plane data.

Special rules would be necessary for optimal coexistence with standard DECT transmissions operating in the middle of the OFDMA block.

9.2.6.3 Capacity of the B-field

Assuming the described A-field / B-field mux schema and not considering possible spare bits from the A-field, the gross capacity of the B-field is the following.

NOTE: The capacity used by preamble symbols (synchronization strategy not defined yet) should be deducted from the given gross rates.

Table 18: 37,5 kHz proposal B-field capacity

Standard CP option 14	First or single slot	64 QAM	12 x 6 x 44	3 168	bits
		256 QAM	12 x 8 x 44	4 224	bits
	Subsequent slots in a multislot burst	64 QAM	14 x 6 x 44	3 696	bits
		256 QAM	14 x 8 x 44	4 928	bits
Extended CP option 13	First or single slot	64 QAM	11 x 6 x 44	2 904	bits
		256 QAM	11 x 8 x 44	3 872	bits
	Subsequent slots in a multislot burst	64 QAM	13 x 6 x 44	3 432	bits
		256 QAM	13 x 8 x 44	4 576	bits
Extended CP option 12	First or single slot	64 QAM	10 x 6 x 44	2 640	bits
		256 QAM	10 x 8 x 44	3 520	bits
	Subsequent slots in a multislot burst	64 QAM	12 x 6 x 44	3 168	bits
		256 QAM	12 x 8 x 44	4 224	bits

Two subcarriers have been assumed as not usable (used for guard space or DC).

The proper coding should be used to protect the data.

9.2.7 Frequency and mask considerations for back-compatibility with DECT "classic"

An OFDMA channel composed of 44 + 2 subcarriers, intends to use as much as possible the DECT channelization in order to achieve spectrum efficiency. On the other hand "classic" DECT was designed with relaxed parameters (GFSK and roll off = 0,5) to allow low cost implementations of single carrier radios. Therefore, a DECT OFDMA transmission using all subcarriers as defined in the frequency structure, cannot fulfil the current DECT spectral mask.

The following rules are proposed:

1) DECT legacy band 1 880 MHz to 1 900 MHz:

   1a) When an OFDM radio operates over a single carrier, some subcarriers close to the channel edge would need to be switched off in order to fulfil DECT spectral mask. This will have an impact on data rate and codec protection level that has to be taken into account.

   1b) Nevertheless, DECT OFDMA transmitters will typically have a better accuracy in LO frequencies compared to DECT "classic". DECT "classic" has an accuracy of ± 50 kHz (see ETSI EN 300 175-2 [i.2], clause 4.1.2 and clause A.2), a relaxed value intended to allow low cost implementations. This has to be taken into account in the design of the "compatibility mask" that may be slightly wider than existing DECT mask due to the better LO accuracy.

   1c) When an OFDM radio operates over several consecutive carriers (wideband transmission), only the edge carriers and only in the edges of the transmission bandwidth, would be impacted by the previous rule. This means that edge subcarriers would not need to be switched off if the adjacent carrier is also part of the OFDMA transmission.

   1d) In the case of RFPs, the rule 1c applies even if the transmissions are addressed to two or more different PPs.

   1e) It is understood that the correct implementation of rule 1b and 1c would need some modifications in some applicable ENs, such as the ETSI EN 301 406 [i.24]. However it is believed that there will be enough technical justification for it.

2) DECT expansion band 1 900 MHz to 1 920 MHz:

   2a) The spectral mask for the new expansion band may be designed optimized for and favouring OFDM transmissions. Therefore, a more demanding mask would be used. Ideally, the target would be that the mask should allow operation with only one subcarrier as guard space (as defined in the basic frequency structure). If this is not possible or economically convenient, one or more additional subcarriers close to the transmission edge may be switched off as done in rule 1a. However, the number of lost sub-carriers may be less than in the legacy band case due to a more "OFDM-friendly" mask.

9.2.8 Further improvements in the 37,5 kHz approach

9.2.8.0 General

Taken into account the frequency and time structure and the relatively high bitrates obtained with a single full slot, two improvements are proposed:

1) Use of half-carrier channels.

2) Use of half slots.

As the calculations will show, both options are possible and relatively easy to implement with the 37,5 kHz approach. The figures will show that a single half-carrier / half-slot channel may carry a voice communication with its associate signalling and a huge (> x2) extra bitrate to allow effective codec protection The result is **a gain in capacity of 4 times** for ADPCM voice compared to DECT "classic".

The numbers will also show that a wideband voice channel coded at 64 kbit/s (such as the relatively inefficient G.722) can also fit into a new half-carrier/half slot channel. This would allow **a gain of 8 times** compared to DECT "classic". However, here the margin for additional coding protection is not as large as in previous case (however, still 8 / 6 for 256 QAM). It should be okay in case of good radio conditions but it may not be enough in poor radio situations. Here, the system may be forced to choose between moving to a full slot or to a full carrier (any of them should be enough), or switching down to non-wideband voice. Another possibility may be moving some bits from the signalling field (A-field) to reinforce the B-field capability. No issue at all is expected if the wideband voice signal is coded with a more efficient codec (bitrate < 48 kbit/s).

9.2.8.1 Use of half-carrier channels

It is proposed to split each DECT OFDMA carrier into two sub-channels: lower and upper. Each one will have 23 subcarriers.

If this option is used, the former DC subcarrier 23 will have to be used as guard sub-carrier, at least in the uplink. A new subcarrier will need to be used as DC carrier for the uplink in each of the blocks. Therefore, the number of usable subcarriers for the uplink in each block is 21.

For the downlink it is believed that a single shared DC on subcarrier 23 would be enough.

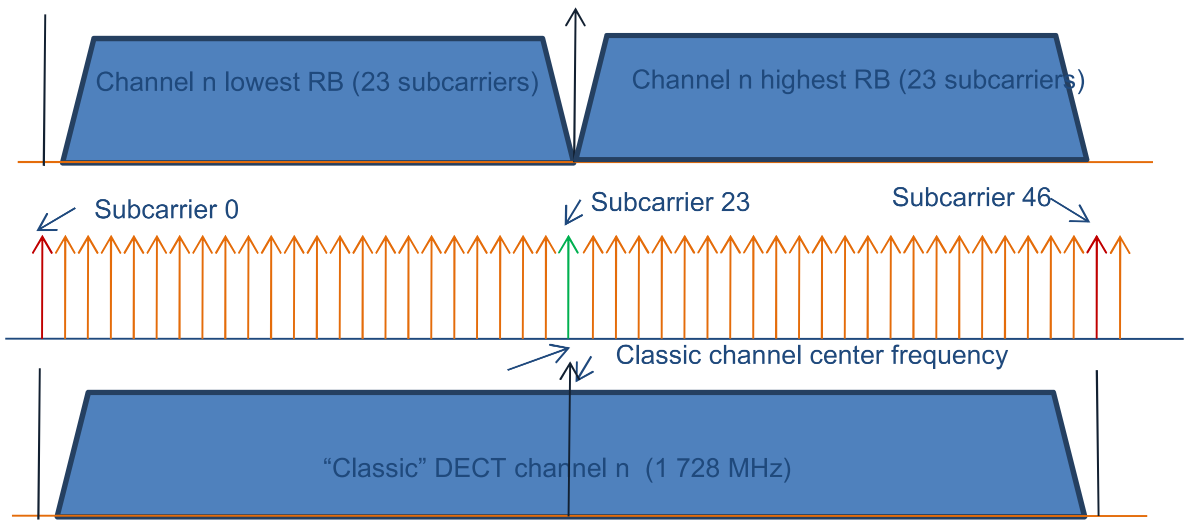

Figure 14: Lower and upper channels for each carrier

The bandwidth of each of the new sub-carriers is 862,5 kHz (1 725 / 2).

9.2.8.2 Use of half-slots

Half slots structures are defined in a similar way to DECT "classic", with some advantages due to the more efficient overhead in DECT OFDMA.

An OFDMA symbol is assumed to be used as guard space between half-slots. This is a very conservative design. There is the option of increasing the CP of the second symbol. The second (first transmitted) symbol may be used for A-field signalling (if this approach is used). To simplify numbers, only the STD option of 14 symbols per slot and the extended CP of 12 symbols for slot are considered. As the figures will show, 1 symbol of signalling and 5 and 4 symbols of user data can be transmitted in each half slot.

The signalling capacity excess in all cases the capacity of current DECT A-field (header plus tail). There is a x2/x2,5 margin that can be used for adding strong codec protection, for increasing the signalling bitrate compared to DECT, or for increasing the B-field capacity.

9.2.8.3 Figures if both improvements are implemented (half-carriers and half-slots)

9.2.8.3.1 A-field signalling capacity (half carrier options)

The capacity of this 2nd symbol assuming half-carrier is the following.

Table 19: 37,5 kHz proposal 2nd symbol capacity (half-carrier)

	Uplink		Downlink	
64 QAM	6 x 21	126 bits	6 x 22	132 bits
256 QAM	8 x 21	168 bits	8 x 22	176 bits

The calculation assumes that one additional subcarrier in each block has to be reallocated as DC carrier in the uplink (previous DC carrier moves into a guard carrier). Therefore, only 21 carriers are available for data.

Figures are large enough to allow transmission of current DECT A-field signalling (header and tail), plus slot CRCs, and there is enough clearance for coding redundancy.

These figures are identical if half-slot or full-slots are used, since one symbol is used for A-field in both cases.

9.2.8.3.2 B-field data rates (half-carriers and half-slot)

Assuming an A-field / B-field mux scheme, the remaining capacity of half-carriers and half-slots is as follows.

NOTE: The capacity used by preamble symbols (synchronization strategy not defined yet) should be deducted from the given gross rates.

Table 20: 37,5 kHz proposal, B-field gross capacity for half carrier and half slots

Standard CP option 14	First or single slot (5 symbols for B-field per half-slot)	Uplink	64 QAM	5 x 6 x 21	630	bits
			256 QAM	5 x 8 x 21	840	bits
		Downlink	64 QAM	5 x 6 x 22	660	bits
			256 QAM	5 x 8 x 22	880	bits
Extended CP option 13	Not considered for the while					
Extended CP option 12	First or single slot (4 symbols for B-field per half-slot)	Uplink	64 QAM	4 x 6 x 21	504	bits
			256 QAM	4 x 8 x 21	672	bits
		Downlink	64 QAM	4 x 6 x 22	528	bits
			256 QAM	4 x 8 x 22	704	bits

The calculation assumes that in the uplink one additional subcarrier in each block has to be reallocated as DC carrier (previous DC carrier moves into a guard carrier). Therefore only 21 carriers are available for data.

For the downlink it is believed that a single shared DC on subcarrier 23 would do the job. Therefore, no additional DC carrier is needed and 22 subcarriers may be used for data.

Figures are large enough to allow transmission of 32 kbit/s ADPCM voice channels in all cases with significant extra bits to include coding redundancy. This means that the capacity of current DECT systems is multiplied by 4.

Wideband 64 kbit/s (G.722) may be even transported in a single half-carrier / half-slot if STD preamble and 256 QAM are used. There seems to be reasonable extra space for coding redundancy (840 / 640 or 880 / 640) that may be enough unless radio conditions be poor. This means that the capacity of current DECT wideband systems is multiplied by 8.

9.2.8.4 Use of SC-FDMA in the uplink

Single Carrier FDMA (SC-FDMA), also called Linearly precoded OFDMA (LP-OFDMA), may be used to improve (lower) the peak-to-average power ratio (PAPR) of the transmitted signal.

OFDMA, as proposed in previous clauses may lead to high Peak-to-Average Power Ratios (PAPR), which forces a design with large backoffs on the power amplifiers (or other techniques, such a pre-distorters). Such design is undesirable in devices powered by batteries. SC-FDMA may be used to allow more convenient and battery efficient implementations at the expense of an additional DFT processing step preceding the OFDMA signal processing.

9.3 Other subcarrier spacings and comparative analysis

9.3.1 Overview of results on other subcarrier spacings

As previously said, the following proposals of subcarrier spacing have been investigated:

- Subcarrier spacing: 54 kHz.
- Subcarrier spacing: 37,5 kHz (as in XGP / NG PHS).
- Subcarrier spacing: 32 kHz.
- Subcarrier spacing: 15 kHz (as in LTE).

The value 54 kHz is sensible from propagation scenario point of view. It has the advantage of providing exact match with the DECT frequency structure (1 728 kHz carriers). It has slightly worse propagation characteristics compared to the 37,5 kHz approach, but still acceptable. However, it is believed that further practical improvements may be achieved by slightly reducing the subcarrier spacing to match exactly the value of 37,5 kHz, used in Japanese XGP (eXtended Global Platform), formerly known as Next-Generation PHS [i.23], what is also a multiple (× 5) of the step value used in 3GPP LTE (7,5 kHz when the reduced subcarrier value option is used).

The value 32 kHz is also sensible from propagation scenario point of view, however, if increases the complexity of the FFT required for handling the whole DECT band (from 512 to 1 024) with a minimum gain in propagation characteristics.

The value 27 kHz is also sensible from propagation scenario point of view. It has the advantage of providing exact match with the DECT frequency structure (1 728 kHz carriers). It also provides a gain in propagation characteristics compared to the 37,5 kHz approach. However, it increases the complexity of the FFT required for handling the whole DECT band (from 512 to 1 024). In addition to that, the number of symbols per slot is reduced, which reduces the efficiency due to the proportionally higher overhead caused by the preamble symbols. It also makes near impossible (by extreme inefficiency) the use of half slot.

The value 15 kHz is the value used in LTE. This value would cause an increase in the FFT complexity of x4 for the handling of the whole DECT (to 2 048). This is also true for any other 20 MHz band. or another 20 MHz band. The gain in propagation characteristics (multipath tolerance) is excessive and will never be profited for the expected cell range of the technology.

Therefore, the value 37,5 kHz is used as main approach in the study and all figures are shown using this value. At the end of the clause, clause 9.3.3.2.2 will show the differences and the comparative analysis for the option of 54 kHz. Clause 9.3.5 will show a short analysis of other identified options.

As comparison, the subcarrier spacing in other technologies are the following:

- Wi-Fi: 312,5 kHz.
- 3GPP LTE: 15 kHz (7,5 kHz as an option).
- XGP (eXtended Global Platform), formerly known as Next-Generation PHS: 10,94 kHz, 12,5 kHz, 15 kHz and 37,5 kHz in current revision (v3.5) of PHS [i.23] (being 37,5 kHz the oldest proposal in v1).
- WIMAX: variable, depending on standard variant and options (range 8 kHz to 15 kHz).

9.3.2 The 54 kHz proposal

9.3.2.1 54 kHz proposal basic parameters

The subcarrier spacing of 54 kHz has also been investigated. The basic parameters of the proposal are summarized in table 21.

Proposal for basic parameters

Table 21: 54 kHz proposal basic parameters

Parameter	DECT-evolution, 54 kHz option						
	Type 5	**Type 6**	**Type 7**	**Type 8**	**Type 9**	**Type 10**	**Type 11**
Channel band width (sampling frequency)	1,728 MHz	3,456 MHz	6,912 MHz	13,824 MHz	27,648 MHz	55,296 MHz	110,592 MHz
Sub carrier spacing	54 kHz						
FFT	32	64	128	256	512	1024	2 048
Transform period (T_{FFT})	500 / 27 µs ~ 18,52 µs						
Guard interval (cyclic prefix)	125 / 54 µs ~ 2,31 µs						
Guard interval ratio	1/8						
Symbol interval (T_{SYM})	125 / 6 µs ~ 20,83 µs						

The proposed sub carrier spacing is compatible with the existing DECT carrier allocations and suitable for the intended DECT ranges and speeds. The 'Guard interval' of 2,3 µs allows for a path difference of 700 m and provides a range of more than 1 kilometre. The sub carrier spacing is higher than in LTE (15 kHz) and XGP (37,5 kHz) and lower than in IEEE 802.11 [i.21] (312,15 kHz). The proposed basic parameters allow to reuse the existing DECT frequency and time structure.

9.3.3 Comparative analysis

9.3.3.1 Identification of key points for the analysis

The following items are identified for the analysis:

NOTE: In **bold**: most relevant items.

1) **Efficiency (GI ratio) for a given multipath protection.**
2) **Multipath protection for a given efficiency (GI ratio).**
3) Number of symbols per slot.
4) **Complexity (FFT size) for a given bandwidth.**
5) Matching with other channelizations that may exist at other frequency ranges.

The result of the analysis is presented in each clause and is summarized in a comparison table.

9.3.3.2 Comparative analysis between 37,5 kHz and 54 kHz proposals

9.3.3.2.1 Overview

This clause shows a comparative analysis of the 54 kHz and 37,5 kHz subcarrier spacing proposals. Several key points for the analysis are identified, and the pros and cons of both options are provided.

This study identifies a slightly better delay protection for a given GI ratio with the 37,5 kHz option, that also has the additional advantage of being a multiple (× 2,5) of the LTE spacing, what may be convenient for dual mode devices. The necessary FFT sizes for 10 MHz, 20 MHz and 40 MHz channels are the same in both options (256, 512 and 1 024). The 3 kHz mismatch with the DECT channel in the 37,5 kHz option is seen as not relevant.

And result of the analysis, this study considers that the 37,5 kHz option has advantages over the 54 kHz option.

9.3.3.2.2 Analysis

9.3.3.2.2.1 Efficiency (GI ratio) for a given multipath protection

The necessary GI ratio for a given multipath protection (GI value) is higher for higher subcarrier spacings, as shown in next table. Therefore, the 37,5 kHz approach requires always less overhead than the 54 kHz value. This happens for all guard interval cases.

Table 22: GI ratio vs. Multipath protection for 37,5 kHz and 54 kHz proposals

Multipath protection		**Option 37,5 kHz**		**Option 54 kHz**	
		TFFT (μs) = 26,67		TFFT (μs) = 18,51	
Time Guard Interval (GI) in ns	***equivalent distance (m)***	***GI ratio (1/x)***	***% overhead***	***GI ratio (1/x)***	***% overhead***
1 000	300	26,67	3,75 %	18,518 518 5	5,40 %
1 500	450	17,78	5,62 %	12,345 679	8,10 %
2 000	600	13,335	7,50 %	9,259 259 26	10,80 %
2 500	750	10,668	9,37 %	7,407 407 41	13,50 %
3 000	900	8,89	11,25 %	6,172 839 51	16,20 %
3 500	1 050	7,62	13,12 %	5,291 005 29	18,90 %
4 000	1 200	6,6675	15,00 %	4,629 629 63	21,60 %
5 000	1 500	5,334	18,75 %	3,703 703 7	27,00 %
6 000	1 800	4,445	22,50 %	3,086 419 75	32,40 %
8 000	2 400	3,33375	30,00 %	2,314 814 81	43,20 %
10 000	3 000	2,667	37,50 %	1,851 851 85	54,00 %

For instance, for a multipath protection of 3 000 ns equivalent to 900 m, the 37,5 kHz solution would require a GI ratio of 1 / 8,89, adding an overhead of 11,25 %. The 54 kHz solution would require GI ratio = 1 / 6,17 adding a 16,20 % overhead.

Conclusion: the 37,5 kHz option would add less GI overhead than the 54 kHz option for all cases.

9.3.3.2.2.2 Multipath protection for a given efficiency (GI ratio)

The multipath protection provided for a given percentage of overhead, increases when reducing the subcarrier spacing. Therefore, it is always better for the 37,5 kHz approach than for the 54 kHz option as shown in table 23.

Table 23: Multipath protection vs. GI ratio for 37,5 kHz and 54 kHz proposals

% GI Overhead		**Multipath protection**			
		Option 37,5 kHz		**Option 54 kHz**	
% overhead	*GI ratio (1/x)*	*time Guard Interval (GI) in ns*	*equivalent distance (m)*	*time Guard Interval (GI) in ns*	*equivalent distance (m)*
3,00 %	33,33	800	240	556	167
4,00 %	25,00	1 067	320	741	222

% GI Overhead		Multipath protection			
		Option 37,5 kHz		Option 54 kHz	
5,00 %	20,00	1 334	400	926	278
6,00 %	16,67	1 600	480	1 111	333
7,00 %	14,29	1 867	560	1 296	389
8,00 %	12,50	2 134	640	1 481	444
9,00 %	11,11	2 400	720	1 667	500
10,00 %	10,00	2 667	800	1 852	556
11,00 %	9,09	2 934	880	2 037	611
12,00 %	8,33	3 200	960	2 222	667
13,00 %	7,69	3 467	1 040	2 407	722
14,00 %	7,14	3 734	1 120	2 593	778
15,00 %	6,67	4 001	1 200	2 778	833
17,00 %	5,88	4 534	1 360	3 148	944
20,00 %	5,00	5 334	1 600	3 704	1 111
25,00 %	4,00	6 668	2 000	4 630	1 389
30,00 %	3,33	8 001	2 400	5 556	1 667

For instance, for a 12 % overhead, the 37,5 kHz solution would be able to compensate for around 3 200 ns, equivalent to 960 meters of multipath, while the 54 kHz solution would be only able to support 2 222 ns, equivalent to 667 meters.

Conclusion: the comparison is always favourable for the 37,5 kHz option.

9.3.3.2.2.3 Number of symbols per slot

The number of symbols per DECT slot in the 37,5 kHz option is 14, assuming a GI ratio of 1 / 8,6, as shown in clause 9.3. This value includes the slot guard space, estimated in 1 symbol , that would be lost in any single bearer transmission. Thus, 13 symbols per DECT full slot may be transmitted.

In the 54 kHz option, the number of symbols per slot with a GI ratio of 1 / 8, would be 20, including the slot guard interval. The total time per symbol is 20,8 µs.

With this symbol time, the time of one symbol is probably not enough for implementing the slot guard time, at least comparing it with current guard time in DECT frame. On the other hand, two symbols (41,6 µs) should be more than enough.

As comparison, the number of symbols per slot in other technologies is as follows:

XGP / NG-PHS

In XGP (initially known as NG-PHS), the frame time is 5 ms and the slot time is 625 µs . There are 19 symbols per slot of plus a 51,67 µs of additional guard space per slot. The first symbol has also an extended GI of ratio ¼, while all others have 1 / 8.

3GPP LTE (FDD variant)

In LTE FDD the frame duration is 10 ms and there are 20 slots of 500 µs. Slots carry 6 or 7 symbols, depending on if the normal GI ratio or the extended GI ratio are used.

The normal GI in LTE is equal to 4,61 µs, except for the first symbol, that is 5,21 µs. The optional extended GI is equal to 16,67 µs. In bad radio conditions, there is the option to use a subcarrier spacing of 7,5 kHz, increasing the GI to 33,33 µs.

Conclusion: taking into account that LTE carries 6/7 symbols per slot and XGP / NG-PHS 19 symbols, the number of symbols per slot provided by both DECT-Advanced proposals (14 or 20) are reasonable, and the possible pros or cons depend on MAC architecture details. At this stage of the work, there is not possible to identify any a-priory advantage or disadvantage by using 14 or 20 symbols per slot.

9.3.3.2.2.4 Complexity (FFT size) for a given bandwidth

In theory, the FFT complexity would be the main identifiable advantage of an increased subcarrier spacing. For a given bandwidth, the FFT complexity would be lower for larger subcarrier spacing than for smaller spacing. However, for this analysis, it is necessary to take into account the real channel bandwidth values and the fact that FFT provides a maximum number of possible subcarriers as a power of 2 (2^n). Table 24 shows the FFT sizes and the usable channel bandwidths for the 37,5 kHz option, taking into account DECT channelization.

Table 24: FFT complexity for 37,5 kHz proposal

Parameter	DECT-evolution, 37 kHz option					
	Type 1	Type 2	Type 3	Type 4 (base DECT band)	Type 5	Type 6
Sampling frequency	2,400 MHz	4,800 MHz	9,600 MHz	19,200 MHz	38,400 MHz	76,800 MHz
Usable channel bandwidth with DECT channels	1,728 MHz (1 DECT carrier)	3,456 MHz (2 DECT carriers)	8,640 MHz (5 DECT carriers)	17,280 MHz (10 DECT carriers)	38,016 MHz (22 DECT carriers)	76,032 MHz (44 DECT carriers)
Subcarrier spacing	37,5 kHz					
FFT	64	128	256	512	1 024	2 048
Transform period (T_{FFT})	$(37,5\ \text{kHz})^{-1}$ = 26,67 µs					

Perhaps, the most important value in table 24 is the FFT size necessary for handling a frequency bandwidth of 20 MHz, equivalent to the DECT band. As shown in the table, it is possible to handle the whole DECT band (20 MHz allocated/17,280 MHz useful) with an FFT of only 512. This value is not especially large, taking into account the proposals for LTE. The 512 FFT would allow to include additional subcarriers until a hypothetical 19,20 MHz usable bandwidth, what seems to be more than enough for a 20 MHz channel.

Furthermore, a hypothetical 40 MHz channel, composed of current DECT band plus the IMT-2000 extension shared with UTRAN TDD (1 900 MHz to 1 920 MHz), may be handled with only 1 024 FFT, assuming current channel positions, see table A.1. It should be noted that two additional channels may be included in the 1 900 MHz area, providing a continuous coverage over up to 38,40 MHz usable bandwidth.

The interest of the 1024 case is that this is the needed value for providing the 1 Gbit/s data rate required by IMT-Advanced specifications (in combination with MIMO 6 x 6.

As comparison, LTE would require an FFT of 4 096 for providing the 1 Gbit/s operating in the same conditions.

Other FFT sizes and channel bandwidths are less relevant, due to the relatively low value needed for handling the whole bandwidth. A half-DECT-band transceiver (able to operate over 5 DECT channels = 10 MHz) would require only an FFT size of 256 while 2 DECT carriers would require 128 and one DECT carrier 64.

9.3.3.2.2.5 Alignment with 3GPP LTE

3GPP LTE used a fixed subcarrier spacing of 15 kHz, with option to 7,5 kHz in special radio conditions. The 37,5 kHz proposal for DECT advanced is a multiple of x2,5 compared to LTE standard spacing and x5 compared to LTE reduced spacing.

The advantage of the alignment with LTE is connected with the possible implementation of dual mode radios. In order to get 100 % alignment, the DECT subcarrier positions may be synchronized with the LTE grid (one of each five LTE subcarriers would have identical frequency to a DECT-Advanced subcarrier). After this alignment is done, it is feasible the design of a dual mode radio by means of signal processing. This may be necessary in bands where DECT-Advanced and LTE may potentially share the spectrum.

9.3.3.2.2.6 Alignment with XGP / NG-PHS

XGP (eXtended Global Platform), formerly known as Next-Generation PHS [i.23] uses a subcarrier spacing of 37,5 kHz (within other options). The 37,5 kHz proposal for DECT-Advanced is therefore fully aligned with XGP.

The interest of the alignment with XGP is multiple:

1) it is convenient for the semiconductor vendors since it makes easier the design of dual mode (XGP / DECT) chipsets;

2) it makes possible the implementation of dual mode systems able to operate in XGP or DECT modes;

3) an additional advantage is the possibility to reuse the experimental experience obtained with the same subcarrier spacing in XGP / NG-PHS, especially in radio channel modelling.

If a complete alignment is required, for instance if the design of dual mode systems able to operate in both modes *simultaneously*, e.g. for the market in Japan, it would be possible to fully align the position of the subcarrier frequencies.

9.3.3.2.2.7 Matching with the DECT channel of 1 728 over the DECT 1 880 MHz to 1 900 MHz band

The 37,5 kHz approach has the drawback of a mismatch of 3 kHz with the current DECT channel (the DECT channel is 1 728 kHz and the DECT-Advanced channel would be 1 725 kHz). The mismatch is 3 kHz per channel. Assuming a perfect match in the middle of the DECT band (channels 4 and 5), the accumulated mismatch in the worst case (at the edges of channels 0 and 9) would be of 15 kHz.

However, taking into account the roll-off factor of current DECT technology (0,5) and the tolerances in centre frequency oscillator frequencies, the practical effect of this 15 kHz mismatch is fully irrelevant. The DECT gross bit rate is 1 152 kbit/s and the Shannon bandwidth is 1 152 kHz. Taking into account that the DECT channel is 1 728 kHz, the excess over Shannon bandwidth is 576 kHz. The tolerance allowed in DECT centre frequencies is 50 kHz (see ETSI EN 300 175-2 [i.2], clause 4.1.2). Taking it into account, the possible shift of the adjacent channel by 12 kHz (worst case possible in the edge between DECT channels 0 and 1 or 8 and 9) is irrelevant in both ways (DECT classic transmitters received by a DECT-Advanced receiver and vice versa).

Conclusion: the 3 kHz mismatch between DECT-Advanced channel, option 37,5 kHz (1 725 kHz) and DECT-Classic channel (1 728 kHz) is irrelevant compared to DECT tolerances.

The 54 kHz approach provides a perfect matching with DECT 1 728 channel.

9.3.3.2.2.8 Matching with other channelizations that may happen at other frequency ranges

When defining DECT-Advanced, it should be taken into account that the technology would (hopefully) have additional frequency allocations out of the DECT band. In many of these allocations, DECT-Advanced may be required to either:

1) share spectrum with other technologies; and/or

2) use channel sizes already defined and different of DECT channels. In such situation, the use of a 1 728 channel bandwidth for DECT-Advanced may not be the optimal solution. In general, it is more convenient to match the channel size and edge positions with the other technologies in operation in the same frequency range, or with a multiple or sub-multiple.

This means, that the definition of the DECT-Advanced channel bandwidth (frequency resource block and subcarriers positions) may be different in bands other than the DECT band. In such scenario, the perfect matching with 1 728 kHz channels is irrelevant. On the other hand, the matching with other systems may be convenient. If the other technology is OFDMA with subcarrier spacing of 15 kHz (such as LTE), the 37,5 kHz approach provides some advantage due to the x2,5 ratio.

9.3.3.2.2.9 Alignment with DECT "classic"

During the review it was pointed that there may be some advantage for the 54 kHz option due to the exact ratio $54 \times 34 = 1\ 728$.

Once it is accepted that the 3 kHz per channel mismatch of the 37,5 kHz option is irrelevant due to subcarrier spacing and tolerances, there is no practical difference.

The exact value 32 is not relevant since radios will typically handle several channels, and the FFT for handling the band is the relevant factor. This is already covered in table 24.

Conclusion: no practical difference.

9.3.3.2.3 Summary of the analysis

Table 25 summarizes the result of the analysis.

Table 25: Summary for 37,5 kHz and 54 kHz proposals

Parameter	37,5 kHz	54 kHz	remarks
Efficiency (GI ratio) for a given multipath protection			The 37,5 kHz approach is always the winner
Multipath protection for a given efficiency (GI ratio)			The 37,5 kHz approach is always the winner
Number of symbols per slot	=	=	Both values are OK. No particular advantage identified at this stage of the design
Complexity (FFT size) for a given bandwidth	=	=	Same FFT sizes for 10 MHz, 20 MHz and 40 MHz channel (256, 512 and 1 024)
Alignment with 3GPP LTE			Better with 37,5 kHz approach. Relative importance of this factor is unclear at this stage
Alignment with XGP [i.23]			Better with 37,5 kHz approach. Relative importance of this factor is unclear at this stage
Matching with the DECT channel of 1 728 over the DECT 1 880 MHz to 1 900 MHz band	=	=	The 3 kHz mismatch with the 37,5 kHz approach is irrelevant taking into account DECT tolerances
Matching with other channelizations that may happen at other frequency ranges	= (?)	=	Possible advantages of the 37,5 kHz approach if the other technology is LTE or XGP. Otherwise, no differences
Alignment with DECT "classic"	=	=	No practical difference

9.3.3.2.4 Conclusion

Taking into account the result of the analysis, the 37,5 kHz approach has advantages over the 54 kHz option and is recommended as primary option.

A major merit of the 37,5 kHz approach is that the entire DECT band (10 channels) can be handled with an FFT of 512.

Furthermore, the DECT band plus the extension band (including the 2 additional carriers gained around 1 900 MHz) may be handled with an FFT of 1 024.

9.3.4 The 27 kHz proposal

The DECT TDMA/FDMA defines a channel spacing of 1 728 kHz and (416 + 2/3) µs time slots.

This contribution focus on initially defining an OFDM FDMA/TDMA system able to employ multiple DECT channels (bonded DECT channels) in each time slot, as well as employing continuous transmission over multiple time slots to be more efficient.

OFDM FDMA/TDMA would allow to keep the general behaviour of DECT, which would not be possible in a OFDMA setup. Use of OFDM enables the use of MIMO (spatial multiplexing, Space Time Block Coding) in a straight forward manner. Nevertheless, MIMO is not included here.

A **general frame structure** within a time slot might be:

- [P][P][H][D]…[0]

where [.] is a OFDM symbol including Cylic Prefix(CP).

- [P]: Preamble symbol (repeated twice to allow for Schmidl & Cox synchronization in time and frequency.
- [H]: Header symbol (m=2, r=1/2).
- [D]: Data symbol.
- [0]: Zero Guard.

Concerning Preamble:

The Preamble is usually based on Zadoff-Chu sequences, which are repeated twice. This allows for Schmidl & Cox synchronization in time (frame detect), frequency (Carrier Frequency Offset) and channel estimation (LS, MMSE).

Concerning header symbol:

If only one DECT channel is employed, the header carries e.g. 54 bits (see table 26). While the legacy DECT has 64 bits in the A-field. It might therefore be required to have [P][P][H][H][D]…[0] frame structure, if only a single DECT channel is used.

Data Symbol:

It is assumed that the frame check sequence is part of the last data symbol [D] transmitted. Each data symbol carries pilots for phase and channel tracking.

Table 26: Basic parameters for the 27 kHz proposal

Name	Symbol	Value	Comment
Number of bonded DECT channels	N_{ch}	{1, 2, 4, 8, 10*}	*to be realized as N_{FFT}=64, if mandated
Bandwidth	B	N_{ch} x 1 728 MHz	1 728 MHz to 17,28 MHz
Subcarrier Spacing	Δf	27 kHz	1 728 kHz = 3 x 3 x 3 x 2^6
FFT Size	N_{FFT}	64 x N_{ch}	
Sampling Clock	N_{FFT} x 27 kHz	N_{ch} x 1 728 kHz	DECT Channel B=1 728 kHz DECT Quartz: 20,763 MHz
Useful Symbol Duration	T_{FFT}= 1/Δf	37,037 μs	
Cylic Prefix	T_{CP}=T_{FFT}/8	4,629 μs	tentatively
Total Symbol Duration	T_{Sym}= T_{FFT} + T_{CP}	41,666 μs	
Number of Slots	N_{slot}	24	DECT standard value
Slot Duration	T_{slot}= 10 ms / N_{slot}	416,666 μs	DECT standard value
OFDM Symbols / Slot	N_{sym}	10	concluded
Number of Data Carriers	N_{data}	54 x N_{ch}	tentatively
Number of Pilot Carriers	N_{pilot}	4 x N_{ch}	tentatively
Number of Zeroed Carriers	N_{zero}	1 (DC) + (3 (low) + 2 (up)) x N_{ch}	tentatively
Modulation Order	m	{2, 3, 4, 6, 8}	tentatively
Code Rate	r	{1/2, 2/3, 3/4, 7/8}	tentatively
Max. Data Rate	(N_{data}x r x m / T_{sym}) x N_{ch}	(1 296 Mbit/s to 9,072 Mbit/s) x N_{ch}	tentatively

Comments:

- **Number of bonded channels:** It should be noted, that a restriction in the number of bonded channels and time slots is required to ensure a proper coexistence of multiple devices.

- **Number of bonded channels:** Ten(10) bonded channels results in an inefficient implementation, e.g. five times FFT-128. Nevertheless, it is questionable if bonding of more than eight channels should be enabled by the standard.
- **Subcarrier Spacing:** One could consider 13,5 kHz and 54 kHz as well. It is a design choice depending on technology and assumed frequency selectivity of RF channel.
- **Sampling Clock:** The sampling clock is here derived from typical DECT oscillators.
- **Slot Duration / Number of OFDM symbols:** It might be useful to reduce the slot duration by a factor of 2 in general, resulting in 5 OFDM symbols, e.g. [P][P][H][D][0], per time slot, but only if more than one DECT channel are bonded to have enough space in the header. The system should be allowed to employ multiple time slots under some restrictions for the frame start and end position. Frame structure is then [D] [D]…[D]. The Temporal Guard [0] should be given in the last time slot.
- **Cyclic Prefix:** A cyclic prefix of 1/8 of the useful symbol duration is selected to get a symbol length of 1/10 of a DECT time slot. The CP is highly dependent on the multipath dispersion of the RF channel and channel measurements of different scenarios are required for a proper selection.
- **Number of Data/Pilot/Zeroed Carriers:** These values are provided without being investigated in detail. The zeroed Carriers at the lower and upper edge of the occupied bandwidth need to scale with the number of bonded DECT channels.
- **Modulation order m, code rate r:** Modulation and Coding Schemes (MCS) are given by modulation order m and code rate r.

NOTE: The data modulation is given in terms of modulation order m. Typical values are shown in table 27.

Table 27: Modulation order

Modulation	Modulation order
4-QAM/QPSK	2
8-PSK	3
16-QAM	4
64-QAM	6
256-QAM	8

256-QAM modulation and code rate 7/8 might be only valid for communication in close proximity. There exist (m,r)-combinations which yield to the same data rate, so these combinations need to be selected based on BER performance and implementation complexity.

It is important to make a certain set of MCS mandatory, so they are implemented.

The parameters above are all only tentatively with some consideration on practice. No simulations have been carried out to check, if they are really viable.

9.3.5 Other identified options

This study has considered other values of the subcarrier spacing. The following text provides a summary of the different options considered and the comparative analysis compared to the 37,5 kHz central option. As general conclusion, variant with lower sub-carrier space than 37,5 kHz for at least one step (x2) of additional FFT complexity with some debateable (over-killing) gain in the propagation characteristics. On the other hand, options with larger sub-carrier space reduce the multipath protection and/or efficiency. The minimum value that provides an advantage in terms of FFT size is 75 kHz.

15 kHz: This is the LTE subcarrier spacing. The problem with the 15 kHz value is that it was designed and optimized for public cellular scenario. It provides excessive delay spread protection (1 000 ns / 3 000 meters for GI ratio of 1/8) and, in turn, it multiplies by 4 the complexity of the FFT compared to the 37,5 kHz approach (4 096 FFT size to achieve the 1 Gbit/s target). Thus, this choice is very difficult to justify for technology intended for residential and business applications. Obviously, the alignment with LTE is the optimal.

27 kHz:	Slightly better propagation characteristics than the 37,5 kHz option. Provides perfect matching with DECT 1,728 MHz channelization. It increases the complexity of the FFT required for handling the whole DECT band (from 512 to 1 024). In addition to that, the number of symbols per slot is reduced, which reduces the efficiency due to the proportionally higher overhead caused by the preamble symbols. It also makes near impossible (by extreme inefficiency) the use of half slot. Analysed in more detail in clause 9.3.4.
30 kHz:	This option provides optimal convergence with LTE (x2); however, there will be a step x2 in FFT size compared to the 37,5 kHz approach (20 MHz bandwidth cannot be handled with a 512 FFT anymore) in turn of a minor improvement in propagation characteristics.
32 kHz:	This option provides perfect match with DECT channelization (54 sub-carriers per channel); however, there will be a step x2 in FFT size compared to the 37,5 kHz approach (20 MHz bandwidth cannot be handled with a 512 FFT anymore) in turn of a minor improvement in propagation characteristics.
45 kHz:	Provides good convergence with LTE (x3) and same FFT size as the 37,5 kHz option. Slightly worse propagation characteristics and/or efficiency.
54 kHz:	Slightly worse propagation characteristics and/or efficiency than the 37,5 kHz option. Same FFT size as the 37,5 kHz option. Provides perfect matching with DECT 1,728 MHz channelization. Analysed in more detail in clause 9.3.2.
75 kHz:	75 kHz is the minimum value that allows a reduction in the FFT size (256 FFT for 20 MHz). Worse propagation characteristics and/or efficiency. Good alignment with LTE (subcarrier space × 5).
108 kHz:	The problem with this approach is that it provides a multipath protection too close to Wi-Fi. (Wi-Fi has 312,5 kHz subcarrier spacing, but operates with GI = 1/4 = 800 ns). With this value, the range of DECT-OFDMA cells would be very close to Wi-Fi cells, sharing most of its indoor coverage limitations. The technology positioning is less clear. On the other hand, FFT size would be as with the 75 kHz option.

9.3.6 For further study

The next step would be building a channel simulation model in order to evaluate propagation characteristics of the selected subcarrier spacing over the candidate frequency ranges. MIMO characteristics should be also be investigated from the beginning.

10 Implementation of the IETF RFC 8105 (IPv6 over DECT ULE)

10.1 Introduction

This clause provides an overview of the new IETF RFC 8105 [i.13]: "Transmission of IPv6 Packets over Digital Enhanced Cordless Telecommunications (DECT) Ultra Low Energy (ULE)", and describes a series of recommendations related to DECT standardization in order to properly support devices and architectures based on IPv6 and IETF RFC 8105 [i.13].

10.2 Overview of IETF RFC 8105

10.2.1 Introduction

Digital Enhanced Cordless Telecommunications (DECT); Ultra Low Energy (ULE) is a low-power air interface technology that is defined and specified by ETSI TC DECT by the multipart deliverable ETSI TS 102 939 [i.19] and [i.20]. Currently there are two phases specified by the two parts ETSI TS 102 939-1 [i.19] and ETSI TS 102 939-2 [i.20].

DECT ULE is a recent addition to the DECT interface primarily intended for low-bandwidth, low-power applications such as sensor devices, smart meters, home automation, etc. As the DECT ULE interface inherits many of the capabilities from DECT, it benefits from operation over reserved frequency band with longer-range and less interference than other technologies.

The use of IPv6 in combination with header compression techniques is seen as ideal for a wide range of Internet of Things applications.

IETF RFC 8105 [i.13] describes how IPv6 is transported over DECT ULE using IPv6 over Low-Power Wireless Personal Area Network (6LoWPAN) techniques.

DECT ULE technology sets strict requirements for low power consumption and, thus, limits the allowed protocol overhead. 6LoWPAN standards IETF RFC 4944 [i.29], IETF RFC 6775 [i.31] and IETF RFC 6282 [i.30] provide useful functionality for reducing overhead that can be applied to DECT ULE.

10.2.2 IETF RFC 8105 protocol stack model

IETF RFC 8105 [i.13] protocol stack model is shown in figure 15.

The general model is that IPv6 is Layer 3 and DECT ULE MAC and DECT ULE DLC are Layer 2 for the RFC model. DECT ULE already implements (see ETSI TS 102 939-1 [i.19]) fragmentation and reassembly functionality; hence, the fragmentation and reassembly function described in IETF RFC 4944 [i.29] is not used.

```
+-------------------+
|    UDP/TCP/other  |
+-------------------+
|       IPv6        |
+-------------------+
|6LoWPAN adapted to |
|    DECT ULE       |
+-------------------+
|  DECT ULE DLC     |
+-------------------+
|  DECT ULE MAC     |
+-------------------+
|  DECT ULE PHY     |
+-------------------+
```

Figure 15: IETF RFC 8105 [i.13] protocol stack model

The layer "6LoWPAN adapted to DECT ULE" is responsible for the IPv6 header compression and is described in detail in IETF RFC 8105 [i.13], that makes reference to other RFCs describing the 6LoWPAN header compression technique, fundamentally to the IETF RFC 6282 [i.30].

10.2.3 Data transmission setup

In order to enable data transmission over DECT ULE, a Permanent Virtual Circuit (PVC) has to be configured and opened between the FP and PP. This is done by setting up a DECT service call between the PP and FP. In the DECT protocol domain, the PP specifies the <<IWU-ATTRIBUTES>> in a service-change (other) message before sending a service-change (resume) message as defined in ETSI TS 102 939-1 [i.19]. The <<IWU-ATTRIBUTES>> sets the ULE Application Protocol Identifier to 0x06 and the MTU size to 1 280 octets or larger. The FP sends a service-change-accept (resume) that contains a valid paging descriptor. The PP listens to paging messages from the FP according to the information in the received paging descriptor. Following this, transmission of IPv6 packets can start.

After the FPs and PPs have connected at the DECT ULE level, the link can be considered up and IPv6 address configuration and transmission can begin. The 6LBR (see IETF RFC 8105 [i.13]) ensures address collisions do not occur.

As discussed in IETF RFC 4903 [i.34], conventional usage of IPv6 anticipates IPv6 subnets spanning a single link at the link layer. In order to avoid the complexity of implementing a separate subnet for each DECT ULE link, a Multi-Link Subnet model IETF RFC 4903 [i.34] has been chosen, specifically Non-Broadcast Multi-Access (NBMA) at Layer 2.

Because of this, link-local multicast communications can happen only within a single DECT ULE connection; thus, 6LN-to-6LN communications using link-local addresses are not possible. 6LNs connected to the same 6LBR have to communicate with each other utilizing the shared prefix used on the subnet. The 6LBR forwards packets sent by one 6LN to another.

The IPv6/6LoWPAN as described in IETF RFC 8105 [i.13] document is considered to be an application-layer protocol on top of DECT ULE. In order to provide interoperability between 6LoWPAN / DECT ULE devices, a common application protocol identifier for 6LoWPAN is defined in ETSI TS 102 939-1 [i.19].This is the 0x06 indicating IPv6 / 6LoWPAN.

10.2.4 IETF RFC 8105 addressing model

Each DECT PP is assigned an IPEI during manufacturing. See ETSI EN 300 175-6 [i.6]. This identity has the size of 40 bits and is globally unique within DECT addressing space and can be used to constitute the MAC address used to derive the Interface Identifier (IID, see IETF RFC 8065 [i.35]), for link-local address.

Each DECT FP is assigned an RFPI during manufacturing or installation. This identity has the size of 40 bits and is globally unique within DECT addressing space and can be used to constitute the MAC address used to derive the IID for link-local address.

Optionally, each DECT PP and DECT FP can be assigned a unique (IEEE) MAC-48 address in addition to the DECT identities to be used by the 6LoWPAN. During the address registration of non-link-local addresses as specified by IETF RFC 8105 [i.14], the FP and PP can use such MAC-48 to construct the IID. However, as these addresses are considered as being permanent, such a scheme is not recommended as per IETF RFC 8065 [i.35].

10.2.5 Stateless Address Auto-configuration

At network interface initialization, both 6LN and 6LBR generate and assign IPv6 link-local addresses to the DECT ULE network interfaces (see IETF RFC 4862 [i.28]) based on the DECT device addresses (see clause 10.2.4) that were used for establishing the underlying DECT ULE connection.

The DECT device addresses IPEI and RFPI are used to derive the IPv6 link-local 64-bit Interface Identifiers (IIDs) for 6LN and 6LBR respectively.

The rule for deriving IIDs from DECT device addresses is as follows:

The DECT device addresses that consist of 40 bits each is expanded with leading zero bits to form 48-bit intermediate addresses.

The most significant bit in this newly formed 48-bit intermediate address is set to one for addresses derived from the RFPI and set to zero for addresses derived from the IPEI.

64-bit IIDs are derived from these intermediate 48-bit addresses following the guidance in Appendix A of IETF RFC 4291 [i.26]. However, because DECT and IEEE address spaces are different, this intermediate address cannot be considered to be unique within an IEEE address space. In the derived IIDs, the Universal/Local (U/L) bit (7th bit) will be zero, which indicates that derived IIDs are not globally unique, see IETF RFC 7136 [i.32].

Global uniqueness of an IID in link-local addresses is not required as they should never be leaked outside the subnet domain.

As defined in IETF RFC 4291 [i.26], the IPv6 link-local address is formed by appending the IID to the prefix FE80::/64, as shown in figure 16.

```
   10 bits       54 bits           64 bits
+----------+-----------------+----------------------+
|1111111010|      zeros      | Interface Identifier |
+----------+-----------------+----------------------+
```

Figure 16: IPv6 Link-Local Address in DECT ULE

After link-local address configuration, 6LN sends Router Solicitation messages as described in clause 6.3.7 of IETF RFC 4861 [i.27] and clause 5.3 of IETF RFC 6775 [i.31]. For non-link-local addresses, 6LNs should not be configured to use IIDs derived from a MAC-48 device address or DECT device addresses.

10.2.6 Header Compression

As defined in IETF RFC 6282 [i.30], which specifies the compression format for IPv6 datagrams on top of IEEE 802.15.4 [i.38], header compression is required in IETF RFC 8105 [i.13] as the basis for IPv6 header compression on top of DECT ULE. All headers are compressed according to encoding formats as described in IETF RFC 6282 [i.30]. The DECT ULE's star topology structure, ARO and 6CO, can be exploited in order to provide a mechanism for address compression. The IETF RFC 8105 [i.13] describes the principles of IPv6 address compression on top of DECT ULE.

10.2.7 Security Considerations

The secure transmission of circuit mode services in DECT is based on the DSAA2 and DSC/DSC2 specifications described in ETSI EN 300 175-7 [i.7].

DECT ULE communications are secured at the link-layer (DLC) by encryption and per-message authentication through CCM (Counter with Cipher Block Chaining Message Authentication Code (CBC-MAC)) mode, as described in ETSI EN 300 175-7 [i.7] and ETSI TS 102 939-1 [i.19]. This mode is similar to and is based on IETF RFC 3610 [i.33]. The underlying algorithm for providing encryption and authentication is AES128.

10.3 Impact and recommendations on DECT standardization

10.3.1 Regarding addressing

IETF RFC 8105 [i.13] provides at least a basic mode of generating the requiring IID from DECT addressing. The address generation scheme in the RFC only allows one link-local address per IPEI (or RFPI). The RFC also allows generation of a single global routable IPv6 address derived from the DECT addresses (IPEI or RFPI). However, for privacy reasons the RFC recommends that global routable IPv6 addresses are generated by other schemes such as CGA, HBA or DHCPv6 [i.36]. These schemes are not limited to single addresses. Normally, applications are expected to use routable addresses rather than link-local address. It may also be expected having multiple applications in single ULE devices, they will use same IP-address, but different IP-port.

10.3.2 Regarding energy handling and initial configuration matters

The RFC does not handle the specific topics of ULE technology regarding energy administration. The most obvious of these topics is the initial allocation of paging cycles and paging descriptor. The correct allocation of cycles and descriptors is seen as a layer 2 matter that should be solved by the underlying technology (DECT ULE in this case). The RFC does not provide solutions and does not handle at all the matter.

In current implementations of ULE an ad-hoc solution was created based on using a configuration application protocol developed by the ULE-Alliance. Such protocol cannot be assumed to exist in an IPv6 IETF RFC 8105 [i.13] scenario, since the whole thing of the IETF RFC 8105 [i.13] approach is providing an industry standard (IPv6) abstraction layer decoupling the ULE mechanics from any application protocol. Such application protocol may be developed by any vendor or organization, and requires transparency regarding the transport technology.

Therefore, this study identifies as gap in DECT standardization the implementation in DECT standards of all operations to be performed during initial or subsequent configuration, previous to the allocation of paging descriptors by means of the IE <<MAC-CONFIGURATION>>.

Several technical approaches are in theory possible, but taken into account that the allocation of descriptors is done by DECT NWK layer mechanisms (IE <<MAC-CONFIGURATION>>), it seems straightforward the use of similar DECT NWK layer mechanisms. Several MM operations and IEs are suitable for the negotiation. The IE <<SETUP-CAPABILITY>> is a candidate for two-way negotiations requiring parameters and responses sent from both sides (PP and FP): The IE <<TERMINAL CAPABILITY>>, already included at registration, may also be used for one-way declaration of parameters (PP only).

Nevertheless, and irrespective of the used transport method, it seems that the semantics of this initial declaration and negotiation is far from being simple. And probably the semantic of existing developed specific application protocols are not enough to cover the wide range of use cases that can exist in a generic use of the ULE technology as universal transport of any application protocol. Therefore a complete analysis and study is needed.

A general scenario should assume that the application protocols are not known at DECT standard development. Also, several applications may be running at the same DECT cell at the same time. It should also consider the case of application protocols (or at least semantics) that may happen in many products.

The most feasible solution foreseen by this study would consist on defining and standardized mechanism of "presentation" of any DECT ULE device with a set or standardized parameters that may be used by any developer of products based on IPv6/ULE. Such parameters should include information about the requirements on power administration and paging cycles. An algorithm at FP side (that may be standardized or just based on the standard) would try to get a compromise on the requirements of the several connected devices to minimize energy consumption and at the same time, providing the requiring paging needs.

A generic algorithm handling the ULE devices as independent entities is probably difficult to implement. However, the problem may also be handled assuming that in most real cases the PP devices are grouped by "sets", each of them implementing a given application or, in case of proprietary products, made by the same vendor. In each set there may be different types of devices (i.e. actuator, sensor type A, sensor type B, etc.). It may also be assumed that multiple devices of the same type (i.e. several switches) are identical regarding configuration needs, which simplifies the problem.

A promising approach would be first grouping the devices by "sets", inside each "set" by type (i.e. actuator, sensor) and then allowing a "master" in each "set" to participate in the exchange or parameters and negotiation with the FP. This approach would probably map better the expected commercial scenarios.

In any case, the topic is not trivial and requires a dedicated analysis and solution. Previous approaches to handle the matter may be investigated but are not enough to solve the problem in a general way. A scheme of the intended commercial architecture of DECT ULE products would also be a part of the analysis. Finally, it is believed that more interaction with a wide community of potential application developers (via the proper standardization bodies or other initiatives) is a requirement for an effective completion of the ULE technology in order to be part of the global IoT arena.

11 Further topics for DECT evolution

11.1 Introduction

A general prerequisite for the proposed topics is that the evolution of DECT should not compromise the current QoS of DECT and should not affect the operation of legacy DECT products in the same spectrum. The proposed evolution topics should preferably be application in both the current core band as well in the potential extension bands.

11.2 Redundant data transmission

Some applications are demanding much lower packet error rate as what is usual in DECT. Some examples are high quality audio streaming from microphones for live stage performance or time critical control applications that cannot accept additional latency from retransmission. Packet errors may happen as result of packet collision, fading or at the range limit.

For lowering the packet error rate, several methods can be considered. The methods are introducing redundant transmissions.

Possible methods to be considered:

a) Dual-slot diversity, a simple but effective scheme for transmission of same information twice on different timeslot and/or frequency. The receiver will then have two opportunities to receive the data, and discards any redundant reception. If different slot and frequencies are used, the method is effective against both interference from other DECT traffic and against fading.

b) The redundant transmitted data in a) could be coded in order improve the receiver performance for some error scenarios, fx a simple data XOR can handle two consecutive lost packets.

c) Data interleaving could further on be added to improve data recovery for other scenarios. However, this increases latency, which is undesired in some applications.

d) Multi-RFP diversity (network diversity), is a method in which a PP simultaneous communicates with multiple RFP that have overlapping coverage areas. This could be arranged by a PP having multiple parallel connections to the different RFPs, or it could be done using more exotic bearer arrangements where the RFPs are receiving the very same bursts from the PP and the PP receives burst from the RFPs in different bearer positions. The RFPs need to be interconnected and redundant reception of the same data are discarded in the RFP network and the PP discards duplicate data from the RFP network. In critical applications, this arrangement may additional be used to avoid single point of failure on the network side. Potentially, methods of network coding of data could also be applied for improved efficiency.

11.3 FEC

Commonly FEC (Forward Error Corrections) is used in many wireless and wired technologies to add redundant information in order to allow correction of errors in the receiver. Effectively, this improves the link budget from transmitter to receiver. It is assumed that FEC may be implemented in digital / baseband processing domain without affecting the RF circuitry implementation. It can be considered as similar to DSSS techniques used in other wireless standards. In some wireless standards, for example, the use of FEC 1:8 yields a gain of +12 dB.

The gain of using FEC can be utilized in several ways to achieve:

a) To improved reliability and lower the bit error rate and packet error rate within the (current) DECT coverage area.

b) To increase indoor and outdoor range. A 12 dB link budget increase should in theory extend the outdoor range by a factor of 4.

c) Lower transmit peak power. In some (ULE) applications it can be difficult to supply the peak current to DECT transmitter, for example when using coin cell batteries. The gain by FEC can be used to lower the transmit power, and peak current, and still maintain the normal DECT range. There will not be any gain on energy consumed by the transmitter - it is just the peak power that can be lowered.

11.4 Lower symbol rate

Instead of coding (FEC), the link budget can be increased by lowering the transmitted symbol rate. For example, the usual DECT symbol rate of 1,152 Mbaud could be lowered to ½, ¼ or 1/8 of that. To achieve receiver sensitivity-gain by that, both RF transmitter and the receiver filter bandwidths should be adapted. Also digital and baseband processing should be modified.

The gain by lower symbol rate can be utilized to achieve:

a) Longer range. For example, lowering the symbol rate to 1/4, the outdoor range should be doubled according the theory.

b) Reduced multipath issues. The rather short symbols (0,868 μs) used in DECT are problematic in some hard environment with many reflections. Usually delay spread should be below 0,1 μs for good receiver performance. By lowering the symbol rate (increasing symbol length), longer delay spread can be handled. By this reliability in hard indoor environments can be improved.

c) Lower transmitter peak power. As for FEC, the improved receiver sensitivity can be used to lower the transmit power, while maintaining the same link budget and same outdoor range. The lower peak current to the transmitter may easier allow usage of coin cell batteries in for example ULE devices. For unchanged overall link budget, the energy consumption to the transmitter are not lowered.

When lowering the symbol rate, the occupied spectrum will also be smaller. With a smaller modulation bandwidth, it should be possible to arrange multiple "narrow band" channels inside a normal DECT 1,728 MHz channel.

11.5 Coherent modulation and demodulation

The current DECT PHY standard GMSK modulation is not specified to be coherent, which means there is no relationship between modulation and carrier phase. If the GMSK modulation is coherent, then on the receiver side, coherent demodulation can be implemented. According to the theory, coherent demodulation has a sensitivity of 3 dB better than non-coherent demodulation. In modern implementation, coherent modulation and demodulation can be implement by a very small additional complexity and cost. In many other radio technologies, coherent modulation/demodulation is already used commonly. A legacy non-coherent receiver is still able to demodulate signal from a coherent GMSK transmitter.

The improved link budget can be utilized similar to the suggestions in the previous clauses.

11.6 Connectionless services

The current DECT standard already specifies connectionless communication. However, it is not commonly used by DECT products today. It is uncertain how well current standard addresses the need data communication in industrial applications. It is proposed to analyse if improvements and further standardization of connectionless services makes DECT more attractive for industrial applications.

DECT is already today used for broadcast/multicast of audio/streaming data. However, there are some limitations in the current set of standards and no interoperable DECT standard for this. An improvement and further standardization of connectionless services could be topics to further studies.

11.7 Lower energy consumption

Especially for ULE devices, low energy consumption is very important. Other competing wireless technologies have improved energy consumption, so there is a desire also to improve energy consumption in DECT ULE. Probably significant energy consumption improvements can be achieved "just" by more optimized chip implementations. However, there might still be some areas in the DECT ULE standard that could be improved to allow lower the energy consumption.

Topics to consider could be:

1) asynchronous transmission (~ opportunistic transmission);
2) reduced number of used RF channels;
3) lower transmit power;
4) shorter transmit burst when data packets are smaller than full slot payload;
5) wakeup radio using other very low power dedicated radio technology.

11.8 Other topics and ideas

Other ideas for improving performance and features of DECT for certain usage scenarios can be considered. The ideas below have not developed in details, but are listed here for discussion:

a) Is it possible to improve channel scanning and bearer selection for procedures for reducing interference further?

b) Spectrum reservation. When many ULE devices or data devices using fx connectionless communications are sending increasingly amount of data packets, collisions will happen. If the devices are attached to the same network, it could be possible to reserve bearer positions in ways to minimize or avoid collision.

c) The current standard specifies a synchronization port on FPs to allow inter-system synchronization in order to improved capacity and avoid sliding collision. However, the standardized synchronization port is rarely used and impractical to install between independent installations. However, when RFPs do have overlapping coverage area, it would be possible to synchronize the RFPs via the DECT air interface. Currently, there is no interoperable standard for doing this.

d) Indoor location. Other wireless technologies do have or are developing some means for providing location information. It should also be considered if indoor location should be addressed by the DECT standard. Natively that could be means for distance measurement or detection of angel of signal arrival. In some industrial applications and wireless PBX solutions, indoor location information is highly desirable. The bandwidth of the DECT signal will probably limit the achievable accuracy.

Annex A: Background and additional information

A.1 Carrier frequencies for OFDMA 37,5 kHz option

Table A.1: Proposed carrier positions for DECT evolution, 37,5 kHz subcarrier spacing option

DECT evolution OFDMA 37,5 kHz option				Classic DECT	
Channel new number	Start Frequency (MHz)	Center Frequency (MHz)	End Frequency (MHz)	Channel number	DECT channel center frequency
1	1 880,943	1 881,805 5	1 882,668	9	1 881,792
2	1 882,668	1 883,530 5	1 884,393	8	1 883,52
3	1 884,393	1 885,255 5	1 886,118	7	1 885,248
4	1 886,118	1 886,980 5	1 887,843	6	1 886,976
5	1 887,843	1 888,705 5	1 889,568	5	1 888,704
6	1 889,568	1 890,430 5	1 891,293	4	1 890,432
7	1 891,293	1 892,155 5	1 893,018	3	1 892,16
8	1 893,018	1 893,880 5	1 894,743	2	1 893,888
9	1 894,743	1 895,605 5	1 896,468	1	1 895,616
10	1 896,468	1 897,330 5	1 898,193	0	1 897,344
11	1 898,193	1 899,055 5	1 899,918	10	1 899,072
12	1 899,918	1 900,780 5	1 901,643	11	1 900,8
13	1 901,643	1 902,505 5	1 903,368	12	1 902,528
14	1 903,368	1 904,230 5	1 905,093	13	1 904,256
15	1 905,093	1 905,955 5	1 906,818	14	1 905,984
16	1 906,818	1 907,680 5	1 908,543	15	1 907,712
17	1 908,543	1 909,405 5	1 910,268	16	1 909,44
18	1 910,268	1 911,130 5	1 911,993	17	1 911,168
19	1 911,993	1 912,855 5	1 913,718	18	1 912,896
20	1 913,718	1 914,580 5	1 915,443	19	1 914,624
21	1 915,443	1 916,305 5	1 917,168	20	1 916,352
22	1 917,168	1 918,030 5	1 918,893	21	1 918,08
23	1 918,893	1 919,755 5	1 920,618	22	1 919,808
24	1 920,618	1 921,480 5	1 922,343	23	1 921,536
25	1 922,343	1 923,205 5	1 924,068	24	1 923,264

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