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Digital Enhanced Cordless Telecommunications (DECT); Study On Low Data Rate Audio Support

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Foreword

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

Modal verbs terminology

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

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1 Scope

The present document provides an investigation on the usage of low bit rates for DECT audio transmissions, e.g. lower data rates voice connections. The present document describes potential uses cases, technical solutions, and options. Especially, aspects such as optimal audio codec frame intervals, optimization of DECT slot formats, channel codec requirements and audio codec requirements are studied in the present document.

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 300 175-8 (V2.9.1): "Digital Enhanced Cordless Telecommunications (DECT); Common Interface (CI); Part 8: Speech and audio coding and transmission".

[i.2] ETSI TS 126 441: "Universal Mobile Telecommunications System (UMTS); LTE; 5G; Codec for Enhanced Voice Services (EVS); General overview (3GPP TS 26.441)".

[i.3] ETSI TS 103 634: "Digital Enhanced Cordless Telecommunications (DECT); Low Complexity Communication Codec plus (LC3plus)".

[i.4] ETSI EN 300 175-2 (V2.9.1): "Digital Enhanced Cordless Telecommunications (DECT); Common Interface (CI); Part 2: Physical Layer (PHL)".

[i.5] ETSI EN 300 175-3 (V2.9.1): "Digital Enhanced Cordless Telecommunications (DECT); Common Interface (CI); Part 3: Medium Access Control (MAC) layer".

[i.6] Recommendation ITU-T P.800 (08/1996): "Methods for subjective determination of transmission quality".

[i.7] ETSI EN 300 175-5 (V2.9.1): "Digital Enhanced Cordless Telecommunications (DECT); Common Interface (CI); Part 5: Network (NWK) layer".

[i.8] ETSI TR 103 590: "Digital Enhanced Cordless Telecommunications (DECT); Study of Super Wideband Codec in DECT for narrowband, wideband and super-wideband audio communication including options of low delay audio connections (≤ 10 ms framing)".

[i.9] ETSI TS 102 527-3 (V1.7.1): "Digital Enhanced Cordless Telecommunications (DECT); New Generation DECT; Part 3: Extended wideband speech services".

[i.10] Recommendation ITU-T G.722 (09/2012): "7 kHz audio-coding within 64 kbit/s".

[i.11] ETSI EN 300 176-2 (V2.4.1): "Digital Enhanced Cordless Telecommunications (DECT); Test specification; Part 2: Audio and speech".

[i.12] IETF RFC 8251: "Update to the Opus Audio Codec".

[i.13] JH. Chen and J. Thyssen: "The Broadvoice Speech Coding Algorithm", Proceeding of 2007 IEEE International Conference on Acoustics, Speech and Signal Processing, Vol. 4, pp. 537-540, 2007.

[i.14] Recommendation ITU-T P.863 (03/2018): "Perceptual objective listening quality prediction".

[i.15] Recommendation ITU-T P.808 (06/2021): "Subjective evaluation of speech quality with a crowdsourcing approach".

[i.16] ETSI TR 126 952: "Universal Mobile Telecommunications System (UMTS); LTE; Codec for Enhanced Voice Services (EVS); Performance characterization (3GPP TR 26.952)".

[i.17] Recommendation ITU-T G.726: "40, 32, 24, 16 kbit/s Adaptive Differential Pulse Code Modulation (ADPCM)".

3 Definition of terms, symbols and abbreviations

3.1 Terms

Void.

3.2 Symbols

Void.

3.3 Abbreviations

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

AEC	Acoustic Echo Cancellation
ACR	Absolute Category Rating
AMR	Adaptive Multi-Rate
BA	B field control (in A field header)
BCK	B field aCKnowledgement
CLMS	ConnectionLess Message Service
CRC	Cyclic Redundancy Check
CuT	Codec under Test
D8PSK	Differential Eight Phase Shift Keying
DBPSK	Differential Binary Phase Shift Keying
DEC	Decoder
DECT	Digital Enhanced Cordless Telecommunications
DLC	Data Link Control
DPRS	DECT Packet Radio Service
DQPSK	Differential Quadrature Phase Shift Keying
DTX	Discontinuous Transmission
DSP	Digital Signal Processing
ENC	Encoder
EP	Error Protection
EPMR	Error Protection Mode Request
ERP	Ear Reference Point
EVS	Codec for Enhanced Voice Service
FB	Full Band
FEC	Forward Error Correction
FMID	Fixed part MAC IDentity
FP	Fixed Part
GFSK	Gaussian Frequency Shift Keying
HS	Handset
HF	Handsfree
HM	Hamming distance

ID	IDentifier
JB	Jitter Buffer
LCE	Link Control Entity
M1	PP channel selection algorithm for ULE
MAC	Medium Access Control
MOS	Mean Opinion Score
MRP	Mouth Reference Point
NB	Narrow Band
NG	New Generation
NR	Noise Reduction
NWK	Network
PBX	Private Branch Exchange
PMID	Portable part MAC IDentity
PP	Portable Part
PSCN	Primary receiver Scan Carrier Number
QAM	Quadrature Amplitude Modulation
RF	Radio Frequency
RFPI	Radio Fixed Part Identifier
RTP	Real-time Transport Protocol
RS	Reed-Solomon
RSSI	Received Signal Strength Indicator
SAD	Signal Activity Detection
SID	Silence Insertion Description
SWB	Super-Wideband
TCL	Terminal Coupling Loss
T-MUX	Tail Multiplexer
TA	Tail identifier (in A field header)
TPUI	Temporary Portable User Identity
ULE	Ultra Low Energy
VAD	Voice Activity Detection
VoIP	Voice Over IP
WB	Wideband
X-CRC	CRC transmitted in DECT X field

4 Use cases

The support of low data rate audio results in less data being transmitted on the DECT air interface. A shorter DECT packet has several benefits which apply in various use cases:

- Increased spectral density for voice calls or audio connections. If a DECT packet size equivalent to a half slot or less can be achieved, then the overall voice call capacity for a system can potentially be doubled.
  - This is particularly interesting in high density use cases, such as call centres.
  - It can also be used to increase the number of supported devices in a small single cell system, or within a single cell of a larger system, where the number of available time slots can be a limiting factor.
- Reduced power consumption of voice call or audio terminals due to reduced RF transmission duration.
  - This is generally beneficial in all voice use cases, but use of a shorter slot format could enable new cases, such as:
    - Occasional voice connection support in ULE-voice hybrid devices which have a very constrained power budget.
    - Support of emergency calls when the battery state of charge is close to empty.
- Improved system robustness. The use of a lower codec net bit rate allows the use of higher channel protection for a given gross bit rate.
  - This is interesting for use cases with poor transmission conditions, for example due to high interference (e.g. industrial scenarios) or in longer range applications.

5 Latency consideration

The following delay considerations are taken from ETSI EN 300 176-2 [i.11], clause F.3:

Figure 5.1 shows an exemplary breakdown of the contributions of the various elements in a typical DECT system consisting of an FP with VoIP interface (FP type 3 or 5) and any PP type.

The following blocks can be identified:

- JB: Jitter Buffer of the VoIP terminal. Although the delay requirements are valid only for perfect network conditions, in practice a jitter buffer will usually keep a reserve in order to cope with any sudden increase of packet interarrival delay variations (jitter) and/or clock skew between sender and receiver: This is calculated with 20 ms.
- VoIP DEC/VoIP ENC: Assuming 10 ms codec frame size, the joint processing time of encoder and decoder is 10 ms due to the real time constraints. Thus, VoIP encoder and decoder are calculated with 5 ms processing time each. 10 ms delay have been added to the VoIP decoder as reserve, whereas 2,5 ms algorithmic delay have been added to the VoIP encoder.
- DSP: 1,5 ms delay contribution are added caused by any required signal processing in FP send and receive direction and in PP receive direction. In PP send direction, echo cancellation is mandatory in order to achieve the TCL(w) requirements (> 55 dB), 11 ms have been added for this task. Another contribution of 10 ms originates from the framing (accumulation of samples until a block of 10 ms is collected). Finally, a reserve of 5 ms has been assigned such that the DSP block in PP send direction sums up to 26 ms.
- DECT ENC/DEC: Due to real time constraints the joint delay of DECT encoder and decoder cannot exceed 10 ms, thus each of them is calculated as contributing 5 ms. The encoder is calculated with additional algorithmic delay of 2,5 ms.
- RTP pack: Commonly RTP packets contain data representing 20 ms of audio date (RTP pTime), thus another 10 ms delay have been calculated. This is the amount by which any first encoded frame in an RTP packet has to be delayed in order to form the 20 ms RTP packet combined with any second encoded frame.
- The DECT transmission delay has been calculated with 5 ms in each direction.
- The sound propagation time from the electro-acoustic interface (D/A) to the ERP and from the MRP to the electro-acoustic interface (A/D) have been calculated with 0,5 ms for handset/headset (HS) applications (very short distance) and with 3,5 ms for loudspeaking/handsfree (HF) applications (longer distance).

The round-trip delay values shown in Figure 5.1 include 5 ms delay for looping back the signal in the respective counterpart.

In case a PP of type 1d, 2c, 3b and 4b is connected to an FP other than type 3 or 5, the round-trip delays of FP and PP can be determined separately. Else, if connected to an FP with VoIP interface (type 3 or 5), the recommendation is to determine and assess the round-trip delay of the complete terminal as shown in Figure 5.1 ("VoIP-FP + PP round-trip delay"). Hence only requirements for the FP+PP roundtrip delays are given for these FP types.

Technically, PPs of type 5a, 5b, 7a-h, 7j, 8a and 8b can only connect to an FP of type 5. Thus, the recommendation is to determine and assess the round-trip delay of the complete terminal as shown in Figure 5.1 ("VoIP-FP + PP round-trip delay").

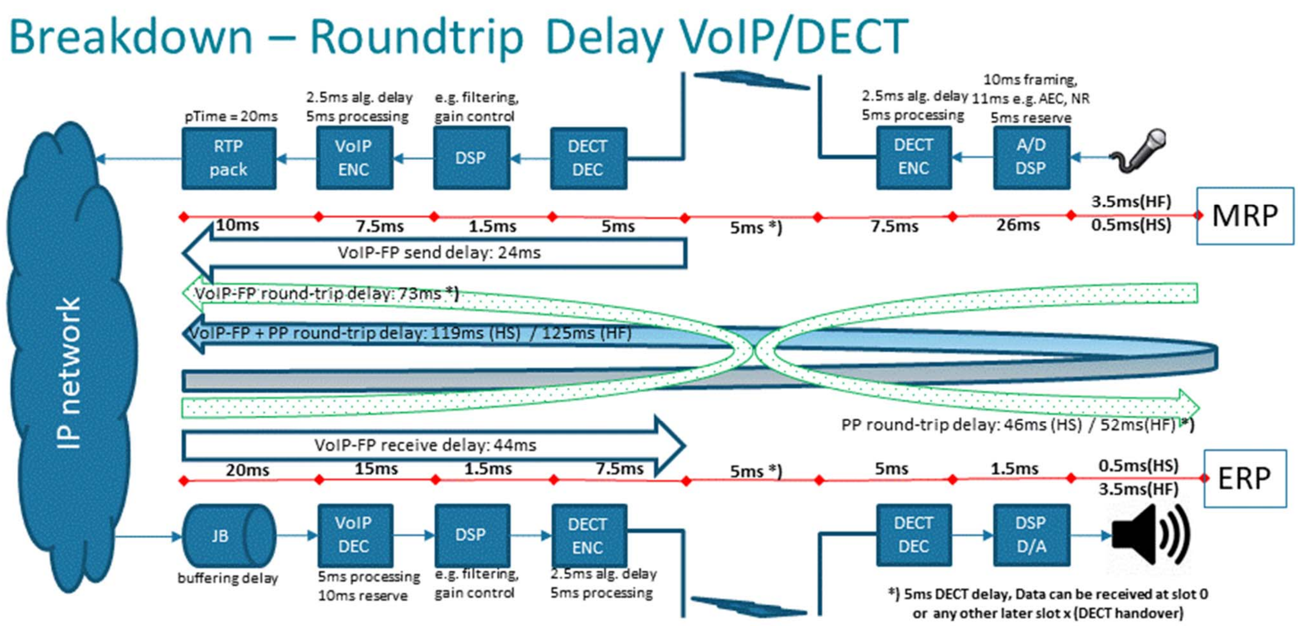

Figure 5.1: Delay breakdown - PP and VoIP FP

Figure 5.1 describes the scenario where LC3plus, ETSI TS 103 634 [i.3], is used as DECT and VoIP codec with a frame duration of 10 ms. Based on Figure 5.1, Table 5.1 shows the delay break down for the following configurations, where some configurations assume that EVS, ETSI TS 126 441 [i.2], is used as DECT codec:

I. LC3plus with 10 ms framing and processing factor 1 (default DSP assumed):

- As described in Figure 5.1.

II. LC3plus with 20 ms framing and processing factor 1:

- Two frames collected after A/D DSP and before DECT ENC.
- Two DECT slots used for transmission, i.e. adding 10 ms delay.
- RTP pack becomes obsolete as framing is already at 20 ms.
- Signal processing delay (filtering, gain control) identical to 1.) as contribution is considered as algorithmic delay rather than computational delay.

III. EVS with 20 ms framing and processing factor 1:

- Algorithmic codec delay changed to 12 ms.
- Assuming four times higher complexity of EVS compared to LC3plus and therefore DECT encoder and decoder processing delay adapted.
- Processing in real-time not possible.

IV. EVS with 20 ms framing and processing factor 4:

- As config III) but all codec processing delays are considered as four times faster.
- DECT codec processing in real-time.

V. EVS with 20 ms framing and processing factor 4 and DECT interval of 20 ms:

- As config IV) but DECT interval is 20 ms (no additional delay of DECT transmission), e.g. by sending complete slots every second frame (see also clause 6.3).
- The system might only work for fully controlled environments, e.g. for enterprise systems.

Table 5.1: Delay contribution for DECT / VoIP interconnection

Configuration		I	II	II	IV	V
System Parameter	Codec	LC3plus	LC3plus	EVS	EVS	EVS
	Codec framing	10 ms	20 ms	20 ms	20 ms	20 ms
	Processor factor	1	1	1	4	4
Delay Source						
Microphone	HS	0,5	0,5	0,5	0,5	0,5
	HF	3,5	3,5	3,5	3,5	3,5
A/D DSP	framing	10	10	10	10	10
	AEC, NR (only alg. delay)	11	11	11	11	11
	reserve	5	5	5	5	5
DEC ENC	algorithmic delay	2,5	2,5	12	12	12
	processing time	5	10	40	10	10
	framing	0	10	10	10	10
DECT radio		5	15	15	15	5
DECT DEC	processing time	5	10	40	10	10
DSP	filtering, gain control (only alg. delay)	1,5	1,5	1,5	1,5	1,5
VoIP ENC	algorithmic delay	2,5	2,5	2,5	2,5	2,5
	processing time	5	5	5	1,25	1,25
RTP pack	20 ms framing	10	0	0	0	0
JB		20	20	20	20	20
VoIP DEC	processing time	5	5	5	1,25	1,25
	reserve	10	10	10	10	10
DSP D/A		1,5	1,5	1,5	1,5	1,5
Transducer	HS	0,5	0,5	0,5	0,5	0,5
	HF	3,5	3,5	3,5	3,5	3,5
Delay Summary	**PP round-trip (HS)**	**46**	**76**	**145,5**	**85,5**	**75,5**
	PP round-trip (HF)	**52,0**	**82,0**	**151,5**	**91,5**	**81,5**
	VoIP round-trip	**73,0**	**83,0**	**152,5**	**85,0**	**75,0**
	VoIP-FP+PP (HS)	**119,0**	**159,0**	**298,0**	**170,5**	**150,5**
	VoIP-FP+PP (HF)	**125,0**	**165,0**	**304,0**	**176,5**	**156,5**
NOTE: All numbers in this table are considered as milliseconds.						

Even though the delay increases from configuration I (default configuration today) to configuration IV (20 ms frame duration using a modern speech codec), the delay values of configuration IV are considered as acceptable.

6 DECT slot format optimizations

6.1 Introduction

One of the main goals of introducing a lower audio data rate is to increase the overall system capacity. In order to achieve this while retaining the current DECT slot structure, there are two potential approaches:

- Use of half slots. This allows up to 24 duplex bearers to exist on a single carrier instead of the current 12, thereby doubling the system capacity. This approach is covered in clause 6.2.

- Use of full slots but reducing the transmission rate such that a packet is sent only once every two (or more) frames. By sharing the same slot position between two (or more) devices, the system capacity can be at least doubled. This approach is covered in clause 6.3.

A further goal of introducing lower rate audio is to save power by reducing the transmit and receive activity. Both of the above approaches also address this goal. However, in the case that these approaches are not feasible, the power reduction goal could also be met by defining a new packet length, somewhere between half slot and full slot. While this no longer increases the capacity in the system which is using this format, the overall DECT system capacity may still be increased by the reduced transmission times. This approach is described in clause 6.4.

6.2 Half slot usage

6.2.1 Standard half slots

The current DECT standards ETSI EN 300 175-2 [i.4] and ETSI EN 300 175-3 [i.5] define the half slot format shown in Figure 6.1.

Size (bits)	16	16	48	16	80	8
Usage	Preamble	Sync	A Field	CRC	B Field (user plane data)	X/Z

Figure 6.1: Standard half slot format

The data rate supported by this format depends on the modulation scheme used for the B field as shown in Table 6.1.

Table 6.1: Standard half slot data rates

B Field Modulation Scheme	B Field Data Size	B Field Data Rate
GFSK or π/2-DBPSK	80 bits	8 kbit/s
π/4-DQPSK	160 bits	16 kbit/s
π/8-D8PSK	240 bits	24 kbit/s
16-QAM	320 bits	32 kbit/s
64-QAM	480 bits	48 kbit/s

The minimum data rate of the audio codecs defined in ETSI EN 300 175-8 [i.1] is 32 kbit/s so at least 16-QAM modulation would be required to support these codecs in a half slot packet. However, higher modulation orders require a higher signal to noise ratio, and also require more complex receiver designs. Hence, lower modulation orders are generally desirable, with support for GFSK being the most robust and lowest complexity option. A complete low-rate codec with error protection scheme should thus ideally support a rate 8 kbit/s if the standard half slot format is used.

6.2.2 Half slot with reduced A field

With the standard half slot format shown in Figure 6.1 it can be seen that the A field and its associated CRC consume a large part of the available data space within the packet. While the A field is used intensively during an audio connection setup, once the connection has been established only limited information is exchanged there. Typical usage may include the following:

- Identities information (N_T). This constitutes the majority of the A field signalling once the call is established. Its primary use is to ensure that the connection is still ongoing with the expected partner device.
- Blind slot information (P_T). This information is needed occasionally to assist potential handover actions.
- Dummy bearer position (P_T). This information is needed occasionally to provide the dummy bearer position to lock to when the connection is released.
- Higher layer signalling (C_T). Most higher layer signalling exchange occurs during the connection setup. After this there is typically only very limited usage, for example for re-keying of the encryption process, or for dialling digits.
- MAC control (M_T). After bearer and connection setup the most common uses are for re-keying of the encryption process and for connection release.

Based on the above, a possible approach could be to use the regular A field initially (potentially without audio content in the B field) until the end of the connection setup and then switch to a format with reduced A field length. This switching could be performed using the same mechanism as already defined in ETSI EN 300 175-3 [i.5] for slot type modification, using ATTRIBUTES_T_req and ATTRIBUTE_T_cfm to configure the modified format.

For the purpose of a bearer handover, the bearer setup could be performed directly with a reduced A-field. This would allow the B-field size to be the same on both bearers for the entire handover procedure thereby ensuring that a seamless handover can be easily implemented.

Note that it is assumed that this bearer is not expected to be used for locking by other devices, otherwise a much larger amount of information would be needed, and the reduction proposed here would not be feasible. This means that if there are other PPs in the system, then the FP needs to retain a regular dummy bearer for locking purposes.

A possible format for the packet with reduced A field size could be as shown in the example of Figure 6.2 where the data and CRC lengths are halved. The corresponding data rates are given in Table 6.2.

Size (bits)	16	16	24	8	112	8
Usage	Preamble	Sync	A Field	CRC	B Field (user plane data)	X/Z

Figure 6.2: Half slot format with reduced A field

Table 6.2: Half slot format with reduced A field, data rates

B Field Modulation Scheme	B Field Data Size	B Field Data Rate
GFSK or π/2-DBPSK	112 bits	11,2 kbit/s
π/4-DQPSK	224 bits	22,4 kbit/s
π/8-D8PSK	336 bits	33,6 kbit/s
16-QAM	448 bits	44,8 kbit/s
64-QAM	672 bits	67,2 kbit/s

With this format only 24 bits remain for control signalling. This is not sufficient to hold the existing message contents so new A field content definitions would need to be created to cover the specific cases that are required.

If the synchronization field remains unchanged compared to the existing standard, then it would be advisable to retain the existing A field header format in order to minimize potential coexistence issues with legacy devices (e.g. trying to decode the packet even though the format cannot be understood). This one-byte field, see Figure 6.3, is already heavily loaded and does not have any unused values at the top level of coding. However, the TA field setting '101' allows for additional coding to be used for the remaining 5 bits of the header (see Table 6.3).

	TA		Q1/ BCK		BA		Q2
a_0	a_1	a_2	a_3	a_4	a_5	a_6	a_7

Figure 6.3: A field header format

Table 6.3: Current combined coding of bits a_3 to a_7

a_3 to a_7	Content of the Frame
00000	Escape
00001	Mesh dummy bearer as defined in clause 9.5.1 in ETSI EN 300 175-3 [i.5]
00010 to 11111	Reserved for future extensions of the standard

A single value of a_3 to a_7 would be sufficient to indicate that the new slot format follows. However, the Q1 and Q2 bits would still be required in a regular duplex connection. Hence, it may be preferable to use four of the values to indicate the new format as shown in Table 6.4. Note that here the Q1 and Q2 positions are retained as today, but they could be moved if a consecutive set of values is preferred.

Table 6.4: Possible future combined coding of bits a_3 to a_7

a_3 to a_7	Content of the Frame
00000	Escape
00001	Mesh dummy bearer as defined in clause 9.5.1 in ETSI EN 300 175-3 [i.5]
00010	New half slot format, Q1 = 0, Q2 = 0
00011	New half slot format, Q1 = 0, Q2 = 1
00100 to 10001	Reserved for future extensions of the standard
10010	New half slot format, Q1 = 1, Q2 = 0
10011	New half slot format, Q1 = 1, Q2 = 1
10100 to 11111	Reserved for future extensions of the standard

The remaining 16 bits of the header would then carry the message type and control information which would need to be newly defined to suit the available space. For example, the 16 bits could be split into a sub-header and content as shown in Figure 6.4.

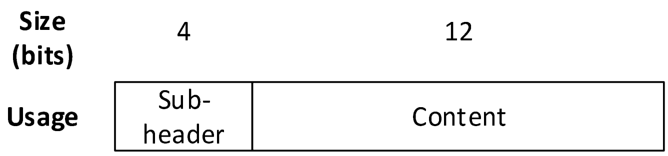

Figure 6.4: Reduced control data format

A possible coding for the sub-header and content is given in Table 6.5. The most problematic aspect appears to be the higher layer messages, which are normally sent as 5-byte fragments. As these no longer fit in a single message, they would need to be split into smaller fragments at the DLC, or each DLC fragment should result in multiple transmissions at the MAC. In Table 6.5 the latter is assumed, with a single 5-byte fragment being split into four parts. This also means that the transmission will take four times longer, though the use cases are limited so this may not be significant.

Table 6.5: Possible coding for new sub-header and content

a_8 to a_{11}	Meaning	a_{12} to a_{23}
0000	Release	Identity, e.g. FMID, or partial PMID
0001	Bearer Handover Request	Identity, e.g. FMID, or partial PMID
0010	Bearer Handover Confirm	Identity, e.g. FMID, or partial PMID
0011	Encryption Start	Req/Cfm/Grant/Reject + key index + partial identity
0100	Encryption Stop	Req/Cfm/Grant + partial identity
0101	Blind Slot Info 1	Blind half slot information for slots 0..5
0110	Blind Slot Info 2	Blind half slot information for slots 6..11
0111	Alternative locking bearer position	Slot number (4 bits) position (2 bits) carrier (6 bits)
1000	Identity	Identity, e.g. FMID, or partial PMID, or partial RFPI, or newly defined identity
1001	Higher layer message, part 1, Ct0	Higher later content, possibly encrypted
1010	Higher layer message, part 1, Ct1	Higher later content, possibly encrypted
1011	Higher layer message, part 2	Higher later content, possibly encrypted
1100	Higher layer message, part 3	Higher later content, possibly encrypted
1101	Higher layer message, part 4	Higher later content, possibly encrypted
1110 to 1111	Reserved for future use	

If the synchronization word were changed, at least in the downlink direction, then there is no longer a need to retain the existing 8-bit header for coexistence reasons, since legacy devices will not lock to these signals. Instead, the header could be used to also include the sub-header information, leaving 16 bits for data. The coding could then be something like that shown in Table 6.6.

Table 6.6: Possible coding for header and content if using a new synchronization word

a_0 to a_5	Meaning	a_6, a_7	a_8 to a_{23}
000000	Release	Q1, Q2	Identity, e.g. FMID, or partial PMID
000001	Bearer Handover Request		Identity, e.g. FMID, or partial PMID
000010	Bearer Handover Confirm		Identity, e.g. FMID, or partial PMID
000011	Encryption Start		Req/Cfm/Grant/Reject + key index + partial identity
000100	Encryption Stop		Req/Cfm/Grant + partial identity
000101	Blind Slot Info 1		Blind half slot information for slots 0..5
000110	Blind Slot Info 2		Blind half slot information for slots 6..11
000111	Alternative locking bearer position		Slot number (4 bits) position (2 bits) carrier (6 bits)
001000	Identity		Identity, e.g. FMID, or partial PMID, or partial RFPI
001001	Higher layer message, part 1, Ct0		Higher later content, possibly encrypted
001010	Higher layer message, part 1, Ct1		Higher later content, possibly encrypted
001011	Higher layer message, part 2		Higher later content, possibly encrypted
001100	Higher layer message, part 3		Higher later content, possibly encrypted
001101 to 111111	Reserved for future use		

In this scheme it is clear that the Ct messages are shorter than those normally expected by higher layers. In order to limit the impact of the new coding as far as possible, it would seem reasonable to insert a further layer of segmentation into the MAC layer. In this way, the higher layer can continue to provide messages which are designed for the 5-byte fragments, but the MAC layer would break it down into the smaller parts as needed, without directly exposing the change to the higher layers.

The main benefits of changing the synchronization word appear to be a reduction of the transmission time for higher layer messages (3 messages instead of 4) and support for longer identity fields (16 bits instead of 12). In addition, there is more room for future message additions due to having additional header bits available.

6.2.3 Treat slots in two frames as a single packet

If the audio codec uses a 20 ms frame length, then a possible way to support this would be to treat two slots in successive frames as a single packet, also from the A field perspective. That is, the normal A field is only included in every second frame as shown in Figure 6.5. The corresponding data rates are given in Table 6.7.

FRAME N

Size (bits)	16	16	48	16	80	8
Usage	Preamble	Sync	A Field	CRC	B Field (user plane data)	X/Z

FRAME N+1

Size (bits)	16	16	8	136	8
Usage	Preamble	Sync	A Field	B Field (user plane data)	X/Z

Figure 6.5: Half slot formats for A field in every second frame

Table 6.7: Half slot format for A field in every second frame, data rates

B Field Modulation Scheme	B Field Data Size (2 frames)	B Field Data Rate
GFSK or π/2-DBPSK	216 bits	10,8 kbit/s
π/4-DQPSK	432 bits	21,6 kbit/s
π/8-D8PSK	648 bits	32,4 kbit/s
16-QAM	864 bits	43,2 kbit/s
64-QAM	1 296 bits	64,8 kbit/s

Here it is assumed that an A field header is included in every frame to limit coexistence issues with legacy devices. As in clause 6.2.2, this header would use a new value for the combined coding of bits a_3-a_7 to indicate frames where no A field is present. One problem with the format shown is that the standalone header has no CRC, and this could lead to the header being incorrectly decoded. If the header is only used for legacy coexistence, then this is not a significant issue, but if the Q1/Q2 feedback bits are also used for bearer feedback then the content should be robust. Hence, either the Q1/Q2 bits should not be used in this format (feedback can be provided in only every second frame), or error detection needs to be added. The latter could be achieved as described in clause 6.2.5.

In this scheme there is no change to the content of the A field, so all existing control messages remain valid.

One aspect that needs to be addressed for this scheme is the tail multiplexing (T-MUX) algorithm (see ETSI EN 300 175-3 [i.5]) which normally assumes presence of an A-field in every frame. If the bearer setup begins in normal mode and then switches to the wanted mode of operation, then a simple option for handling this is just to change the index used for the T-MUX after the mode switch. Instead of using the frame number as the index, the upper three bits of the frame number are combined with the least significant bit of the multiframe number to provide the index. This results in the same T-MUX algorithm as normal but running at half of the frame rate. A drawback of this approach is that the transfer of Ct messages will then take twice as long as normal, so slowing down any higher layer signalling.

An alternative is to use the regular T-MUX with some minor modifications, which would then allow bearer setup directly in the wanted mode of operation. In this case the full C_T bandwidth and M_T bandwidth can be maintained even if transmitting only in every second frame. This can be seen in the standard T-MUX algorithms shown in Table 6.8 and Table 6.9. If the FP transmits the A field in odd frames, and the PP transmits the A field in even frames then there is no direct impact to C_T or M_T transmissions.

Table 6.8: FP T-MUX algorithm

Frame	Priority scheme	Frame	Priority scheme
0	PT, NT	1	MT, CT, NT
2	PT, NT	3	MT, CT, NT
4	PT, NT	5	MT, CT, NT
6	PT, NT	7	MT, CT, NT
8	QT	9	MT, CT, NT
10	PT, NT	11	MT, CT, NT
12	PT, NT	13	MT, CT, NT
14	NT	15	MT, CT, NT

Table 6.9: PP T-MUX algorithm

Frame	Priority scheme	Frame	Priority scheme
0	M_T, C_T, N_T	1	N_T
2	M_T, C_T, N_T	3	N_T
4	M_T, C_T, N_T	5	N_T
6	M_T, C_T, N_T	7	N_T
8	M_T, C_T, N_T	9	N_T
10	M_T, C_T, N_T	11	N_T
12	M_T, C_T, N_T	13	N_T
14	M_T, C_T, N_T	15	N_T

Some minor T-MUX updates would nevertheless be required to handle the missing A field transmissions:

- Some P_T transmissions are required from the FP in order to provide blind slot information and dummy/other bearer position for example. These would need to be scheduled in odd frames when needed.
- There is no longer a guarantee that N_T messages will be scheduled, but these are needed periodically to confirm the connection with the correct partner. Hence, a minimum rate of scheduling for these messages needs to be ensured in odd frames, even if higher priority messages are always pending.
- When transmitting an access request, the PP may transmit an M_T message in any frame. As this is the first message in the setup, this could be treated as a special case and transmitted with A field since no audio data will be included in the packet anyway.

- When responding to an access request, the FP may need to respond with an M_T message in an even frame. As above, because this is the first message in the setup, this could be treated as a special case and transmitted with A field since no audio data will be included in the packet anyway.

If the bearer setup begins in a mode where only every second frame contains the A field, then ideally this should be indicated in the first message from the PP. There are some reserved codes available for both basic and advanced connection M_T messages so this should be possible. Alternatively, the special header in the first response from the PP could be used to indicate this operating mode.

If an FP initiated connection should begin in this specific mode, then this would have to be indicated in the slot type of the page message (for TPUI based paging, see ETSI EN 300 175-5 [i.7], clause 8.2.2). In the slot type field of the page message there are 11 reserved codes available so it would be possible to signal this specific mode.

One potential drawback of transmitting a regular A field on the bearer is that legacy devices may synchronize to the FP transmission and wait for N_T and Q_T messages to synchronize. However, in frames where these might be expected, only CRC failures will be experienced (there is no A field CRC present in the even FP frames). This might result in the PP remaining synchronized unnecessarily on an unusable bearer. In order to avoid this issue, it may be necessary for the FP transmissions to use a different synchronization field or, to preload the A field CRC with a different initialization value, thereby preventing legacy PPs from successfully decoding the transmissions. The PP transmissions should probably retain the normal synchronization field or A field CRC initialization, at least for the initial transmission, otherwise the FP would need to simultaneously detect setups with different synchronization or CRC configurations, which may not be possible with existing hardware.

6.2.4 Combination of reduced A field and two frames as a single packet

It is also possible to combine the options described in clauses 6.2.2 and 6.2.3. In this case there is a packet spread over two frames and also with a reduced A field size as shown in Figure 6.6.

From Table 6.10 it can be seen that with GFSK this approach gains over 1 kbit/s compared to the individual approaches. However, it also has the complexities of both approaches combined.

As in clause 6.2.3, it is assumed here that an A field header is included in every frame to limit coexistence issues with legacy devices. However, this may be unnecessary.

FRAME N

Size (bits)	16	16	24	8	112	8
Usage	Preamble	Sync	A Field	CRC	B Field (user plane data)	X/Z

FRAME N+1

Size (bits)	16	16	8	136	8
Usage	Preamble	Sync	A Field	B Field (user plane data)	X/Z

Figure 6.6: Half slot formats for A field in every second frame

Table 6.10: Half slot format for A field in every second frame, data rates

B Field Modulation Scheme	B Field Data Size (2 frames)	B Field Data Rate
GFSK or π/2-DBPSK	248 bits	12,4 kbit/s
π/4-DQPSK	496 bits	24,8 kbit/s
π/8-D8PSK	744 bits	37,2 kbit/s
16-QAM	992 bits	49,6 kbit/s
64-QAM	1 488 bits	74,4 kbit/s

6.2.5 Shared CRC (or equivalent) for A field and B field

This idea could apply to the options described in clauses 6.2.2, 6.2.3 and 6.2.4.

It is likely that any selected audio codec will include some channel coding, CRC check or similar, which will be used to determine the validity of the audio data. As the A field data sizes are relatively small, it seems that it could be reasonable to share audio codec's protection scheme with the A field. This could reduce total overhead by removing the CRC applied to the A field. In the case of the transmissions without A field in clause 6.2.3, it could also give the single byte header some protection which it does not currently have.

Some examples of how such a format could look when combined with the scheme of clause 6.2.3 are given in clause 8.3.2.

While this type of configuration has the advantage of reducing overhead, any backward compatibility is lost because the slot format will be fundamentally different. At least for FP transmissions, changing the synchronization word should be considered, to avoid devices receiving erroneous data from these transmissions.

6.2.6 Transmit A field only when necessary

This idea could apply to the options described in clauses 6.2.2, 6.2.3 and 6.2.4 and could also be combined with the idea in clause 6.2.5.

As shown in Annex A, for much of the time (at least 85 %) the A field is only used to transfer identity information between PP and FP. This information is largely redundant and is only needed from time to time to confirm that the partner device has not changed unexpectedly. Hence, in frames where a redundant A field would normally be transmitted, the space within the packet can be used to increase the data available for the audio codec and channel coding information.

In order to support such a scheme, the audio codec / channel coding needs to be adaptable to suit the available space within the packet. For some A field transmissions, e.g. Ct messages, the timing is not very critical and the transmission could be notified to the codec in advance so that it can adapt to the available space. For other A field transmissions, e.g. Mt messages for procedures such as encryption control, the timing is critical and the A field should be inserted immediately.

Considering the examples in Annex A, it is possible to estimate the need for different types of A field during an active connection. Using the worst-case figures, i.e. most demanding A field usage, from all examples and assuming that the FP A field is not used for locking (no need for Qt or paging other than blind slot information or the bearer information), the A field usage in Table 6.11 can be derived.

Table 6.11: Estimate for likely A Field usage

A Field Usage	Call Active Phase		Comment
	FP	PP	
Time critical Mt A field	0,1 %	0,1 %	Re-keying, handovers, typically 2 A fields per action
Time critical Ct A field	0,4 %	0,2 %	Re-keying, 12 A fields per 30 s @ FP, 7 A fields per 30 s @ PP
Non-time critical Mt A field	1,1 %	0,0 %	Application specific, 1 A field per second @ FP
Non-time critical Ct A field	5,0 %	1,0 %	Application specific, 5 A fields per second @ FP, 1 A field per second @ PP
Estimate for non-time critical Pt A field	3,0 %	-	Approximately 1 A field per 16 frames
Estimate for non-time critical Nt A field	3,0 %	3,0 %	Approximately 1 A field per 16 frames
No A field needed	87,4 %	95,7 %	

As an example, this scheme could be applied to the proposal in clause 6.2.3, giving the data rates shown in Table 6.12. Here it is assumed that there is still a 1-byte header in all frames where no A field is present. If this is not the case, then the B-field data rate increases by a further 0,8 kbit/s for GFSK.

Table 6.12: Example data rates for scheme in clause 6.2.3 if also omitting unnecessary A fields

B Field Modulation Scheme	A field present (< 15 %)		A field not present (> 85 %)	
	B Field Data Size (2 frames)	B Field Data Rate	B Field Data Size (2 frames)	B Field Data Rate
GFSK or π/2-DBPSK	216 bits	10,8 kbit/s	272 bits	13,6 kbit/s
π/4-DQPSK	432 bits	21,6 kbit/s	544 bits	27,2 kbit/s
π/8-D8PSK	648 bits	32,4 kbit/s	816 bits	40,8 kbit/s
16-QAM	864 bits	43,2 kbit/s	1 088 bits	54,4 kbit/s
64-QAM	1 296 bits	64,8 kbit/s	1 632 bits	81,6 kbit/s

As already indicated, there are some complexities associated with this scheme:

- The codec needs to be able to adapt to the required A field.
- In order to avoid audible impact of the A field, it is likely that regularly repeating transmissions should be avoided, e.g. no regular Pt transmission every 16 frames. This would further affect the protocol scheduling behaviour. Alternatively, the audio codec can schedule the non-time critical messages in a perceptual optimized way, see clause 9.2.
- Even in cases where no A field is required, feedback information (MAC Header Q1, Q2 bits) is still often necessary, so a possibility to include these two bits would still be needed. One option is to include these bits in the channel coder, see clause 8.3.2.3, another option is to include those as part of the audio codec payload, see clause 9.3.2.

6.2.7 Blind slot information for half slots

If using any scheme with half slots, there is a gap in the MAC specification ETSI EN 300 175-3 [i.5] regarding transmission of blind slot information for these slots. For existing slot types, only 12 bits are required to indicate all slots which are available or unavailable. For half slots however, 24 bits are required. Hence, a new method is needed to signal the half slot availability.

Possible options to broadcast the 24 bits are:

1) Use the existing MAC layer information for PT, transmitting the data over two frames.
This approach does not appear to be feasible because there are no free header coding values available for the 'MAC layer information for PT' field.

2) Use a short page message and carry the information within the 36 bits normally used for a device identity.
This approach also seems to have a similar problem to approach 1 above. The LCE header used within the short page message also has no free coding values available. Here it could be considered to repurpose one of the less common cases (e.g. one for DPRS) but the approach still does not match the protocol layering well. The message would logically be handled at the network layer despite the content being aimed at the MAC layer.

3) Redefine the long page message header to create a new page message type for long MAC information.
The long page message type is still defined as 'for further study' in the network layer ETSI EN 300 175-5 [i.7], clause 8.2.3, so it is not used for paging directly. However, long pages are used for the CLMS-fixed service in NG-DECT ETSI TS 102 527-3 [i.9] but in this case the minimum message length always exceeds a single page message. Hence, the coding "all of a long page (first and last)" is not used and could be reused here to create a new paging type. Since it is theoretically possible that there are systems using the "all of a long page (first and last)" setting, the alternative coding should be configured via network layer signalling.

4) Define new page types for the long page combinations where the extend bit is '1' (currently long page messages have the extend bit set to '0' according to ETSI EN 300 175-3 [i.5], clause 9.1.3.1). This would free 4 code points in the P_T header format. However, it is possible that legacy devices decode the extend bit without checking whether the long page content is present, so there could be backward compatibility issues.

5) Use the Q_T system information message. Here there is currently one reserved header code which could potentially be used to signal the blind half slot information. One limitation of this scheme is that Q_T messages are only allowed once per multiframe and there are several Q_T messages to be transmitted, so the update rate would be low. Although this method seems feasible, it is a solution which is inconsistent with the other blind slot information messages that are all carried in P_T messages.

6) Instead of having just one dummy bearer, place a dummy bearer in each half slot so that the dummy bearer can carry the blind slot information for that half slot without requiring any format change. This appears in principle relatively simple, but one disadvantage would be that a PP then needs to receive both dummy bearers in order to get a full view of the blind slot situation. Furthermore, the dummy bearer transmissions then occupy a whole slot, and cannot be removed because there is no other source of blind slot information.

7) Use a longer dummy bearer occupying a half slot instead of a short slot. The 80-bit B field could then be used to carry additional information such as the blind slot information for half slots. This is similar to proposal 6 in that the amount of information transferred is increased to make space for the new information. It has the advantage that the PP still gets all information from a single dummy bearer reception, and it only occupies a single half slot. Drawbacks are that the dummy bearer cannot be removed (the PP should always listen here for blind slot information) and this dummy bearer cannot be combined with a ULE dummy bearer, so a hybrid device would require two dummy bearers.

8) Use higher level modulation in the A field to create additional space for signalling on the dummy bearer. By using π/4-DQPSK, the amount of data in the A field could be doubled, thereby creating additional space for blind slot information for half slots. However, currently the use of such modulation in the A field has not been specified and this would be a major extension to the standard. Furthermore, this would not be compatible with any current device if backward compatibility is required then two dummy bearers would be needed.

Table 6.13 summarizes the advantages and disadvantages of the schemes.

Table 6.13: Advantages and Disadvantages of proposed blind-slot schemes

Scheme	Advantages	Disadvantages
1. Use existing MAC Info paging format	Would be logical, simplest approach.	Not feasible due to no available header codings.
2. Use short page	Would be easy way to get data transfer capacity.	No available LCE header coding, logically this information does not belong in higher layers.
3. Redefine a long page header type	Straightforward.	Risk that some existing devices uses this header, so cannot use such devices in the same system.
4. Redefine meaning of long page + extend bit to exploit unused extend bit	Fairly straightforward.	Higher risk that existing devices interpret the extend bit without considering the other bits in the field.
5. Use Qt message	No impact on any existing message.	Very low update rate, different mechanism to all other blind slot information.
6. Use two dummy bearers, one in each half slot.	Relatively simple change for FP.	Permanently occupies two half slots, requires PP to receive two dummy bearers to get all blind slot information.
7. Extend length of dummy bearer to a half slot.	Straightforward.	No half slot dummy defined in the standard; many additions required. Requires PP to receive dummy even during a call. Not compatible with ULE.
8. Use higher level modulation in the A field	Allows considerably more signalling information in the same (or less) space that a 2-level A field.	Not currently defined in the standard, many additions required. Not possible for devices supporting only GFSK. No compatibility with existing devices.

From the above options, numbers 3 and 7 appear to be promising directions.

For option 3, the page message header coding could be updated as shown in Table 6.14.

Table 6.14: Possible page message header coding

a_9	a_{10}	a_{11}	Length indication
0	0	0	zero length page
0	0	1	short page
0	1	0	full page
0	1	1	MAC resume and control page
1	0	0	not the last 36 bits of a long page
1	0	1	the first 36 bits of a long page
1	1	0	the last 36 bits of a long page
1	1	1	all of a long page (first and last), or, extended MAC broadcast information (see note)
NOTE: The meaning of this field is configured via higher layer signalling.			

This would free 36 bits to carry additional information such as the half slot availability. A possible format for the P_T message in this case could be as shown in Figure 6.7.

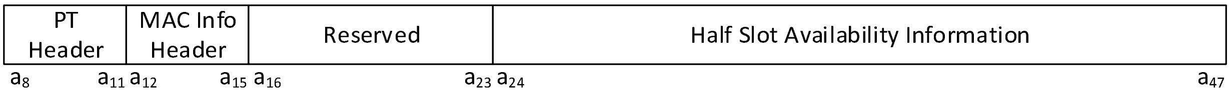

Figure 6.7: Possible extended MAC broadcast information format

In the format of Figure 6.7 a further sub-header is included after the P_T header. This would allow the usage of the message for purposes other than half slot information. For example, it could allow more compact transmission of existing MAC layer broadcast information than using the current zero length page messages.

For option 7, a new B field definition would be required to support dummy bearer information. This could be as shown in Figure 6.8.

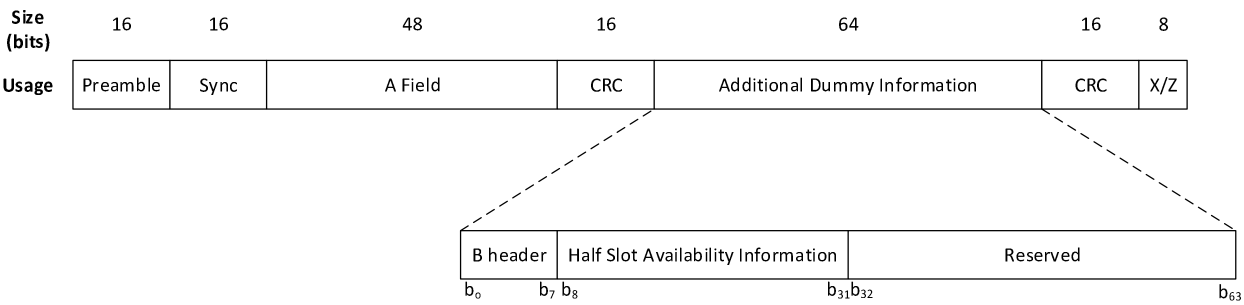

Figure 6.8: Possible half slot dummy format

Note that the addition of the B field opens up the possibility of transmitting other additional information on this type of dummy bearer, but this is not considered further here.

6.3 Full slot usage, with reduced transmission rate ('slow slots')

6.3.1 Introduction

In this scheme a full slot transmission is used, but transmissions are not performed in all frames, leading to a so-called 'slow slot'. An example showing the slot usage for an FP operating with two PPs, where each only uses every second frame, is shown in Figure 6.9. Here it can be seen that the two PPs share the same slot position in a frame, thereby allowing the capacity of the system to be doubled.

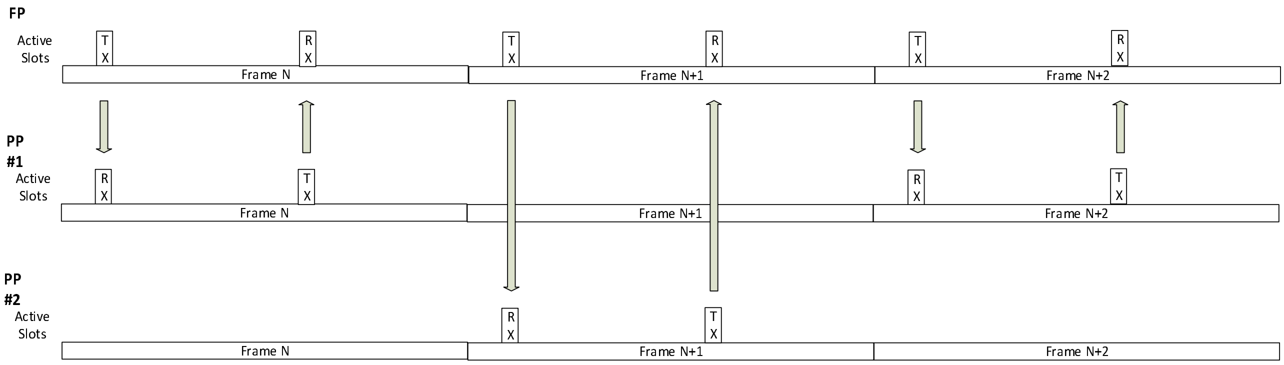

Figure 6.9: Slow slot operation, with transmission in every second frame

The data rate supported by this scheme depends on the modulation scheme used for the B field as shown in Table 6.15.

Table 6.15: Slow slot data rates, with full slot and transmission in every second frame

B Field Modulation Scheme	B Field Data Size	B Field Data Rate
GFSK or π/2-DBPSK	320 bits	16 kbit/s
π/4-DQPSK	640 bits	32 kbit/s
π/8-D8PSK	960 bits	48 kbit/s
16-QAM	1 280 bits	64 kbit/s
64-QAM	1 920 bits	96 kbit/s

Similar to the half slot scheme in clause 6.2.3, this scheme will align better to an audio codec which also delivers an audio packet every 20 ms. Longer gaps between transmission could also be used, but these are less likely to match the audio codec and would introduce additional delay to the audio path.

The main issue that needs to be addressed for this approach is the coexistence with legacy devices. This is because the active slots are not occupied continuously, and this could lead to problems with channel selection in legacy devices. In addition, problems with message multiplexing and blind slot information also need to be considered, similar to those in the half slot cases. These items are covered in the subsequent clauses.

6.3.2 Coexistence

The main issue to be solved with this approach is coexistence with other systems. In regular DECT, there is a general assumption that if a channel is occupied then it is occupied in every frame. Hence, systems can reliably check whether a channel is free by doing a single measurement on a channel. However, if the channel is only occupied in every second frame, then the channel could be detected as free even though there is a connection ongoing there. This could in turn lead to collisions between the transmissions from different systems.

A possible mechanism to resolve this problem could be based on the use of some new rules at the FP which should minimize the risk of collisions.

Rule 1: The FP will try to group two slow slot transmissions over the same slot, whenever it is possible. This will reduce the number of cases where a channel is occupied only every second frame:

- To do this it may use seamless handover and current handover triggering (quality and channel selection) commands.
- New procedures with mandatory actions would need to be written for such commands.

Rule 2: When only one transmission exists in the cycle, the FP will "fill" the unused slow channel position with a dummy (full slot) simplex bearer:

- Such a dummy bearer will not be recognized by legacy devices, but it will impact their channel selection processes because the slot will be detected as occupied due to its high RSSI.

- The dummy can transmit special (new) control information content that can be recognized by DECT devices supporting this feature. This could, for example, be used to carry the blind slot information as indicated in clause 6.3.4.
- DECT devices in the same system which support this feature can perform channel access using the channel that carries the dummy. The FP will change the dummy to a "normal" duplex bearer when this happens. The receive scan cycle for this bearer can follow the regular PSCN sequence or could be fixed to the same carrier as the dummy bearer was using.

Rule 3: If rule 2 is insufficient to ensure that the channel is detected as occupied, the FP could also insert a dummy in the other unused slow channel position (normally direction PP to FP) of the duplex bearer:

- This should make even clearer the identification of the channel as occupied by other devices.
- The drawback is that the channel cannot be directly used for new bearer access by other PTs of the same system. In most cases this can be acceptable because:
  - A PP can initially access using any other channel, then the FP can transfer the connection to the channel following rule 1.
  - In very congested systems where the only available free channel is the slow slot, then, do not insert the dummy bearer in the PP to FP slot, i.e. just follow rule 2.
- Although occupying both channels will prevent most collision cases, there is still a remaining chance that collisions occur because other devices can only detect PP transmissions and not those from the FP, due to their relative locations. Hence, the approach is not perfect, but may be sufficient for the majority of practical cases.

This proposal should enable increased device density in cases where the low-rate audio devices are part of the same system, for example in PBX type applications. However, in an environment where many independent systems exist, increased capacity would not result because the different systems cannot directly exploit each other's unused slow slots. In particular, if rules 2 and 3 are applied then communication between the systems would be required in order to identify the slow slot positions and to coordinate their usage.

6.3.3 A field time multiplexing

Similar to the half slot scheme in clause 6.2.3, the A field signalling here only occurs in every second frame. However, the more optimized solution which is proposed in clause 6.2.3 cannot directly apply in this case, because some bearers would use only even frames and some only odd frames. Nevertheless, the principle of operation from clause 6.2.3 can be applied here with some modification:

- FP transmissions are made in the active frame as if it were an odd frame.
- PP transmissions are made in the active frame as if it were an even frame.
- Some P_T transmissions are required from the FP in order to provide blind slot information and dummy/other bearer position for example. These would need to be scheduled in the active frame when needed.
- There is no longer a guarantee that N_T messages will be scheduled, but these are needed periodically to confirm the connection with the correct partner. Hence, a minimum rate of scheduling for these messages needs to be ensured in the active frames, even if higher priority messages are always pending.

Unlike the approach in clause 6.2.3, during bearer setups, it may not be possible to override the normal scheduling in order to respond in the next frame, because there is no transmission possible there if another device is using the slot. Hence the setup procedures will need to be adapted to take this into account.

6.3.4 Blind slot information

As for half slots, the use of slow slots requires more blind slot information than the current information fields can support. Assuming full slots are transmitted in every second frame for a bearer, then in this case it is required to provide information on:

- Blind full slot information for even frames.

- Blind full slot information for odd frames.

As for half slots, this requires 24 bits of information to be provided. Hence, the approaches proposed in clause 6.2.7 apply equally to this case. If a new dummy bearer type is introduced as proposed in clause 6.3.2, then this could also be used to carry this new blind slot information.

6.3.5 Other considerations

When compared to the half slot solutions of clause 6.2, a significant advantage of this scheme is lower audio delay (see clause 5 for details on this topic). This is because the latency for transmission of a single codec frame would be 5 ms, as in regular DECT. For the half slot schemes the transmission time is 15 ms because the codec frame transmission is split across two DECT frames. However, in order to ensure that the delay is always 5 ms, any handover should also be to the same even or odd frame as the original bearer. If this cannot be ensured, then a 15 ms delay should also be assumed here otherwise there will be an audio disturbance at every handover.

This scheme also has similar coexistence issues to those highlighted in the half slot scheme of clause 6.2.3. That is, legacy devices may synchronize to the FP transmission and wait for N_T and Q_T messages to synchronize. However, in frames where these might be expected, only CRC failures will be experienced because there is no transmission in every second frame. This might result in the PP remaining synchronized unnecessarily on an unusable bearer. In order to avoid this issue, it may be necessary for the FP transmissions to use a different synchronization field or, to preload the A field CRC with a different initialization value, thereby preventing legacy PPs from successfully decoding the transmissions.

Another aspect which has similarity to the scheme in clause 6.2.3 is the signalling that a particular connection will use slow slots. As for the half slot case, ideally this should be indicated in the first message from the PP. There are some reserved codes available for both basic and advanced connection M_T messages so this should be possible. Alternatively, the special header in the first response from the PP could be used to indicate this operating mode.

6.4 New packet length

If the goal of the shorter packet length is primarily to save power, then the strict limit to half slots or to transmissions in only every second frame does not apply. In this case, the variable capacity slot definitions of ETSI EN 300 175-2 [i.4], clause 4.2.1 can be used to define a shortened full slot format. According to these definitions, the format would be defined as P00j, which has a duration of 104+j symbols (assuming Z field is transmitted). For example, to support a 16 kbit/s audio codec while using GFSK, the value of j is 160. This example is shown in Figure 6.10.

Size (bits)	16	16	48	16	160	8	
Usage	Preamble	Sync	A Field	CRC	B Field (user plane data)	X/Z	16 kbit/s

Figure 6.10: Example shortened full slot with j=160

Note that any value of j between 80 and 320 could be used, j=160 is just an example. The value can be adapted to the requirements of the audio codec and modulation schemes to be supported.

It is not possible to setup a connection using this variable length format because there are no available B field identification bit-codings free (see ETSI EN 300 175-3 [i.5], clause 7.1.4). Instead, the bearer would have to be established as a full slot with 320-bit B field and then switched to the shorter length. This switching could be performed using the same mechanism as already defined in ETSI EN 300 175-3 [i.5] for slot type modification, using ATTRIBUTES_T_req and ATTRIBUTE_T_cfm to configure the modified format. There are sufficient free code values in these messages to support new slot formats.

As this format still occupies an entire slot in the system, there is no requirement to introduce new blind slot information to support it.

This scheme only requires definition of a new packet length, and the signalling/capabilities around that, so it is considerably simpler than the schemes in clauses 6.2 and 6.3, but with the major disadvantage that it does not increase the capacity of the system. It does, however, increase the overall spectral density in a given location due to the reduced transmission time.

6.5 Summary

Table 6.16 gives an overview of the schemes present in clause 6 and their main advantages and disadvantages.

Table 6.16: Advantages and disadvantages of proposed slot format optimizations

Scheme	Advantages	Disadvantages
6.2.1. Standard half slot	No significant extensions to standard.	Only 8 kbit/s B-field data rate with GFSK.
6.2.2. Half slot with reduced A field	11,2 kbit/s B-field data with GFSK.	Completely new A field definitions, slower Ct message transfer. Increases air i/f delay for proposed codec by 10 ms.
6.2.3.Treat half slots in 2 frames as a single packet	10,8 kbit/s B-field data with GFSK.	Modified T-MUX required. Regular A fields could lead to confusion in existing PPs, need to take care of compatibility issues (e.g. new sync or CRC). Increases air i/f delay for proposed codec by 10 ms.
6.2.4. Combination of 2 and 3	12,4 kbit/s B-field data rate with GFSK.	Completely new A field definitions. Slower Ct message transfer. Handover procedure needs care to ensure same data on old and new bearers. Modified T-MUX required. Increases air i/f delay for proposed codec by 10 ms.
6.2.5. Shared CRC for A field and B field	Gain of up to 1,6 kbit/s in B-field data rate (depends on new packet CRC) with GFSK.	Need to take care of compatibility issues (e.g. new sync or CRC).
6.2.6. Transmit half slot A field only when necessary	Gain of 3,2 kbit/s in B-field data rate (assuming regular A field) with GFSK, at least 85 % of the time.	Needs codec which can adapt to different available data rates. May need protocol scheduling changes to avoid regularly repeating data.
6.3. Full slot usage with reduced transmission rate (assume every 2nd frame)	16 kbit/s raw data rate with GFSK. No increase in air i/f latency if odd/even frame usage never changes.	Coexistence issues with legacy devices can be largely resolved only in enterprise PBX type systems, standalone devices have no capacity benefit from this scheme.
6.4. New packet length	Straightforward, simple to create for any rate between 8 and 32 kbit/s.	No gain in individual system capacity compared to full slot, though overall spectral density will still be improved.
NOTE: Blind slot information handling is not listed here as this solution is needed in all cases. See clause 6.2.7.		

7 Discontinuous Transmission

7.1 General

Discontinuous Transmission (DTX) means that the transmission of coded audio data is stopped when no active voice or signal is at the input of the encoder. Typically, the input signal is analysed by a Voice or Signal Activity Detector (VAD or SAD) which decides to transmit data or not. In general, this scheme enables a higher capacity, as only active channels are transmitted, and lowers the power consumption, because for non-active channels the transmission power and computational power for the codec is reduced.

7.2 DTX in DECT

The main issue that needs to be addressed for this approach is the coexistence with legacy devices. This is because the active slots are not occupied continuously, and this could lead to problems with channel selection in legacy devices. A similar problem is also discussed for 'slow slots' in clause 6.3.

It should be noted that the use of DTX on audio connections in DECT does not increase the spectral capacity for other DECT audio connections. This is because the slots are still effectively occupied, even when there is no transmission, and the audio transmission can restart at any time. DTX does potentially increase capacity for ULE or DECT-2020 devices which may perform time-limited packet transmissions in DTX gaps. However, this is only possible if the slots are not occupied by FP transmissions. This means that rule 3 in clause 6.3.2 cannot be applied in this case because at least the PP setup channel should remain free. For ULE devices, the PP channel selection process M1 (see ETSI EN 300 175-3 [i.5], clause 11.12.5) requires that both slots in the duplex pair are measured, so in many cases legacy ULE devices will not succeed to use the DTX channel even if the PP transmission slot is left free. Hence, any free spectrum may only be usable for DECT-2020 devices making packet data exchanges which do not use both channels in the duplex pair, or ULE devices operating in the same system as the audio devices, and which are updated to support this use case.

Assuming the FP transmission remains active in order to avoid coexistence issues, then the only radio power save option that remains is to drop transmission of the B field / audio payload at the PP. A similar option already exists in standard DECT which uses B-field suppression to shorten the transmit packet, see ETSI EN 300 175-3 [i.5], clause 5.6.1.1 for details. If a similar approach is taken here then the A field, or reduced A field, would still be transmitted but the audio data would be omitted. If a combined CRC or channel coding is used as described in clause 6.2.5, then a new slot format without audio content would be needed for this case. Similarly, for a scheme where the A field is only transmitted when needed as in clause 6.2.6, a new slot format to indicate the DTX state would be needed. A significant drawback of the B-field suppression scheme is that upon resumption, a collision can occur with a transmission which was not present prior to the B-field transmission being dropped. In this case a handover would probably be needed to avoid the resulting interference.

Besides savings due to reduced radio power, DTX can help saving power by reducing the average computational complexity of the audio codec. DTX is typically designed in a way that a VAD determines the signal activity before the encoding process starts. In case of inactive signals, a Silence Insertion Description (SID) is generated and transmitted. Encoding and decoding of the SID is significantly less complex compared to the main coding algorithm and therefore, power savings are possible. To avoid any interoperability issues with legacy devices as described above, the SID frames could be transmitted in every frame as continuous stream.

8 Channel coder requirements

8.1 Protection strength

A study on the bit error behaviour of a DECT transmission can be found in ETSI TR 103 590 [i.8]. The study was the basis to define the Forward Error Correction (FEC) configuration of LC3plus which consists of four error protection classes which can be adapted to the channel condition. It is assumed that the error behaviour of DECT normal slots is similar to half slots and therefore the conclusions of the study are valid.

Transmissions over DECT can be affected by bit errors as the build-in CRC protection in the X/Y field for the B-field only consists of 4 bits. The A-field is already protected by a 16-bit CRC which corresponds to a Hamming distance of 6. The FEC of LC3plus gives an indication of the required Hamming distance of the B field as outlined in Table 8.1.

Table 8.1: LC3plus FEC configurations

Gross bytes	Error Protection	B-field	Coder rate	Error Protection Bytes (CRC+RS)	Min HM Distance CRC	Min HM Distance RS
40	0	40	32 kbps	-	-	-
40	1	36	28,8 kbps	3 + 1	6	1
40	2	32	25,6 kbps	2 + 6	4	3
40	3	26	20,8 kbps	2 + 12	4	5
40	4	20	16 kbps	2 + 18	4	7

The LC3plus FEC consists of a CRC component and a Reed-Solomon component. By multiplying both values for the Hamming distance, the total distance is given. The minimum Hamming distance of the configuration given in Table 8.1 is 6. This value should define the minimum protection level of a DECT A or B field transmission.

8.2 Error Protection Mode Request (EPMR)

The channel coder of LC3plus can transmit so called Error Protection Mode Requests (EPMR). The EPMR consists of 2 bits indicating which error protection level is requested by the receiving side from the transmitting side. For bi-directional systems, this mechanism is essential to enable fast adaptation to the channel error characteristics. It is therefore recommended that a channel coder supports this feature.

8.3 FEC considerations

8.3.1 Regular half slots

For protecting the B field of a regular half slot (see Figure 6.1), a 16 bit CRC is considered as strong enough. This leads to a maximum payload size for the codec of 6,4 kbps or 64 bits. Any additional FEC needs to be subtracted from the payload size.

8.3.2 Optimized half slots

8.3.2.1 Combined protection for A and B field

Considering the case where two consecutive half slots transport one 20 ms payload (see clause 6.2.3), the combined payload size of A field and B field becomes 36 bytes. This is close to the LC3plus FEC configuration (see Table 8.1) and therefore an adaptive error protection and correction scheme can be applied, protecting A field as well as B field. The FEC bits are interchanged between B field and FEC coder, meaning for low distortion the B field get maximized. This can be applied for static A field sizes as well as for dynamic A field sizes.

One possible optimization is the option to check that only the A field is error free. However, this would require different CRCs for A and B field which is less efficient. To achieve a HM of 6, a CRC of at least 13 bits is required to protect a 48 bit A field and a CRC of 17 bits is required to protect a 30 bytes B field. In comparison a combined CRC only requires 19 bits.

8.3.2.2 Optimized half slot format with static A-field

The following example assumes a half slot format for 20 ms packets, 6 bytes A field in first frame, no A field in second frame and LC3plus FEC or 24 bit CRC over A and B field, as outlined in Figure 8.1.

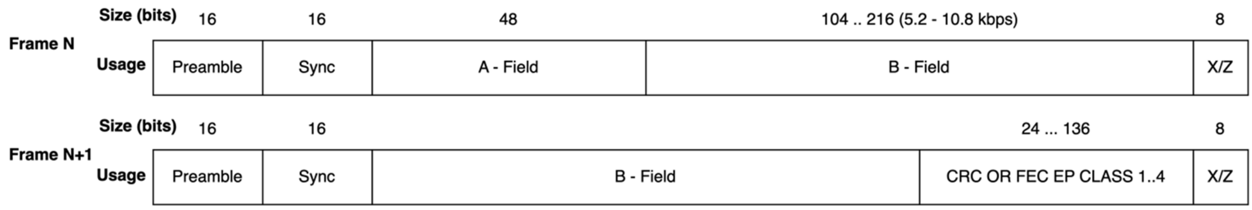

Figure 8.1: FEC for optimized half slot and static A field

Table 8.2 outlines the resulting coder rates depending on the protection mode.

Table 8.2: FEC configuration static A field

Gross bytes	Error Protection	B-field	Coder rate	Error Protection Bytes (CRC+RS)	Min HM Distance CRC	Min HM Distance RS
36	CRC only	27	**10,8 kbps**	3 + 0	6	-
36	1	26	10,4 kbps	3 + 1	6	1
36	2	23	9,2 kbps	2 + 5	4	3
36	3	18	7,2 kbps	2 + 10	4	5
36	4	13	**5,2 kbps**	2 + 15	4	7

8.3.2.3 Optimized half slot format with dynamic A-field

The following example assumes a half slot format for 20 ms packets, dynamically allocated A field of 0, 3 or 6 bytes in the second frame and LC3plus FEC or 24 bit CRC over A and B field are outlined in Figure 8.2. The A field size is signalled with two bits at the beginning of the B field. Alternatively, two bits of the CRC/FEC might be used.

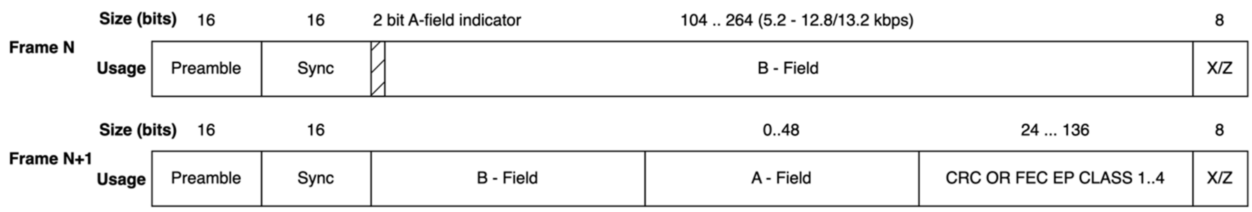

Figure 8.2: FEC for optimized half slot and dynamic A field

Table 8.3 outlines the resulting coder rates depending on the protection mode. As the error protection bytes are identical to Table 8.2, the HM distance is also identical.

Table 8.3: FEC configuration dynamic A field

Gross bytes	A-field	Error Protection	Error Protection Bytes (CRC+RS)	B-field	Coder rate
36	6	CRC	3 + 0	27	10,8 kbps
36	6	1	3 + 1	26	10,4 kbps
36	6	2	2 + 5	23	9,2 kbps
36	6	3	2 + 10	18	7,2 kbps
36	6	4	2 + 15	13	**5,2 kbps**
36	3	CRC	3 + 0	27	12,0 kbps
36	3	1	3 + 1	29	11,6 kbps
36	3	2	2 + 5	26	10,4 kbps
36	3	3	2 + 10	21	8,4 kbps
36	3	4	2 + 15	16	6,4 kbps
36	0	CRC	3 + 0	27	**13,2 kbps**
36	0	1	3 + 1	32	**12,8 kbps**
36	0	2	2 + 5	29	11,6 kbps
36	0	3	2 + 10	24	9,6 kbps
36	0	4	2 + 15	19	7,6 kbps

As mentioned in clause 6.2.6, this slot format implies that 2 bits signalling space is required for feedback information (MAC Header Q1, Q2 bits) for the non-existing A-field case. The 2 bits can be taken from the CRC bytes of the channels coder and if required, the RS bytes further reduced. However, whenever the A-field is transmitted, the feedback information is redundantly transmitted in the channel coder as well. Additionally, the presence and length A-field needs to be indicated.

8.3.3 Regular full slots (slow slots)

'Slow slots' are discussed in clause 6.3. Those consist of regular full slots format sent every 2nd frame to achieve an update interval of 20 ms. Due to regular slot size, the channel coder of LC3plus can be used as outlined in Table 8.4.

Table 8.4: FEC configuration for 'slow slots'

Gross bytes	Error Protection	B-field	Coder rate	Error Protection Bytes (CRC+RS)	Min HM Distance CRC	Min HM Distance RS
40	0	40	16 kbps	-	-	-
40	1	36	14,4 kbps	3 + 1	6	1
40	2	32	12,8 kbps	2 + 6	4	3
40	3	26	10,4 kbps	2 + 12	4	5
40	4	20	8 kbps	2 + 18	4	7

8.3.4 Aspects for audio codecs

To support a flexible protection scheme as outlined in Table 8.3, it is advantageous that the audio codec can operate at any byte aligned bitrate. However, established speech codecs like AMR, AMR-WB and EVS operate at distinct bit rates optimized for 3GPP transport block sizes. For such codecs, the channel codec scheme needs to be adapted and may not support all four EP modes. Table 8.5 gives some examples for EP configs and speech codecs operating at bitrate lower than 12,8 kbps.

Table 8.5

Codec	Codec rate in EP 4	Codec rate in EP 3	Codec rate in EP 2
AMR (note 2)	5,15 kbps / 13 bytes	7,4 kbps / 19 bytes	10,2 kbps / 26 bytes
AMR-WB	6,60 kbps / 17 bytes	8,85 kbps / 23 bytes	12,65 kbps / 32 bytes (note1)
EVS (note 3)	7,2 kbps / 18 bytes	8,0 kbps / 20 bytes	9,6 kbps / 24 bytes
NOTE 1: Rate only possible if adaptive A-field is supported. NOTE 2: AMR supports more rates between 4,75 kbps and 12,2 kbps. NOTE 3: EVS at 13,2 kbps possible, if 3 bytes CRC as protection is used.			

9 Audio codec requirements

9.1 Complexity trade-off

On one hand, more sophisticated audio and speech codecs require higher computational complexity and therefore higher power consumption. On the other hand, these codecs can operate at lower bitrate and therefore provide energy savings regarding the transceiver power consumption.

The transceiver power consumption depends on the transmit power used and on the technology of the device. Based on estimations and measurements made on some existing devices, power consumption savings which may result from changing a duplex bearer from full slot to half slot could be in the order of 1 mW for a device at low transmit power (e.g. 0 dBm), to over 30 mW for a device at high transmit power (e.g. +24 dBm). Note that these values will vary for different devices and so should only be taken as examples, not absolute values.

The power consumption saving for a slow slot duplex bearer using every second frame will be very similar to that for a half slot. At low transmit power, the saving may be slightly more because there is only processing in every second frame, so some additional overhead is saved. At high transmit power the saving may be slightly less because the total transmit time is slightly longer than for half slots.

For the purpose of comparison, it is assumed that a low-rate audio codec would require a processing power equivalent to approximately 100 MHz on an ARM v8 architecture core with DSP extension. In this case the power consumption penalty compared to a hardware G.726 [i.17] codec could be in the order of 1 mW, again depending heavily on the technology of the device.

Based on the above values it can be seen that in most cases there should be a power consumption benefit from the use of a low-rate codec. At low transmit power there may be little or no saving, but also no significant penalty. At higher transmit power the saving could be considerable.

9.2 Audio quality expectations

This clause provides an overview on standardized audio codecs and analyses their capability and audio quality performance regarding a DECT low rate audio system. The following codecs are analysed:

- Opus, v1.3.1 IETF RFC 8251 [i.12], complexity level 10, frame duration of 10 ms.
- Opus, v1.3.1, complexity level 10, frame duration of 20 ms.
- 3GPP EVS, ETSI TS 126 441 [i.2].

AMR-WB was excluded as the EVS characterization [i.16] indicate that EVS performs significantly better than AMR-WB.

Additionally, the WB codecs G.722 [i.10] and BroadVoice [i.13] are included as benchmark. For a DECT system including a channel coder, an audio codec needs to operate in the bitrate range from 5,2 kbps to 13,2 kbps (see clause 8.3.2.3). While Opus can operate at any byte aligned bitrate, EVS is optimized for specific 3GPP transport block sizes. Therefore, the bitrates 6 kbps, 7,2 kbps, 9,6 kbps, 13,2 kbps and 16,4 kbps have been selected for quality comparison. At 5,9 kbps, EVS only operates in variable bitrate mode and therefore this test point has been excluded. All bitrates are constant bitrates, no simulation for dynamic A-field switching was included.

The input signal is always 32 kHz, however some codecs reduce the audio bandwidth below a certain bitrate.

Table 9.1: Audio bandwidth configuration per codec and bitrate

Bitrate	OPUS 10 ms	OPUS 20 ms	EVS
6 kbps	NB	NB	not availalbe
7,2 kbps	NB	NB	WB
9,6 kbps	NB	NB	SWB
13,2 kbps	NB	WB	SWB
16,4 kbps	WB	SWB	SWB

Figure 9.1 shows the Mean Opinion Score (MOS) and the 95 % confidence intervals for an objective analysis following Recommendation ITU-T P.863 [i.14] (Polqa) and a subjective test following Recommendation ITU-T P.808 [i.15] ACR.

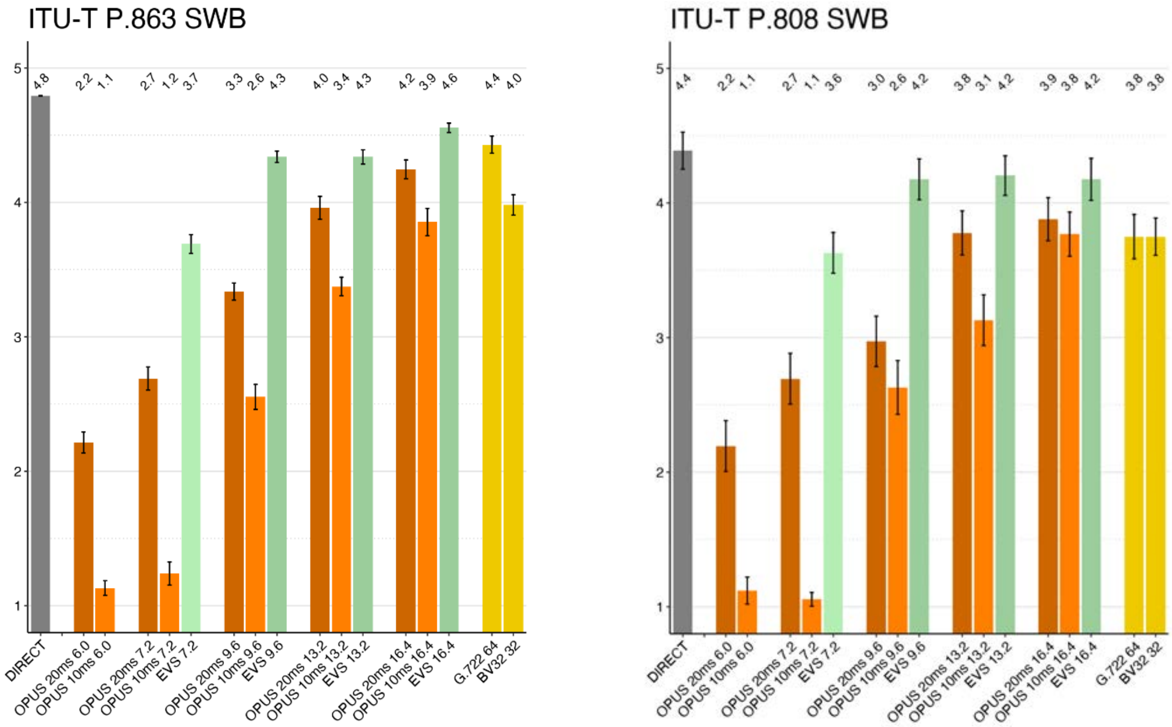

Figure 9.1: Quality overview of standardized audio codecs

The following conclusion can be drawn:

- OPUS using a frame duration of 10 ms performs significantly worse than OPUS using a frame duration of 20 ms. Therefore, using 20 ms frame duration for DECT, the system could gain a significant quality boost.
- The performance of OPUS using 10 ms frame duration is not sufficient for the target data rate of 6,4 kbps assuming half slots with GFSK modulation.
- None of the codecs can offer a consistent audio bandwidth for SWB or WB over the complete target bitrate range from 5,2 kbps to 13,2 kbps.
- At comparable bitrates, EVS performs significantly better than OPUS.
- Technically, OPUS at 20 ms frame duration can operate over the envisioned bitrate range from 5,2 kbps to 13,2 kbps, assuming bandwidth switching is not an issue.
- For WB only operations, EVS might be appropriate candidate for a DECT low rate service.
- An ideal codec provides the following features:
    - Support of at any byte aligned bitrate.
    - Consistent audio bandwidth over complete bit rate range.
    - Quality level comparable to or better than EVS.

9.3 Design constraints

9.3.1 General

The following table summarizes the design constraints for the audio codec which have been derived from the use cases, latency considerations, slot formats optimizations and the resulting available payload sizes including a channel codec.

Table 9.2: Design Constraints for DECT low rates

Parameter	Mandatory values		Justification	
Audio bandwidth	WB and SWB consistently over the entire bitrate range (see note 1)		Support for voice calls	
Sampling frequency	16 kHz and 32 kHz		Support for voice calls	
Audio channels	mono		Support for voice calls	
Frame duration	20 ms	10 ms (if supported) (see note 2)	See clause 6.2 for half slot optimizations	Standard half slot interval
Supported Bitrates	Any byte aligned rate starting from 5,2 kbps to 13,2 kbps	Gross rate equal to 8 kbps or 10 bytes; max. codec rate 6,4 kbps	See Table 8.3 of FEC configuration	See clause 8.3.1
Algorithmic codec delay w/o framing	Equal or less than 12 ms		See clause 5 where delay of EVS codec is considered as acceptable	
Rate Switching	Yes		To support channel coder adaptation and maybe dynamic A-field	
Packet Loss Concealment	Yes		Required for DECT transmissions	
Channel Coder	Similar to LC3plus		Required for DECT transmissions	
NOTE 1: Band limited input should be possible. NOTE 2: Currently there is no known coding scheme which can operate at 10 ms frame duration, SWB at 6,4 kbps.				

Table 9.2 just lists the mandatory operation modes for DECT low rates. It is envisioned that the codec also supports other use cases, e.g. SWB or FB over DECT normal slots. Therefore, additional optional modes are envisioned, e.g. higher bitrates or sample rates.

As described in clause 7.2, a DTX scheme can help to reduce the computational complexity of the audio codec and therefore extend battery lifetime of PPs.

9.3.2 Considerations for slots with dynamic A-field

As outlined in clause 6.2.6, most of the DECT messages are not time critical meaning the transmission does need to happen at a specific point in time. This allows the codec to decide on the perceptually optimal frame, within a given time window, to reduce the bitrate and to transmit the message. This can also minimize any perceptually annoying patterns caused by regularly switched codec bitrate for transmitting DECT messages.

For frames where no A-field is transmitted, 2 bits for feedback information are required (MAC Header Q1, Q2 bits). One option is to use the codec payload to transmit the feedback information. In that case, the codec needs to be able to adapt the bitrate exactly to certain number of required bit and provide some spare bits used for the feedback information. Alternatively, the 2 bits are transmitted within the channel coder, see clause 8.3.2.3.

In order to assess the influence of the dynamic A-field on the audio quality, the following experiment was derived:

- Implementation of a dynamic A-field simulator sending messages according to Table 6.11, FP part:
  - All messages of the FP columns are sent on a strictly regular basis meaning every x^{th} frame.
  - The simulator provides information on message size and timing, i.e. the maximum allowed time span to send the message.

- An audio codec (CuTA) which can read the information on dynamic A-field and adapt the bit rate accordingly:
  - Main bitrate where 12,8 kbps and 7,6 kbps reflecting EP mode 1 and EP mode 4 of Table 8.3.
  - For non-time critical messages, the audio codec can decide at which frame the bitrate is reduced within a certain time window.
- The following configurations have been tested:
  - Dynrate 0: All frames contain A field of 6 bytes.
  - Dynrate 1: Adaptive A-field, 6 bytes per message, always time critical.
  - Dynrate 2: Adaptive A-field, 3 and 6 bytes per message, always time critical.
  - Dynrate 3: Adaptive A-field, 6 bytes per message, time critical and non-time critical.
  - Dynrate 4: Adaptive A-field, 3 and 6 bytes per message, time critical and non-time critical.
  - Dynrate 10: reference where all non-time critical message neglected.
- The results according to Recommendation ITU-T P.808 [i.15] (left) and Recommendation ITU-T P.863 [i.14] (right) are given below.

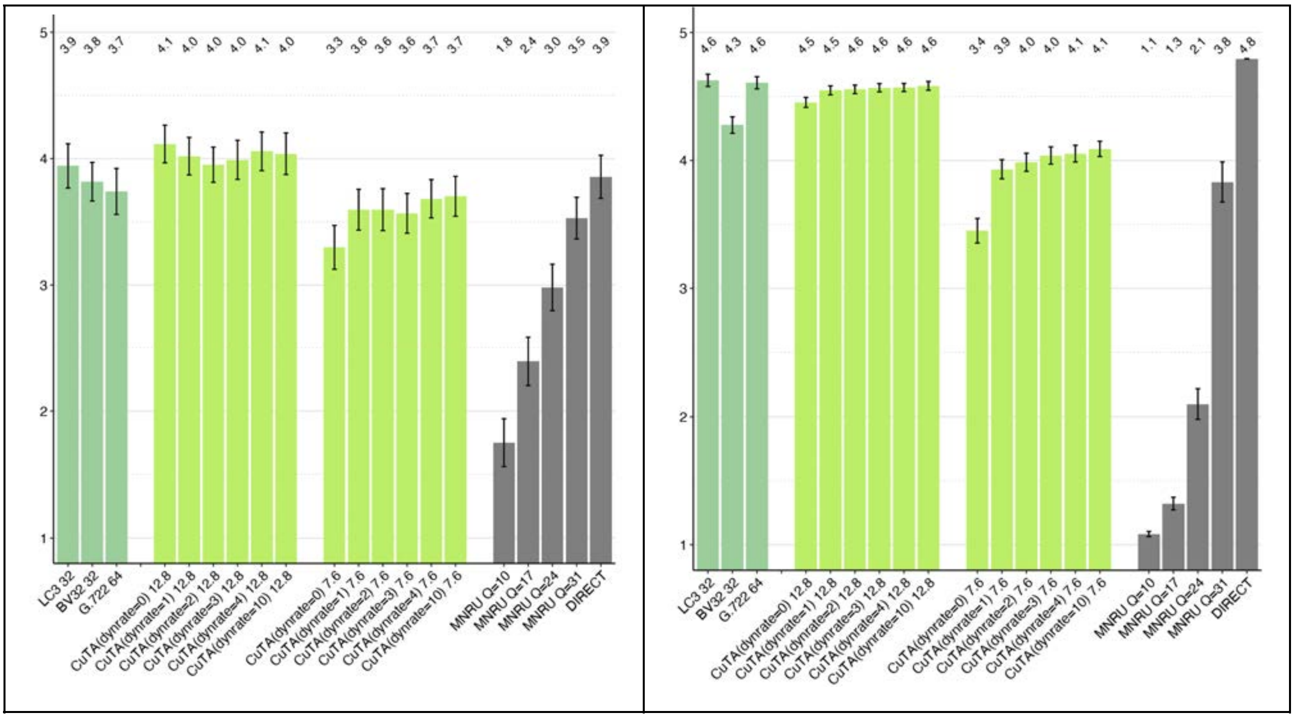

Figure 9.2: Quality evaluation of dynamic A-field, Recommendation ITU-T P.808 [i.15] (left) and Recommendation ITU-T P.863 [i.14] (right)

The following conclusions can be drawn:

- For the low rate (7,6 kbps), the dynamic A-field significantly improves the audio quality.
- For the high rate (12,8 kbps):
  - the benefit of the dynamic A-field is not significant;
  - the static 6 byte A-field (dynrate 0) in the P.808 test is on par with the most efficient dynamic A-field version (dynrate 4). To ensure that this is there is no perceptual effect due to the dynamic A-field causing a specific audible pattern, a second test has been conducted.

As follow up, a comparison test evaluating dynrate configuration 0 and 4 has been conducted on 12 item and with experienced listeners. The items were taken from Recommendation ITU-T P.808 [i.15]. Only 12 items out of 24 were taken, the ones which got the highest scores in favour of dynrate 0. The results of the listening test are plotted in Figure 9.3.

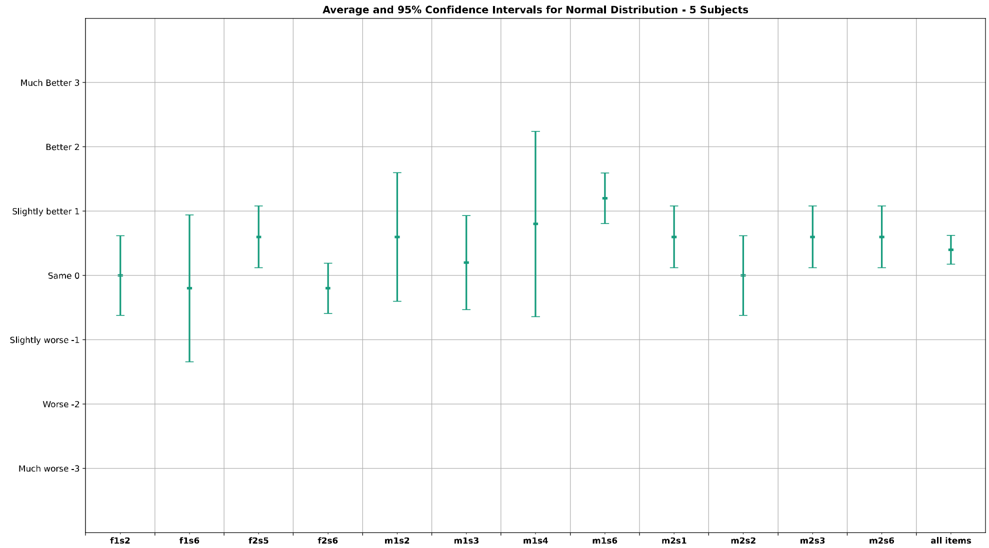

Figure 9.3: Comparison test, positive rating corresponds to Dynrate 4 better than Dynrate 0

According to the comparison test, the dynamic A-field brings a slight improvement.

Table 9.3 lists the pros and cons of the various A-field configurations.

Table 9.3: Pro and Con per A-field configuration

A-field configuration	Pro	Con
0: All frames contain A field of 6 bytes	Changes for DECT system minimal; no extra requirements for codec and channel coder	Significantly worse quality for lower rates/higher EP modes
1: Adaptive A-field, 6 bytes per message, always time critical	Significantly better quality for higher EP modes	Channel coder needs to signal A-field on/off (1 bit)
2: Adaptive A-field, 3 and 6 bytes per message, always time critical	Significantly better quality for higher EP modes	Channel coder needs to signal 3 A-field types (2 bit)
3: Adaptive A-field, 6 bytes per message, time critical and non-time critical	Significantly better quality for higher EP modes	Channel coder needs to signal A-field on/off (1 bit) Audio codec needs to provide interface for dynamic rate adaptation
4: Adaptive A-field, 3 and 6 bytes per message, always time critical and non-time critical	Significantly better quality for higher EP modes	Channel coder needs to signal 3 A-field types (2 bit) Audio codec needs to provide interface for dynamic rate adaptation

According to the results above, it is recommended to implement the 6 byte dynamic A-field. The benefit of using shorter 3 bytes A-field is insignificant. This simplifies the A-field signalling to 1 bit on/off indication. The A-field for non time-critical messages can be transmitted instantaneous or optionally controlled by the audio codec, if such an interface is provided.

9.4 Audio quality recommendations

The audio quality should be at least as good as the BroadVoice codec at 32 kbps (BV32) for clean speech and clean channel condition. A second benchmark codec can be G.722 at 64 kbps and Opus at the same bitrate. For evaluation in the DECT context, the error protection should be always enabled. Mainly, the audio bandwidth WB and SWB in clean channel conditions are of interest. The evaluation procedure is typically Recommendation ITU-T P.800 [i.6] or Recommendation ITU-T P.808 [i.15].

Depending on the envisioned slot configuration for DECT, a dynamic A-field simulator based on Table 6.11 can be included. Clause 9.3.2 shows a listening experiment indicating that audio codecs can reach the expected audio quality at the required bit rates.

10 Conclusions

The goal of this study is to identify options to increase the system capacity, improve the robustness and lower the power consumption of DECT devices.

As shown in clause 9.1, in most cases the total system power consumption can be reduced by utilizing computationally more complex low-rate codecs which can be transmitted over half slots. In such scenarios, the transmitted payload size is significantly reduced, which will also increase the robustness as less bit errors are expected for smaller payloads. Due to the usage of half slots, the number of parallel bi-directional voice calls is twice as high compared to systems using full slots with 32 kbps data.

As an alternative to half slots, a second option to support a low-rate codec is to support a 20 ms frame interval through the usage of a full slot with a reduced transmission rate of every second frame, as shown in clause 6.3. Although this approach may be of interest in some enterprise applications, this option is not generically applicable so is not considered further in these conclusions. Nevertheless, this approach could be further developed by interested parties.

A third option is to define a new packet length, shorter than a full slot, which could carry the data from a reduced rate codec. This is the simplest approach but does not increase the spectral density as much as the above options.

A final option would be to use a half slot with higher order modulation, for example DQPSK or D8PSK. This, however, requires hardware with a DQPSK radio transceiver. Audio codecs operating at the regular DECT frame interval of 10 ms provide insufficient audio quality for the required half-slot bit rates, see clause 9.2. Typical high-quality low-rate codecs, e.g. EVS, operate at frame durations of 20 ms. This implies a higher end-to-end system latency, as outlined in clause 5, however, the resulting latency is in an acceptable range.

In order to increase the available data rate, a payload split over two consecutive half slots can be implemented including a common error protection for A and B-field, see clause 6.2, which leads to a gross rate of 36 bytes for the A/B-field and error protection. By transmitting the A-field only when required, see clause 6.2.6, the maximum payload size of the B-field can be estimated at 12,8 kbps. Audio experiments, see clause 9.3.2, indicate that only A-field sizes 6 bytes and 0 bytes are considered as relevant. These experiments further indicate that the default DECT audio quality determined by Recommendation ITU-T G.722 [i.10] can be reached at this bit rate.

Further, a potential channel coder configuration is given in clause 8.3.2.3. Besides the forward error correction capability, the channel coder also needs to able to indicate the EPMR, the A-field indicator and Q1/Q2 feedback bits. The latter ones are required if the A-field is not present, see clause 6.2.6.

Depending on the envisioned use cases, e.g. WB vs SWB voice calls, either an existing speech codec such as EVS or Opus can be used or for best performance, a new codec design is preferred. In any case, the audio quality has to be comparable to today's DECT performance, see clause 9.4.

Considering the activities required to implement the proposals presented in the present document, the DECT protocol enhancements to support the proposed half slot scheme can be summarized as follows:

1) Add support for transmission of packets without an A field, but with an extended B field instead. This is required in order to split the packet across two frames and to allow dynamic A field transmission. In order to avoid any coexistence issues, it is suggested to use a different synchronization field for bearers using packets of this type, at least in the downlink.

NOTE 1: This also impacts aspects of the packet generation such as scrambling, X-CRC generation and encryption.

2) Define the minimal signalling required in packets without A field, e.g. A field presence indicator, Q1/Q2 feedback, codec error protection mode (see point 5 below).

3) Define the signalling required to switch from a normal bearer to the new bearer type (and potentially also back again). This should include both MAC and NWK layer procedures. Handling of handovers should also be considered.

4) Define new A field time multiplexing rules to support the presence of an A field in only every second frame and only when necessary. This may require modification of some related MAC procedures if their timing is changed by the modified multiplexing. Support for codecs which provide information on preferred occasions for signalling may also be included.

5) Extend network layer signalling to include any required 20 ms audio codec(s) in the codec list.

6) Add support for combined error protection and/or forward error correction across a two-frame transmission, including A field (if present) and B field over 36 gross bytes.

7) Add new signalling for half-slot blind slot information. Option 3 from clause 6.2.7 is suggested.

Note that not all of the above points need to be standardized in order to enable the proposed half slot scheme:

- For proprietary implementations, e.g. using escape A fields, proprietary protocol, and proprietary codec payload, only item 1 and possibly the A field presence part of 2 is necessary in the base standard.

- To support devices using regular MAC and network layer signalling, then points 2, 3, 4 and 5 are also required.

- Points 6 and 7 are optional, depending on whether there is a wish to standardize these aspects.

The DECT protocol enhancements required to support an alternative scheme using an extended packet length can be summarized as follows:

1) Define a new packet type, for example P00j with j=128 for a 12,8 kbit/s codec and CRC or FEC bytes.

NOTE 2: This also impacts aspects of the packet generation such as scrambling, X-CRC generation and encryption.

2) Define the signalling required to switch from a normal bearer to the new bearer type (and potentially also back again). This should include both MAC and network layer procedures. Handling of handovers should also be considered.

3) Extend network layer signalling to include the required 20 ms audio codec(s) in the codec list.

Note that for a proprietary solution, only point 1 needs to be standardized, while points 2 and 3 are needed to support regular signalling.

For a scheme using short slots with DQPSK, the only protocol enhancement needed is to extend the regular network layer signalling to include the 20 ms audio codec(s) in the codec list.

While it is not proposed to standardize a new audio codec in the scope of the present document, some DECT standard additions are nevertheless recommended to support the use of externally defined codecs:

1) Definition of the minimum requirements for the codec, e.g. frame rate, payload size.

2) Definition of the configuration(s) needed for existing codecs, so that signalling the use of a certain codec results in a defined configuration.

3) Definition of new audio types to cover any new/proprietary codec if none of the existing types is appropriate. For signalling proprietary codecs, the option to use vendor tag and codec ID can be considered.

Annex A: Example A Field Usage

Tables A.1 to A.6 give examples of the A field usage in several different audio devices from different manufacturers. The usage was measured by observing the air interface message exchanges on commercially available products. The following examples are included:

- Table A.1 and Table A.2: Headset, manufacturer #1, used with a personal computer for a phone conference.
- Table A.3 and Table A.4: Cordless telephone, manufacturer #2, used for an external call.
- Table A.5 and Table A.6: Cordless telephone, manufacturer #3, used for an internal call.

The call setup phase was considered here to be complete when the initial Ct message exchange was complete.

Table A.1: A Field Usage, headset, manufacturer #1, FP transmissions

A Field Type	Overall Call		Call Setup Phase		Call Active Phase	
	Number	Percentage	Number	Percentage	Number	Percentage
All	62 301	100,00 %	249	100,00 %	62 052	100,00 %
Ct	455	0,73 %	49	19,68 %	406	0,65 % (note 1)
Nt	53 732	86,25 %	133	53,41 %	53 599	86,38 %
Qt	3 930	6,31 %	16	6,43 %	3 914	6,31 %
Mt	658	1,06 %	13	5,22 %	645	1,04 % (note 2)
Pt	3 526	5,66 %	38	15,26 %	3 488	5,62 %
NOTE 1: The Ct messages occur as bursts of 3-4 A fields, every 5-10 seconds. NOTE 2: The Mt messages during the active call are proprietary escape messages, sent every 1 second.						

Table A.2: A Field Usage, headset, manufacturer #1, PP transmissions

A Field Type	Overall Call		Call Setup Phase		Call Active Phase	
	Number	Percentage	Number	Percentage	Number	Percentage
All	62 301	100,00 %	249	100,00 %	62 052	100,00 %
Ct	792	1,27 %	42	16,87 %	750	1,21 % (note)
Nt	61 487	98,69 %	193	77,51 %	61 294	98,78 %
Mt	19	0,03 %	12	4,82 %	7	0,01 %
MT-first	3	0,00 %	2	0,80 %	1	0,00 %
NOTE: The Ct messages occur as bursts of 5-10 A fields, every 5-10 seconds.						

Table A.3: A Field Usage, cordless telephone, manufacturer #2, FP transmissions

A Field Type	Overall Call		Call Setup Phase		Call Active Phase	
	Number	Percentage	Number	Percentage	Number	Percentage
All	60 359	100,00 %	442	100,00 %	59 917	100,00 %
Ct	34	0,06 %	27	6,11 %	7	0,01 %
Nt	52 885	87,62 %	327	73,98 %	52 558	87,72 %
Qt	3 773	6,25 %	27	6,11 %	3 746	6,25 %
Mt	11	0,02 %	3	0,68 %	8	0,01 %
Pt	3 656	6,06 %	58	13,12 %	3 598	6,00 %

Table A.4: A Field Usage, cordless telephone, manufacturer #2, PP transmissions

A Field Type	Overall Call		Call Setup Phase		Call Active Phase	
	Number	Percentage	Number	Percentage	Number	Percentage
All	60 373	100,00 %	442	100,00 %	59 931	100,00 %
Ct	56	0,09 %	40	9,05 %	16	0,03 %
Nt	60 309	99,89 %	398	90,05 %	59 911	99,97 %
Mt	5	0,01 %	3	0,68 %	2	0,00 %
MT-first	3	0,00 %	1	0,23 %	2	0,00 %

Table A.5: A Field Usage, cordless telephone, manufacturer #3, FP transmissions

A Field Type	Overall Call		Call Setup Phase		Call Active Phase	
	Number	Percentage	Number	Percentage	Number	Percentage
All	31 805	100,00 %	1 139	100,00 %	30 666	100,00 %
Ct	1682	5,29 %	76	6,67 %	1 606	5,24 % (note 1)
Nt	26 911	84,61 %	936	82,18 %	25 975	84,70 %
Qt	1988	6,25 %	72	6,32 %	1 916	6,25 %
Mt	29	0,09 %	10	0,88 %	19	0,06 % (note 2)
Pt	1 195	3,76 %	45	3,95 %	1 150	3,75 %
NOTE 1: The Ct messages occur as bursts of 12 A fields every 30 seconds, and 5 A fields every second. NOTE 2: The Mt messages are mainly used for re-keying every 30 seconds, 2 messages sent each time.						

Table A.6: A Field Usage, cordless telephone, manufacturer #3, PP transmissions

A Field Type	Overall Call		Call Setup Phase		Call Active Phase	
	Number	Percentage	Number	Percentage	Number	Percentage
All	31 797	100,00 %	1 136	100,00 %	30 661	100,00 %
Ct	408	1,28 %	43	3,79 %	365	1,19 % (note 1)
Nt	31 337	98,55 %	1 079	94,98 %	30 258	98,69 %
Mt	50	0,16 %	12	1,06 %	38	0,12 % (note 2)
MT-first	2	0,01 %	2	0,18 %	0	0,00 %
NOTE 1: The Ct message exchanges occur as bursts of 7 A fields every 30 seconds, and a single A field every second. NOTE 2: The Mt messages are mainly used for re-keying every 30 seconds, 2 messages sent each time.						

History

Document history		
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