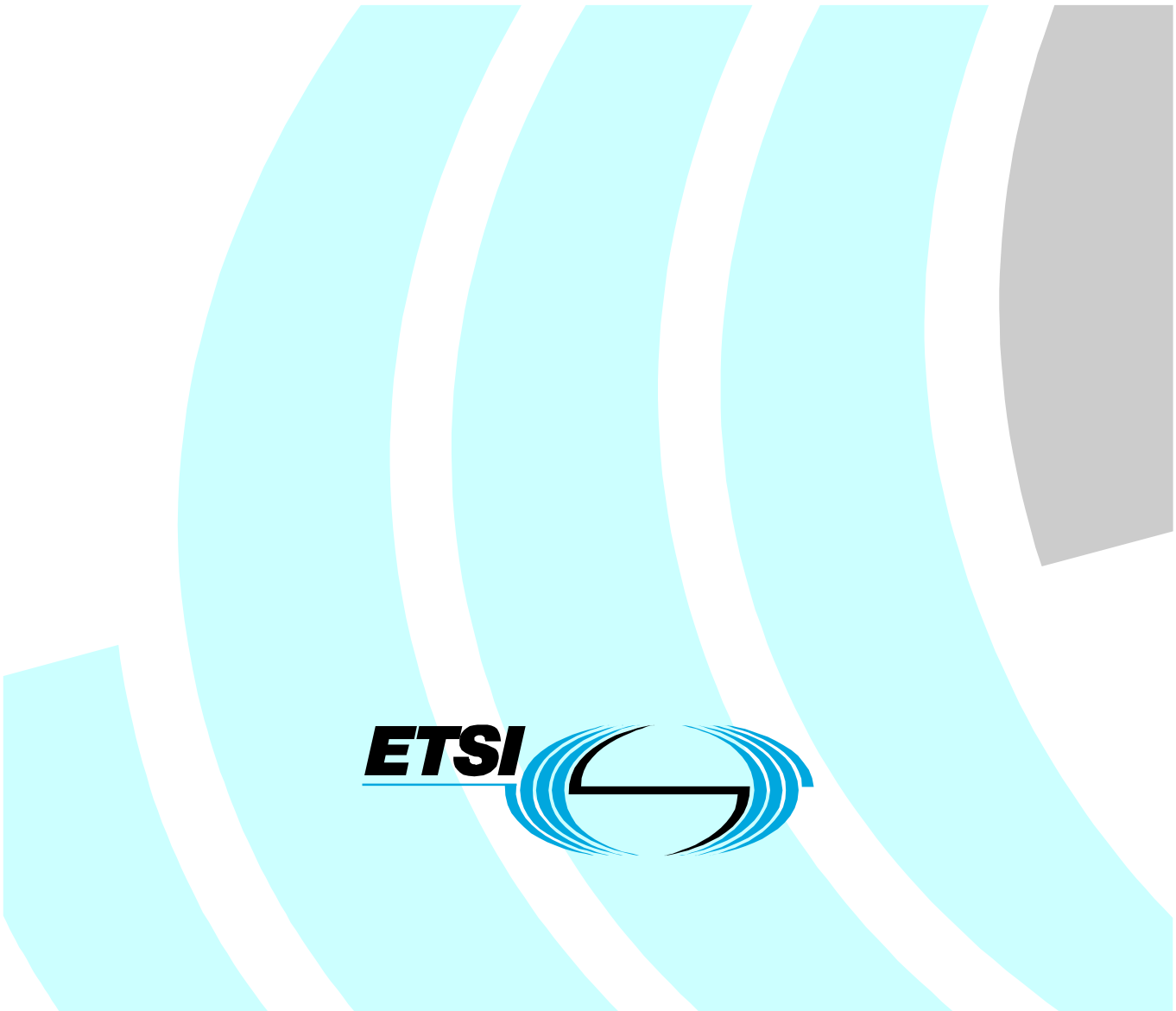
ETSI TS 102 527-1 V1.1.1 (2007-04)

Technical Specification

Digital Enhanced Cordless Telecommunications (DECT); New Generation DECT; Part 1: Wideband speech



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Contents

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Foreword

This Technical Specification (TS) has been produced by ETSI Project Digital Enhanced Cordless Telecommunications (DECT).

The present document is based on EN 300 175 parts 1 [1] to 8 [8] and EN 300 444 [13]. General attachment requirements and speech attachment requirements are based on EN 301 406 [10] (replacing TBR 006 [25]) and EN 300 176-2 [9] (previously covered by TBR 010 [26]).

The present document has been developed in accordance to the rules of documenting a profile specification as described in ISO/IEC 9646-6 [11].

The information in the present document is believed to be correct at the time of publication. However, DECT standardization is a rapidly changing area, and it is possible that some of the information contained in the present document may become outdated or incomplete within relatively short time-scales

The present document is part 1 of a multi-part deliverable covering the New Generation DECT as identified below:

Part 1: "Wideband speech";

Part 2: "Support of transparent IP packet data";

Part 3: "Support of phase 2 services".

1 Scope

The present document specifies a set of functionalities of the New Generation DECT.

The New Generation DECT provides the following basic new functionalities:

- Wideband voice service.
- Packet-mode data service supporting Internet Protocol with efficient spectrum usage and high data rates.

The present document describes the first part: Wideband speech service. For the description of the support of transparent IP packet data, see TS 102 527-2 [14].

All New Generation DECT devices will offer at least one or both of these services. If the device offers the wideband voice service, it will support also the DECT standard 32 kbit/s voice service according to EN 300 444 (GAP) [13].

All DECT devices claiming to be compliant with this Application Profile will offer at least the basic services defined as mandatory. In addition to that, optional features can be implemented to offer additional DECT services.

The aim of the present document is to guarantee a sufficient level of interoperability and to provide an easy route for development of DECT wideband speech applications, with the features of the present document being a common fall-back option available in all compliant to this profile equipment.

2 References

The following documents contains provisions which, through reference in this text, constitute provisions of the present document.

- References are either specific (identified by date of publication and/or edition number or version number) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version applies.

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

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

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

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

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

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

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

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

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

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

[9] ETSI EN 300 176-2: "Digital Enhanced Cordless Telecommunications (DECT); Approval test specification; Part 2: Speech".

[10] ETSI EN 301 406: "Digital Enhanced Cordless Telecommunications (DECT); Harmonized EN for Digital Enhanced Cordless Telecommunications (DECT) covering essential requirements under article 3.2 of the R&TTE Directive; Generic radio".

[11] ISO/IEC 9646-6: "Information technology - Open Systems Interconnection - Conformance testing methodology and framework - Part 6: Protocol profile test specification".

[12] ISO/IEC 9646-7: "Information technology - Open Systems Interconnection - Conformance testing methodology and framework - Part 7: Implementation Conformance Statements".

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

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

[15] ITU-T Recommendation G.726 (12/1990): "40, 32, 24, 16 kbit/s Adaptive Differential Pulse Code Modulation (ADPCM) ".

[16] ITU-T Recommendation G.711 (11/1988): "Pulse code modulation (PCM) of voice frequencies".

[17] ITU-T Recommendation G.722 (11/1988): "7 kHz audio-coding within 64 kbit/s".

[18] ITU-T Recommendation G.729.1 (05/2006): "G.729 based Embedded Variable bit-rate coder: An 8-32 kbit/s scalable wideband coder bitstream interoperable with G.729".

[19] ISO/IEC JTC1/SC29/WG11 (MPEG) (2006): "International Standard ISO/IEC 14496-3: 2005/FDAM1:2006(E) "Coding of audio-visual objects Part 3: Audio AMENDMENT 1: Low Delay AAC profile"".

[20] ISO/IEC JTC1/SC29/WG11 (MPEG): "International Standard ISO/IEC 14496-3: "Coding of audio-visual objects: Audio, 2005"".

[21] ITU-T Recommendation P.311 (06/2005): "Transmission characteristics for wideband (150-7000 Hz) digital handset telephones".

[22] IETF RFC 3640: "RTP Payload Format for Transport of MPEG-4 Elementary Streams".

[23] IETF RFC 3016: "RTP Payload Format for MPEG-4 Audio/Visual Streams".

[24] IETF RFC 4749: "RTP Payload Format for the G.729.1 Audio Codec".

[25] ETSI TBR 006: "Digital Enhanced Cordless Telecommunications (DECT); General terminal attachment requirements".

[26] ETSI TBR 010: "Digital Enhanced Cordless Telecommunications (DECT); General terminal attachment requirements: Telephony applications".

[27] IETF RFC 3261: "SIP: Session Initiation Protocol".

3 Definitions, symbols and abbreviations

3.1 Definitions

For the purposes of the present document, the terms and definitions given in EN 300 444 [13] and the following apply:

New Generation DECT: a further development of the DECT standard introducing wideband speech, improved data services, new slot types and other technical enhancements

wideband speech: voice service with enhanced quality compared to ADPCM G.726 and allowing the transmission of a vocal frequency range of at least 150 Hz to 7 kHz, and fulfilling the audio performance requirements described in the ITU-T Recommendation P.311 [21]

super-wideband speech: voice service with enhanced quality compared to ADPCM G.726 and allowing the transmission of a maximum vocal frequency of at least 14 kHz

3.2 Symbols

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

M	mandatory to support (provision mandatory, process mandatory)
O	optional to support (provision optional, process mandatory)
I	out-of-scope (provision optional, process optional) not subject for testing
C	conditional to support (process mandatory)
N/A	not applicable (in the given context the specification makes it impossible to use this capability)

Provision mandatory, process mandatory means that the indicated feature service or procedure shall be implemented as described in the present document, and may be subject to testing.

Provision optional, process mandatory means that the indicated feature, service or procedure may be implemented, and if implemented, the feature, service or procedure shall be implemented as described in the present document, and may be subject to testing.

NOTE: The used notation is based on the notation proposed in ISO/IEC 9646-7 [12].

3.3 Abbreviations

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

AAC	Advanced Audio Coding (MPEG)
AC	Authentication Code
ADPCM	Adaptive Differential Pulse Code Modulation
AI	Air Interface
BCD	Binary Coded Decimal
CC	Call Control
CI	Common Interface
DECT	Digital Enhanced Cordless Telecommunications
DLC	Data Link Control
DTMF	Dual Tone Multi-Frequency
ER	Error Resilient (MPEG)
FP	Fixed Part
FT	Fixed radio Termination
GAP	Generic Access Profile
GAP	Generic Access Profile
IA	Implementation Alternative
IE	Information Element
IP	Internet Protocol
IPUI	International Portable User Identity
ISDN	Integrated Services Digital Network

IWU	InterWorking Unit
KS	PP authentication Session Key
LA	Location Area
LD	Low Delay (MPEG)
LLME	Lower Layer Management Entity
LSB	Least Significant Bit
MAC	Medium Access Control
MM	Mobility Management
MSB	Most Significant Bit
NB	Narrow Band
NG	New Generation
NG-DECT	New Generation DECT
NWK	NetWorK
P	Public (environment)
PA	Portable Application
PABX	Private Automatic Branch eXchange
PARK	Portable Access Rights Key
PHL	PHysical Layer
PP	Portable Part
PRA	Primary Rate Access (ISDN)
PT	Portable radio Termination
R/B	Residential/Business (environment)
RFP	Radio Fixed Part
S/T	ISDN S/T Interface
SARI	Secondary Access Rights Identity
TCL	Telephone Coupling Loss
TPUI	Temporary Portable User Identity
TRUP	TRansparent UnProtected service
U	ISDN U-Interface
WB	Wideband

4 Description of Services

4.1 Enhanced wideband speech

In traditional telephony applications the supported bandwidth is 3,1 kHz (300 Hz to 3,4 kHz). For better speech quality and a more natural sound, a bandwidth of at least 150 Hz to 7 kHz should be supported and may be extended even further.

New Generation DECT improves audio quality by implementing wideband enhanced quality audio codecs. All New Generation DECT wideband speech devices shall implement wideband (150 Hz to 7 kHz) audio (16 kHz frequency sampling). DECT devices supporting wideband audio shall support the speech coding format according to ITU-T Recommendation G.722 [17]. In addition to that, other wideband and superwideband audio codecs, providing even better audio quality, may be implemented.

In order to transport the higher bitrate of the new enhanced codecs, the bitrate per channel at the air interface is doubled from 32 kbit/s in traditional DECT to 64 kbit/s.

All New Generation DECT wideband speech devices shall be backward compatible with traditional DECT 32 kbit/s voice (GAP) devices. New PPs shall operate with legacy base stations, and new bases shall support existing PPs. In such cases, the voice quality is the traditional DECT quality (32 kbit/s ADPCM).

4.1.1 Audio performance requirements

New Generation DECT handsets shall fulfil the audio performance requirements described in ITU-T Recommendation P.311 [21].

4.2 Wideband speech scenarios

The following scenarios are envisaged:

4.2.1 Internal calls inside a New Generation DECT system

In such a case, wideband (150 Hz to 7 kHz) communication is possible between both terminals without any special issue.

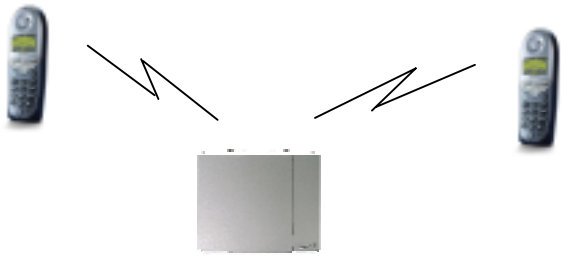

Figure 1: Internal wideband call

4.2.2 Calls between two New Generation DECT systems interconnected by ISDN

Two subscribers owning New Generation DECT base stations and handsets could establish a wideband voice communication between them if the DECT FPs are interconnected by an ISDN network with digital U or S/T interface, (or PRA) to the local exchange. The ISDN call should be digital unrestricted 64 kbit/s.

The scenario is also possible for business customers using PABX with DECT support and digital links to the public exchange.

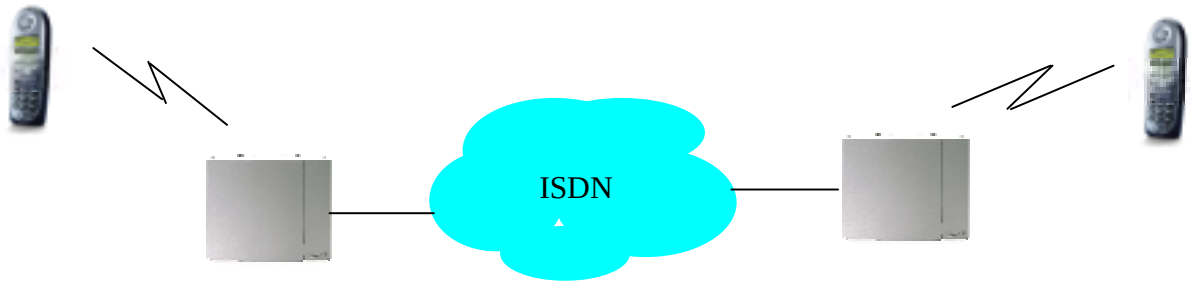

Figure 2: Wideband call via ISDN

4.2.3 Calls between two New Generation DECT systems interconnected by IP packet based network

Two subscribers owning New Generation DECT base stations and handsets, and interconnected via VoIP over an IP packet based network, could establish a wideband voice communication between them.

The IP packed based network can be either the Internet or a dedicated IP based network.

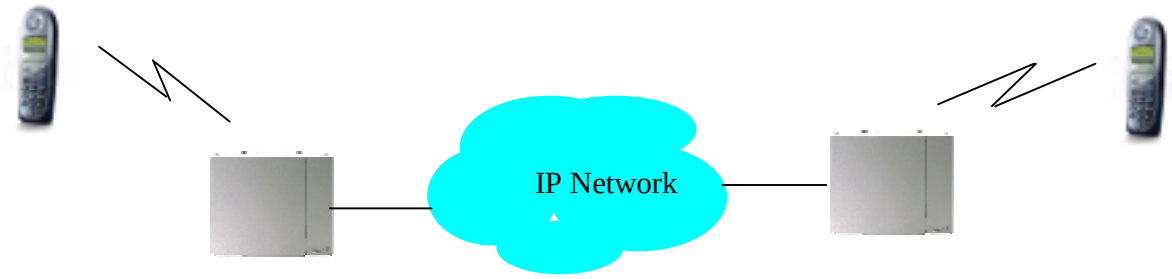

Figure 3: Wideband call via Internet

4.2.4 Calls between a New Generation DECT system and a digital phone supporting compatible codecs

This scenario is possible, at least in the following cases:

4.2.4.1 Via ISDN

ISDN digital phones with S/T interface and supporting the ITU-T Recommendation G.722 [17] codec could establish wideband calls with New Generation DECT equipment. Identical scenario is possible for PABX digital terminals calling or called by New Generation DECT systems.

4.2.4.2 Via IP network

Digital phones with a VoIP interface could also establish wideband communications with New Generation DECT equipment. This scenario includes both, dedicated VoIP phone devices and computers implementing the necessary software. Due to the evolution of computer industry, nearly all modern Personal Computers have the capability to become a wideband phone with DECT compatible codecs.

4.2.4.3 Internal PABX calls

PABX supporting New Generation DECT and digital extensions with compatible wideband codecs could also benefit from the wideband voice quality for their internal calls.

4.2.5 Legacy scenarios

Existing DECT GAP compliant equipment (both FT and PT) should be able to interoperate with New Generation DECT systems. In such cases, communication will be traditional 32 kbit/s ADPCM voice links.

Interoperability shall be possible in both directions:

- A new Generation DECT wideband speech PT should be interoperable with legacy DECT FTs.
- A legacy PT should be interoperable with new Generation DECT FTs.

5 Service and feature definitions

5.1 New Generation DECT Speech Services

For the purposes of the present document, the following definitions shall apply:

Narrow band ADPCM G.726 voice service (NG1.1): ITU-T Recommendation G.726 [15] narrow band codec **(NG1.SC.1)** over 32 kbit/s unprotected transmission channel

Narrow band PCM G.711 voice service (NG1.2): ITU-T Recommendation G.711 [16] narrow band codec **(NG1.SC.2)** over 64 kbit/s protected or unprotected transmission channels

Wideband 7 kHz G.722 voice service (NG1.3): ITU-T Recommendation G.722 [17] wideband codec **(NG1.SC.3)** over 64 kbit/s protected or unprotected transmission channels

Wideband 7 kHz low rate G.729.1 voice service (NG1.4): ITU-T Recommendation G.729.1 [18] wideband codec **(NG1.SC.4)** over 32 kbit/s unprotected transmission channels

Super wideband 14 kHz MPEG-4 ER AAC-LD voice service (NG1.5): MPEG-4 ER AAC-LD super wideband codec **(NG1.SC.5)** over 64 kbit/s protected or unprotected transmission channels

Wideband 11 kHz low rate MPEG-4 ER AAC-LD voice service (NG1.6): MPEG-4 ER AAC-LD super wideband codec **(NG1.SC.6)** over 32 kbit/s unprotected transmission channels

5.2 Network (NWK) features

For the purposes of the present document, all definitions of EN 300 444 [13] clause 4.1, plus the following shall apply:

Codec Negotiation (NG1.N.1): capability to negotiate the speech codec to be used in a communication, based on the supported capabilities in both peers and the previsions included in the present document. This feature may require slot type change

Codec Switching (NG1.N.2): capability to switch between different speech codecs during a call. This feature may require slot type change

5.3 Data Link Control (DLC) service definitions

For the purposes of the present document, all definitions of EN 300 444 [13] clause 5.1 plus the following shall apply:

LU1 Transparent UnProtected service (TRUP) Class 0/minimum_delay (NG1.D.1): transparent unprotected service introducing minimum delay , transmission Class 0/min_delay as defined by EN 300 175-4 [4] clause 11.2

LU1 Transparent UnProtected service (TRUP) Class 0 (NG1.D.2): transparent unprotected service introducing minimum delay , transmission Class 0 as defined by EN 300 175-4 [4] clause 11.2

LU7 64 kbit/s protected bearer service (NG1.D.3): protected service providing reliable 64 kbit/s transmission over packet type P80 incorporating FEC and ARQ protection mechanisms. Defined by EN 300 175-4 [4] clause 11.9

FU1 DLC frame (NG1.D.4): bidirectional frame used in LU1 service. Defined in EN 300 175-4 [4] clause 12.2. Frame length depends on slot type and is defined in table 12.2.1.1 of EN 300 175-4 [4] clause 12.2.1

FU7 DLC frame (NG1.D.5): bidirectional frame used in LU7 service. Defined in EN 300 175-4 [4] clause 11.9

5.4 Medium Access Control (MAC) service definitions

For the purposes of the present document, all definitions of EN 300 444 [13] clause 5.2 plus the following shall apply:

I_N_minimum delay symmetric MAC service type (NG1.M.1): I_N_minimum delay symmetric connection as defined in EN 300 175-3 [3] clause 5.6.2.1

I_N_normal delay symmetric MAC service type (NG1.M.2): I_N_normal delay symmetric connection as defined in EN 300 175-3 [3] clause 5.6.2.1

I_{PQ}_error_detection symmetric MAC service type (NG1.M.3): I_{PQ}_ error_detection symmetric connection as defined in EN 300 175-3 [3] clause 5.6.2.1. (type 3: I_P_ error_detection with single-subfield protected B-field as defined in [3] clause 6.2.1.3.4)

5.5 Physical Layer (PHL) service definitions

For the purposes of the present document the following definitions shall apply:

2 level GFSK modulation (NG1.P.1): 2 level Gaussian frequency Shift Key (GFSK) modulation as defined by EN 300 175-2 [2] clause 5

Physical Packet P32 (NG1.P.2): physical packet P32 (full slot) as defined by EN 300 175-2 [2] clause 4.4.2

Physical Packet P64 (NG1.P.3): variable capacity Physical packet P00j as defined by EN 300 175-2 [2] clause 4.4.3, with j = 640

Physical Packet P67 (NG1.P.4): variable capacity Physical packet P00j as defined by EN 300 175-2 [2] clause 4.4.3, with j = 672

Physical Packet P80 (NG1.P.5): physical packet P80 (double slot) as defined by EN 300 175-2 [2] clause 4.4.4

5.6 Speech coding service definitions

For the purposes of the present document the following definitions shall apply:

G.726 32 kbit/s ADPCM (NG1.SC.1): ITU-T Recommendation G.726 [15] narrow band codec 15 as defined by EN 300 175-8 [8] clause 5.1. ITU-T Recommendation G.726 [15] codec is mandatory for New Generation DECT in order to ensure interoperability with existing DECT systems

G.711 64 kbit/s log-PCM (NG1.SC.2): ITU-T Recommendation G.711 narrow band codec [16] as defined by EN 300 175-8 [8] clause 5.2. ITU-T Recommendation G.711 [16] codec is optional for New Generation DECT in order to improve the quality of narrow band communications, and fax/modem transmissions. ITU-T Recommendation G.711 [16] provides a slightly higher intrinsic voice quality and no transcoding for PSTN calls. Both, A-Law and μ-Law are supported

G.722 64 kbit/s wideband (NG1.SC.3): ITU-T Recommendation G.722 wideband SB-ADPCM 7 kHz codec [17] as defined by EN 300 175-8 [8] clause 5.3. ITU-T Recommendation G.722 [17] is chosen as mandatory wideband codec for New Generation DECT in order to greatly increase the voice quality by extending the bandwidth from narrow band to wideband. ITU-T Recommendation G.722 [17] provides a high wideband quality at a bit rate of 64 kbit/s with low complexity and very low delay

G.729.1 32 kbit/s wideband (NG1.SC.4): ITU-T Recommendation G.729.1 wideband codec [18] as defined by EN 300 175-8 [8] clause 5.4. ITU-T Recommendation G.729.1 [18] codec is optional for New Generation DECT in order to provide even higher wideband quality and better robustness to packets/frames losses than ITU-T Recommendation G.722 [17] at half the bit rate of ITU-T Recommendation G.722 [17]. This allows a better transport efficiency on the network side and over the DECT air interface (one full slot). In addition, it is seamless interoperable with largely deployed ITU-T Recommendation G.729 (see bibliography) based VoIP networks and terminals. ITU-T Recommendation G.729.1 [18] encodes signals in frames of 20 ms. It is a scalable codec operating at bitrates of 8 kbit/s and from 12 kbit/s to 32 kbit/s per steps of 2 kbit/s, in narrow band or in wideband from 14 kbit/s. ITU-T Recommendation G.729.1 [18] already incorporates a high efficiency packet loss concealment mechanism

MPEG-4 ER AAC-LD 64 kbit/s super wideband (NG1.SC.5): MPEG-4 ER AAC-LD codec [17] as defined by EN 300 175-8 [8] clause 5.5.1. MPEG-4 ER AAC-LD is optional for New Generation DECT in order to provide higher quality than G.722 by further extending the bandwidth to superwideband (50 Hz to 14 kHz) (and even further, up to full audio bandwidth (20 Hz to 20 kHz)). MPEG-4 ER AAC-LD is designed for high quality communication applications including all kind of audio signals e.g. speech and music and provides a high quality for music streaming or other multimedia applications mixing speech and music. It provides an audio bandwidth of 14 kHz or more at a bitrate of 64 kbit/s. MPEG 4 ER AAC-LD is standardized in ISO/IEC 14496-3 [20]. The frame size is 10 ms and the algorithmic delay 20 ms

MPEG-4 ER AAC-LD 32 kbit/s wideband (NG1.SC.6): as (NG1.SC5), but using the 32 kbit/s mode, as defined by EN 300 175-8 [8] clause 5.5.2. It provides a bandwidth of 11,5 kHz or more. The frame size is 20 ms and the algorithmic delay 40 ms

PLC (Packet Loss Concealment) G.722 Appendix III & IV (NG1.SC.7): to better cope with transmission errors, a Packet Loss Concealment algorithm (PLC) as defined by EN 300 175-8 [8] clause 5.3.2 may be optionally implemented for ITU-T Recommendation G.722 [17]. Appendices III and IV describe packet loss concealment solutions extending the ITU-T Recommendation G.722 [17] decoder. These PLC algorithms may be optionally implemented to improve voice quality in degraded transmission conditions where packets/frames may be lost (in IP network or on DECT air interface)

NOTE 1: Both appendices meet the same quality requirements but address two different quality/complexity trade offs:

1) Appendix III aims at maximizing the robustness at a price of additional complexity.

2) Appendix IV proposes an optimized complexity/quality trade off with almost no additional complexity compared with ITU-T Recommendation G.722 [17] normal decoding (0,07 WMOPS).

Since ITU-T Recommendation G.722 [17] does not incorporate any mechanism to cope with lost frames/packets, the use of a PLC algorithm is strongly recommended to avoid annoying effects in case of packet/frame losses.

NOTE 2: ITU-T Recommendation G.729.1 [18] already incorporates a packet loss concealment mechanism.

Detection of Modem/fax tone (NG1.SC.8): detection of the 1 100 Hz, 1 300 Hz and 2 100 Hz standard tones indicating a fax/modem transmission and answering, as defined by EN 300 175-8 [8] clause 5.2.2. The main utility of this function is the switching of codecs to transparent PCM (ITU-T Recommendation G.711 [16]) in order to facilitate modem/fax transmission. The tone detection can also be used to de-activate echo suppression if present

Codec selection and switching (NG1.SC.9): to handle several codecs (at least ITU-T Recommendation G.726 [15] and ITU-T Recommendation G.722 [17]), New Generation DECT will support a codec selection and switching mechanism. This may consequently allow the use of other codecs that could be specified in next releases as additional optional codecs according to future application or interoperability needs

5.7 Application features

For the purposes of the present document, all definitions of EN 300 444 [13] clause 4.2 shall apply:

6 Inter-operability requirements

6.1 General

The tables listed in this clause define the status of all protocol elements (i.e. features, services, and procedures) which can be: mandatory, optional, conditional under the provision of another protocol element, outside the scope of the present document, or not applicable. The status is identified by the status column designations defined in clause 3.2, and is described separately for FT and PT. In the case of FT, the status can be different for products intended for the Residential/Business (R/B) market or for the Public market segment.

All optional elements shall be process mandatory according to the procedures described in the present document.

Protocol elements defined as mandatory, optional or conditional in this clause are further defined in the referenced DECT specification, or, if needed, in clause 7 of the present document.

New Generation DECT wideband speech is defined as a backcompatible enhancement of DECT Generic Access Profile (GAP) [13]. All procedures not specific of the New Generation DECT, are referenced to their original description in EN 300 444 (GAP) [13].

NOTE: Annexes A and D are informative and may be used as additional information, but do not mandate requirements. Annexes B, C, E, and F are normative.

The requirements of EN 301 406 [10] and EN 300 176-2 [9] shall be met by all equipment conforming to the present document.

6.2 New Generation DECT Speech Services support status

The following end-user services shall be supported by New Generation DECT wideband voice specification:

Table 1: Speech service status

Feature supported					
			Status		
Item no.	Name of Service	Reference	PT	FT	
				R/B	P
NG1.1	Narrow band ADPCM G.726 32 kbit/s voice service	5.1	M	M	M
NG1.2	Narrow band PCM G.711 64 kbit/s voice service	5.1	O	O	O
NG1.3	Wideband G.722 64 kbit/s voice service	5.1	M	M	M
NG1.4	Wideband G.729.1 32 kbit/s voice service	5.1	O	O	O
NG1.5	MPEG-4 ER AAC-LD super wideband 64 kbit/s voice service	5.1	O	O	O
NG1.6	MPEG-4 ER AAC-LD wideband 32 kbit/s voice service	5.1	O	O	O

6.3 Services to DECT feature implementation mappings

New Generation DECT services shall be implemented using the following DECT services and features, according to the following implementation alternatives:

Table 2: Speech service to DECT features implementation mappings

Service/DECT Feature mapping						
				Status		
Service	IA	DECT feature/service	Reference	PT	FT	
					R/B	P
NG1.1 Narrow band ADPCM G.726 32 kbit/s voice service	**I**		**5.1**	**M**	**M**	**M**
		NG1.P.1 2 level GFSK modulation	5.5	M	M	M
		NG1.P.2 Physical Packet P32	5.5	M	M	M
		NG1.M.1 I_N_minimum delay symmetric MAC service type	5.4	M	M	M
		NG1.D.1 DLC Service LU1 TRUP Class 0/min_delay	5.3	M	M	M
		NG1.D.4 DLC frame FU1	5.3	M	M	M
		NG1.SC.1 ITU-T Recommendation G.726 [15] 32 kbit/s ADPCM codec	5.6	M	M	M
NG1.2 Narrow band PCM G.711 64 kbit/s voice service	**I**		**5.1**	**O**	**O**	**O**
		NG1.P.1 2 level GFSK modulation	5.5	M	M	M
		NG1.P.3 Physical Packet P64	5.5	M	M	M
		NG1.M.1 I_N_minimum delay symmetric MAC service type	5.4	M	M	M
		NG1.D.1 DLC Service LU1 TRUP Class 0/min_delay	5.3	M	M	M
		NG1.D.4 DLC frame FU1	5.3	M	M	M

Service/DECT Feature mapping						
				Status		
Service	**IA**	**DECT feature/service**	**Reference**	**PT**	**FT**	
					R/B	**P**
		NG1.SC.2 ITU-T Recommendation G.711 [16] 64 kbit/s PCM codec	5.6	M	M	M
		NG1.SC.8 Detection of Fax/modem tone	5.6	O	O	O
		NG1.SC.9 Codec selection and switching	5.6	M	M	M
NG1.2 Narrow band PCM G.711 64 kbit/s voice service	**II**		**5.1**	**O**	**O**	**O**
		NG1.P.1 2 level GFSK modulation	5.5	M	M	M
		NG1.P.4 Physical Packet P67	5.5	M	M	M
		NG1.M.3 I_{PQ}_error_detection symmetric MAC service type	5.4	M	M	M
		NG1.D.1 DLC Service LU1 TRUP Class 0/min_delay	5.3	M	M	M
		NG1.D.4 DLC frame FU1	5.3	M	M	M
		NG1.SC.2 ITU-T Recommendation G.711 [16] 64 kbit/s PCM codec	5.6	M	M	M
		NG1.SC.8 Detection of Fax/modem tone	5.6	O	O	O
		NG1.SC.9 Codec selection and switching	5.6	M	M	M
NG1.2 Narrow band PCM G.711 64 kbit/s voice service	**III**		**5.1**	**O**	**O**	**O**
		NG1.P.1 2 level GFSK modulation	5.5	M	M	M
		NG1.P.5 Physical Packet P80	5.5	M	M	M
		NG1.M.2 I_N_normal_delay symmetric MAC service type	5.4	M	M	M
		NG1.D.3 DLC Service LU7 64 kbit/s protected bearer	5.3	M	M	M
		NG1.D.5 DLC frame FU7	5.3	M	M	M
		NG1.SC.2 ITU-T Recommendation G.711 [16] 64 kbit/s PCM codec	5.6	M	M	M
		NG1.SC.8 Detection of Fax/modem tone	5.6	O	O	O
		NG1.SC.9 Codec selection and switching	5.6	M	M	M
NG1.3 Wideband 7 kHz G.722 64 kbit/s voice service	**I**		**5.1**	**M**	**M**	**M**
		NG1.P.1 2 level GFSK modulation	5.5	M	M	M
		NG1.P.3 Physical Packet P64	5.5	M	M	M
		NG1.M.1 I_N_minimum delay symmetric MAC service type	5.4	M	M	M
		NG1.D.1 DLC Service LU1 TRUP Class 0/min_delay	5.3	M	M	M
		NG1.D.4 DLC frame FU1	5.3	M	M	M
		NG1.SC.3 ITU-T Recommendation G.722 [17] 64 kbit/s 7 kHz wideband codec	5.6	M	M	M
		NG1.SC.7 Packet loss Concealment (PLC) for G.722	5.6	O	O	O
		NG1.SC.9 Codec selection and switching	5.6	M	M	M
NG1.3 Wideband 7 kHz G.722 64 kbit/s voice service	**II**		**5.1**	**O**	**O**	**O**
		NG1.P.1 2 level GFSK modulation	5.5	M	M	M
		NG1.P.3 Physical Packet P67	5.5	M	M	M
		NG1.M.3 I_{PQ}_error_detection symmetric MAC service type	5.4	M	M	M
		NG1.D.1 DLC Service LU1 TRUP Class 0/min_delay	5.3	M	M	M
		NG1.D.4 DLC frame FU1	5.3	M	M	M

Service/DECT Feature mapping						
				Status		
Service	**IA**	**DECT feature/service**	**Reference**	**PT**	**FT**	
					R/B	**P**
		NG1.SC.3 ITU-T Recommendation G.722 [17] 64 kbit/s 7 kHz wideband codec	5.6	M	M	M
		NG1.SC.7 Packet loss Concealment (PLC) for ITU-T Recommendation G.722 [17]	5.6	O	O	O
		NG1.SC.9 Codec selection and switching	5.6	M	M	M
NG1.4 Wideband 7 kHz G.729.1 32 kbit/s voice service	**I**		**5.1**	**O**	**O**	**O**
		NG1.P.1 2 level GFSK modulation	5.5	M	M	M
		NG1.P.3 Physical Packet P32	5.5	M	M	M
		NG1.M.2 I_N_normal_delay symmetric MAC service type	5.4	M	M	M
		NG1.D.2 DLC Service LU1 TRUP Class 0	5.3	M	M	M
		NG1.D.4 DLC frame FU1	5.3	M	M	M
		NG1.SC.4 ITU-T Recommendation G.729.1 [18] 32 kbit/s 7 kHz wideband codec	5.6	M	M	M
		NG1.SC.9 Codec selection and switching	5.6	M	M	M
NG1.5 Superwideband 14 kHz MPEG-4 ER AAC-LD 64 kbit/s voice service	**I**		**5.1**	**O**	**O**	**O**
		NG1.P.1 2 level GFSK modulation	5.5	M	M	M
		NG1.P.3 Physical Packet P64	5.5	M	M	M
		NG1.M.2 I_N_normal_delay symmetric MAC service type	5.4	M	M	M
		NG1.D.2 DLC Service LU1 TRUP Class 0	5.3	M	M	M
		NG1.D.4 DLC frame FU1	5.3	M	M	M
		NG1.SC.5 MPEG4 AAC-LD 64 kbit/s 14 kHz superwideband codec	5.6	M	M	M
		NG1.SC.9 Codec selection and switching	5.6	M	M	M
NG1.5 Superwideband 14 kHz MPEG-4 ER AAC-LD 64 kbit/s voice service	**II**		**5.1**	**O**	**O**	**O**
		NG1.P.1 2 level GFSK modulation	5.5	M	M	M
		NG1.P.3 Physical Packet P67	5.5	M	M	M
		NG1.M.3 I_{PQ}_error_detection symmetric MAC service type	5.4	M	M	M
		NG1.D.1 DLC Service LU1 TRUP Class 0/min_delay	5.3	M	M	M
		NG1.D.4 DLC frame FU1	5.3	M	M	M
		NG1.SC.5 MPEG4 AAC-LD 64 kbit/s 14 kHz superwideband codec	5.6	M	M	M
		NG1.SC.9 Codec selection and switching	5.6	M	M	M
NG1.6 Wideband 11 kHz MPEG-4 ER AAC-LD 32 kbit/s voice service	**I**		**5.1**	**O**	**O**	**O**
		NG1.P.1 2 level GFSK modulation	5.5	M	M	M
		NG1.P.3 Physical Packet P32	5.5	M	M	M
		NG1.M.2 I_N_normal_delay symmetric MAC service type	5.4	M	M	M
		NG1.D.2 DLC Service LU1 TRUP Class 0	5.3	M	M	M
		NG1.D.4 DLC frame FU1	5.3	M	M	M
		NG1.SC.6 MPEG4 AAC-LD 32 kbit/s 11 kHz wideband codec	5.6	M	M	M
		NG1.SC.9 Codec selection and switching	5.6	M	M	M
IA = Implementation Alternative						

6.4 NWK features

New Generation DECT wideband speech devices shall support the following Network layer features:

Table 3: NWK features status

Feature supported					
			Status		
Item no.	Name of feature	Reference	PT	FT	
				R/B	P
NG1.N.1	Codec Negotiation	5.2	M	M	M
NG1.N.2	Codec Switching	5.2	M	M	M
GAP.N.1	Outgoing call	4.1 [13]	M	M	M
GAP.N.2	Off hook	4.1 [13]	M	M	M
GAP.N.3	On hook (full release)	4.1 [13]	M	M	M
GAP.N.4	Dialled digits (basic)	4.1 [13]	M	M	M
GAP.N.5	Register recall (note 4 and note 5)	4.1 [13]	M	O	O
GAP.N.6	Go to DTMF signalling (defined tone length) (note 1)	4.1 [13]	M	O	M
GAP.N.7	Pause (dialling pause) (note 3)	4.1 [13]	M	O	O
GAP.N.8	Incoming call	4.1 [13]	M	M	M
GAP.N.9	Authentication of PP	4.1 [13]	M	O	M
GAP.N10	Authentication of user (note 2)	4.1 [13]	M	O	O
GAP.N.11	Location registration	4.1 [13]	M	O	M
GAP.N.12	On air key allocation (note 2)	4.1 [13]	M	O	O
GAP.N.13	Identification of PP	4.1 [13]	M	O	O
GAP.N.14	Service class indication/assignment	4.1 [13]	M	O	M
GAP.N.15	Alerting	4.1 [13]	M	M	M
GAP.N.16	ZAP (note 2)	4.1 [13]	M	O	O
GAP.N.17	Encryption activation FT initiated	4.1 [13]	M	O	M
GAP.N.18	Subscription registration procedure on-air	4.1 [13]	M	M	M
GAP.N.19	Link control	4.1 [13]	M	M	M
GAP.N.20	Terminate access rights FT initiated (note 2)	4.1 [13]	M	O	O
GAP.N.21	Partial release	4.1 [13]	O	O	O
GAP.N.22	Go to DTMF (infinite tone length)	4.1 [13]	O	O	O
GAP.N.23	Go to Pulse	4.1 [13]	O	O	O
GAP.N.24	Signalling of display characters	4.1 [13]	O	O	O
GAP.N.25	Display control characters	4.1 [13]	O	O	O
GAP.N.26	Authentication of FT	4.1 [13]	O	O	O
GAP.N.27	Encryption activation PT initiated	4.1 [13]	O	O	O
GAP.N.28	Encryption deactivation FT initiated	4.1 [13]	O	O	O
GAP.N.29	Encryption deactivation PT initiated	4.1 [13]	O	O	O
GAP.N.30	Calling Line Identification Presentation (CLIP)	4.1 [13]	M	M	M
GAP.N.31	Internal call	4.1 [13]	O	O	O
GAP.N.32	Service call	4.1 [13]	O	O	O
GAP.N.33	Enhanced U- plane connection	4.1 [13]	O	O	O
GAP.N.34	Calling Name Identification Presentation (CNIP)	4.1 [13], F.1.2.1	O	O	O

NOTE 1: The PT is only required to be able to send the <<MULTI-KEYPAD>> information element containing the DECT standard 8-bit character (EN 300 175-5 [5], annex D) codings "Go to DTMF", defined tone length and the FT is required to be able to understand it in the public environment.
NOTE 2: This feature is required to be supported in the PT to guarantee the same level of security among all the handsets that operates in a system. The invocation of the feature is however optional to the operator.
NOTE 3: The PT is required to be able to send the <<MULTI-KEYPAD>> information element containing the DECT standard 8-bit character (EN 300 175-5 [5], annex D) codings "Dialling Pause". This guarantees automatic access to secondary or alternative networks.
NOTE 4: This feature uses keypad code 15 hex.
NOTE 5: The FT is not mandated to receive and understand the register recall DECT character. However, if a FT supports it there may be no corresponding action that the FT can take with the local network as a result of this function.

6.5 Data Link Control (DLC) services

New Generation DECT wideband speech devices shall support the following DLC services:

Table 4: DLC services status

Service supported					
			Status		
Item no.	Name of service	Reference	PT	FT	
				R/B	P
NG1.D.1	LU1 Transparent UnProtected service (TRUP) Class 0 /minimum_delay	5.3	M	M	M
NG1.D.2	LU1 Transparent UnProtected service (TRUP) Class 0	5.3	C401	C401	C401
NG1.D.3	LU7 64 kbit/s protected bearer service	5.3	C401	C401	C401
NG1.D.4	FU1 DLC frame	5.3	M	M	M
NG1.D.5	FU7 DLC frame	5.3	C401	C401	C401
GAP.D.1	LAPC class A service and Lc	5.1 [13]	M	M	M
GAP.D.2	C_S channel fragmentation and recombination	5.1 [13]	M	M	M
GAP.D.3	Broadcast Lb service	5.1 [13]	M	M	M
GAP.D.4	Intra-cell voluntary connection handover	5.1 [13]	M	C402	C402
GAP.D.5	Intercell voluntary connection handover (note)	5.1 [13]	M	O	O
GAP.D.6	Encryption activation	5.1 [13]	M	C404	M
GAP.D.7	LU1 TRUP Class 0/min_delay	5.1 [13]	M	M	M
GAP.D.8	FU1	5.1 [13]	M	M	M
GAP.D.9	Encryption deactivation	5.1 [13]	C403	C403	C403
NOTE: The PT is required to be able to support handover between RFPs. The invocation of the feature is however optional to the operator.					
C401: Status defined by clause 6.3, table 2. C402: IF service GAP.M.9 THEN O ELSE M. C403: IF feature GAP.N.29 OR N.28 THEN M ELSE I. C404: IF feature GAP.N.17 OR N.27 THEN M ELSE I.					

6.6 Medium Access Control (MAC) services

New Generation DECT wideband speech devices shall support the following MAC layer services:

Table 5: MAC services status

Service supported					
			Status		
Item no.	Name of service	Reference	PT	FT	
				R/B	P
NG1.M.1	I_N_minimum delay symmetric MAC service type	5.4	M	M	M
NG1.M.2	I_N_normal delay symmetric MAC service type	5.4	C501	C501	C501
NG1.M.3	I_{PQ}_error_detection symmetric MAC service type	5.4	C501	C501	C501
GAP.M.1	General	5.2 [13]	M	M	M
GAP.M.2	Continuous broadcast	5.2 [[13]	M	M	M
GAP.M.3	Paging broadcast	5.2 [13]	M	M	M
GAP.M.4	Basic connections	5.2 [13]	M	M	M
GAP.M.5	C_S higher layer signalling	5.2 [13]	M	M	M
GAP.M.6	Quality control	5.2 [13]	M	M	M
GAP.M.7	Encryption activation	5.2 [13]	M	C505	M
GAP.M.8	Extended frequency allocation (note)	5.2 [13]	M	O	O
GAP.M.9	Bearer Handover, intra-cell	5.2 [13]	M	C502	C502
GAP.M.10	Bearer Handover, inter-cell	5.2 [13]	M	O	O
GAP.M.11	Connection Handover, intra-cell	5.2 [13]	M	C503	C503
GAP.M.12	Connection Handover, inter-cell	5.2 [13]	M	O	O
GAP.M.13	SARI support	5.2 [13]	M	O	O
GAP.M.14	Encryption deactivation	5.2 [13]	C504	C504	C504
NOTE: Handsets not supporting these extra frequencies need only adapt scanning to allow continued use of the standard DECT frequencies.					
C501: Status defined by clause 6.3, table 2. C502: IF service GAP.M.11 THEN O ELSE M. C503: IF service GAP.M.9 THEN O ELSE M. C504: IF feature GAP.N.29 OR N.28 THEN M ELSE I. C505: IF feature GAP.N.17 OR N.27 THEN M ELSE I.					

6.7 Physical layer (PHL) services

New Generation DECT wideband speech devices shall support the following Physical layer (PHL) services:

Table 6: PHL services status

Service supported					
			Status		
Item no.	Name of service	Reference	PT	FT	
				R/B	P
NG1.P.1	2 level GFSK modulation	5.5	M	M	M
NG1.P.2	Physical Packet P32	5.5	M	M	M
NG1.P.3	Physical Packet P64	5.5	M	M	M
NG1.P.4	Physical Packet P67	5.5	O	O	O
NG1.P.5	Physical Packet P80	5.5	O	O	O

The requirements of EN 300 444 [13] clause 11 also apply.

6.8 Speech codecs

New Generation DECT wideband speech devices shall support the following Speech codecs and related services:

Table 7: Speech Codecs

Service supported					
			Status		
Item no.	Name of service	Reference	PT	FT	
				R/B	P
NG1.SC.1	G.726 32 kbit/s ADPCM codec	5.6	M	M	M
NG1.SC.2	G.711 64 kbit/s PCM codec	5.6	C701	C701	C701
NG1.SC.3	G.722 64 kbit/s 7 kHz wideband codec	5.6	M	M	M
NG1.SC.4	G.729.1 32 kbit/s 7 kHz wideband codec	5.6	C701	C701	C701
NG1.SC.5	MPEG4 AAC-LD 64 kbit/s 14 kHz superwideband codec	5.6	C701	C701	C701
NG1.SC.6	MPEG4 AAC-LD 32 kbit/s 11 kHz wideband codec	5.6	C701	C701	C701
NG1.SC.7	Packet loss Concealment (PLC) for G.722]	5.6	C701	C701	C701
NG1.SC.8	Detection of Fax/modem tone	5.6	C701	C701	C701
NG1.SC.9	Codec selection and switching	5.6	M	M	M
C701: Status defined by clause 6.3, table 2.					

6.9 Application features

New Generation DECT wideband speech devices shall support the following Application features:

Table 8: Application features status

Feature supported					
			Status		
Item no.	Name of feature	Reference	PT	FT	
				R/B	P
GAP.A.1	AC_bitstring_mapping	4.2 [13]	M	C801	M
GAP.A.2	Multiple subscription registration	4.2 [13]	M	N/A	N/A
GAP.A.3	Manual entry of the PARK	4.2 [13]	O	N/A	N/A
GAP.A.4	Terminal identity number assignment in mono cell system	4.2 [13], F.1.4.1	O	O	N/A
C801: IF feature GAP.N.9 OR GAP.N.10 OR N.12 OR N.26 THEN M ELSE N/A.					

6.10 Network (NWK) feature to procedure mapping

The NWK features to procedure mapping of EN 300 444 (GAP) [13], clause 6.7 apply with the following changes and additional features:

Table 9: NWK feature to procedure mapping

Feature/Procedure mapping					
			Status		
Feature	**Procedure**	**Reference**	**PT**	**FT**	
				R/B	**P**
NG1.N.1 Codec Negotiation		5.2	M	M	M
	Exchange of codec list during registration and location registration	7.2.1	M	M	M
	Basic service wideband speech and default attributes	7.2.2	M	M	M
	Codec Negotiation during call establishment	7.2.3	M	M	M
NG1.N.2 Codec Switching		5.2	M	M	M
	Codec Change	7.2.4	M	M	M
	Slot type modification	7.2.5	M	M	M
	MAC layer advanced connection: service type or slot type modification	10.3.2 [3]			
	MAC layer connection type modification	10.3.3 [3]	M	M	M
GAP.N.31 Internal Call		4.1 [13]	O	O	O
	Internal call setup	7.3.4	M	M	M
	Internal call keypad	8.19 [13]	M	O	O
	Internal call CLIP	F.1.3.2	O	O	O
	Internal call CNIP	F.1.3.3	O	O	O

6.11 Data Link Control (DLC) Service to procedure mapping

The DLC service to procedure mapping of EN 300 444 (GAP) [13], clause 6.8.1 apply with the following changes and additional services:

Table 10: DLC service to procedure mapping

Service/Procedure mapping					
			Status		
Service	**Procedure**	**Reference**	**PT**	**FT**	
				R/B	**P**
NG1.D.1 LU1 Transparent UnProtected service (TRUP) Class 0/minimum_delay		5.3	M	M	M
	LU1 Transparent UnProtected service (TRUP) operation	11.2 [4]	M	M	M
	Class 0: No Lu_x retransmission or sequencing	14.2.3.1 [4]	M	M	M
	Class 0 procedures	14.3.2 [4]	M	M	M
	Minimum delay (speech) operation	14.2.3 [4]	M	M	M
	LLME U-plane establishment	9.9.1 [13]	M	M	M
NG1.D.2 LU1 Transparent UnProtected service (TRUP) Class 0		5.3	O	O	O
	LU1 Transparent UnProtected service (TRUP) operation	11.2 [4]	M	M	M
	Class 0: No Lu_x retransmission or sequencing	14.2.3.1 [4]	M	M	M
	Class 0 procedures	14.3.2 [4]	M	M	M
	LLME U-plane establishment	9.9.1 [13]	M	M	M
NG1.D.3 LU7 64 kbit/s protected bearer service			O	O	O
	LU7 DLC layer service	11.9.4 [4]	M	M	M
NG1.D.4 FU1 DLC frame			O	O	O
	FU1 frame operation	7.9.1	M	M	M
	FU1 frame structure	12.2 [4]	M	M	M
NG1.D.5 FU7 DLC frame			O	O	O
	FU7 frame structure	11.9.4.2 [4]	M	M	M

6.12 Medium Access Control (MAC) service to procedure mapping

The MAC service to procedure mapping of EN 300 444 (GAP) [13], clause 6.8.2 apply with the following changes and additional services:

Table 11: MAC service to procedure mapping

Service/Procedure mapping					
			Status		
Service	**Procedure**	**Reference**	**PT**	**FT**	
				R/B	**P**
NG1M.1 I_N_minimum delay symmetric MAC service type		5.4	M	M	M
	MAC layer procedures: general	7.9.1	M	M	M
	MAC Connection oriented service	5.6 [3]	M	M	M
	MAC Basic connection	5.6.1.1 [3]	M	M	M
	MAC Advanced connection	5.6.1.2 [3]	M	M	M
	I_N_minimum delay symmetric MAC service, type 1	5.6.2.1 [3]	M	M	M
NG1.M.2 I_N_normal delay symmetric MAC service type			O	O	O
	MAC layer procedures: general	7.9.1	M	M	M
	MAC Connection oriented service	5.6 [3]	M	M	M
	MAC Basic connection	5.6.1.1 [3]	M	M	M
	MAC Advanced connection	5.6.1.2 [3]	M	M	M
	I_N_normal delay symmetric MAC service type 2	5.6.2.1 [3]	M	M	M
NG1.M.3 I_{PQ}_error_detection symmetric MAC service type			O	O	O
	MAC layer procedures: general	7.9.1	M	M	M
	MAC Connection oriented service	5.6 [3]	M	M	M
	MAC Basic connection	5.6.1.1 [3]	M	M	M
	MAC Advanced connection	5.6.1.2 [3]	M	M	M
	I_P_error_detection symmetric MAC service type 3	5.6.2.1 [3]	M	M	M
	Single-subfield protected format	6.2.1.3.4 [3]	M	M	M

6.13 Application feature to procedure mapping

The Application feature to procedure mapping of EN 300 444 (GAP) [13], clause 6.8.3 shall apply.

6.14 General requirements

6.14.1 Network (NWK) layer message contents

All reserved single bits shall be set to 0.

6.14.2 Transaction identifier

The transaction identifier value for a CC call shall always get assigned the lowest available free number.

6.14.3 Length of a Network (NWK) layer message

PP and the FP shall be capable of receiving and processing NWK layer messages of at least 63 octets long. All mandatory information elements as defined in the present document shall be included in the first 63 octets.

This requires only one DLC segment to be supported as mandatory. The DLC shall convey the first segment of a layer 3 message to the NWK layer. Additional segments of a layer 3 message may be discarded by the receiving side, (see clause 9.2.3 of EN 300 444 [13]).

6.14.4 Handling of error and exception conditions

If a MM message, requesting initiation of a MM procedure, is received in a CC state where the receiving entity is not required to support it and does not support it, this message shall be ignored.

Whenever an unexpected CC message, except {CC-RELEASE} or {CC-RELEASE-COM}, or an unrecognized message is received in any CC state, the message shall be ignored.

When a message other than {CC-SETUP}, {CC-RELEASE} or {CC-RELEASE-COM} is received which has one or more mandatory information elements missing or with invalid content, the normal release procedure as described in clause 8.7 shall be invoked.

EN 300 175-5 [5], clause 17.6.4 shall also apply to mandatory information elements in MM messages with a length exceeding the allowed maximum value.

The usage of a reserved value in an information element field shall not by itself constitute an error. The receiver of such a value shall process the value if it understands it or shall ignore it otherwise.

In all other cases the rules and order of precedence specified in EN 300 175-5 [5], clause 17, shall be obeyed.

6.14.5 Generic Access Profile (GAP) default setup attributes

The <<IWU-ATTRIBUTES>> and <<CALL-ATTRIBUTES>> information elements are not required to be understood by a "GAP" equipment. The values, as stated in EN 300 175-5 [5], annex E shall be considered as default. The value "1" of the field <Network layer attributes> in <<CALL-ATTRIBUTES>> shall be interpreted as indicating "Generic Access Profile (GAP)".

6.14.6 Coexistence of Mobility Management (MM) and Call Control (CC) procedures

Table 12 below describes whether a MM procedure is supported in any CC state or whether a restriction applies. The restriction has been made in order to limit the complexity of the receiving side so that it is not mandated to understand MM messages in all CC states for the purpose of achieving inter-operability.

Table 12: Support of MM procedures in CC states

Procedure	Mandatory support in CC state
Identification of PT	All states
Authentication of FT	All states
Authentication of PT	All states
Authentication of user	All states
Location registration	All states
Location update	All states
Obtaining access rights	T(F)-00
FT terminating access rights	F(T)-00, T-01, T-10
Key allocation	F(T)-00
Cipher-switching initiated by FT	All states
Cipher switching initiated by PT	All states

The CC and MM entities may work independently one from the other. If a FT decides to perform a MM procedure prior to proceeding with a PT initiated CC procedure, the FT has the rights to restart the CC timers in the PT to prevent the CC state machine from waiting on a response delayed because of the MM procedure execution. For this purpose the FT may send a {CC-NOTIFY} message. The support of this message is mandatory for the PT and optional for the FT. The {CC-NOTIFY} shall include the <<TIMER-RESTART>> information element.

6.14.7 Coding rules for information elements

For mandatory information elements, at least the first octet within any octet group shall be present. It is not permitted to use the information element field <Length of Contents> to omit an octet group. However, if explicitly stated a mandatory information element may contain zero length contents.

7 Procedure description

The following clauses define the process mandatory procedures which are in the scope of the New Generation DECT wideband speech. Each procedure (if appropriate) is divided into three parts:

a) normal (i.e. successful) case(s). This part defines the functions and respective protocol element values in normal operation;

b) associated procedure(s). This is an integral part of the actual procedure (if defined in the present document), i.e. if a procedure is being declared to be supported, the respective entity shall also support the associated procedures, e.g. timer management, in the clause following the description of the normal case;

c) exceptional case(s). This is an integral part of the actual procedure (if defined in the present document), i.e. if a procedure is being declared to be supported, the respective entity shall also support the exception handling defined in the clause following the description of the normal case.

All protocol elements listed in the following clauses are process mandatory, i.e. the FT and PT depending on their role in the procedure shall send or shall receive and process the relevant protocol elements as listed in the respective tables if not explicitly stated as being optional.

The primitives used in procedure descriptions are defined only for the purpose of describing layer-to-layer interactions. The primitives are defined as an abstract list of parameters, and their concrete realization may vary between implementations. No formal testing of primitives is intended. The primitive definitions have no normative significance.

7.1 Generic Access Profile (GAP) Backward compatibility with DECT standard

7.1.1 New Generation DECT Fixed Part (FP) requirement

The FP shall support the GAP standard procedures (full slot and ITU-T Recommendation G.726 [15]). In other words, it shall inter-operate with a GAP compliant PP.

7.1.2 New Generation DECT Portable Part (PP) registered on standard FP

The PP shall use the GAP standard procedures (full slot and ITU-T Recommendation G.726 [15]) in front of standard FP.

7.2 Generic Access Profile (GAP) procedures

Unless otherwise noted, all procedures defined in GAP [13] are automatically applicable to New Generation DECT wideband speech. Therefore New Generation DECT wideband speech can be considered an extension of GAP.

The following clauses describe the additional procedures specific for New Generation DECT wideband speech.

7.3 Network (NWK) layer procedures

This clause specifies the additional NWK layer procedures, messages and information elements required in New Generation DECT wideband speech not described in GAP [13], or incorporating modifications to the GAP specification.

This profile does not prevent any PT or FT from transmitting or receiving and processing any other NWK layer message or information element not specified in the profile. A PT or FT receiving an unsupported NWK layer message or information element, which it does not recognize, shall ignore it, as specified in EN 300 175-5 [5], clause 17.

7.3.1 Exchange of codec list during registration and location registration

Equipment supporting New Generation DECT wideband speech shall add the IE << CODEC-LIST>> indicating the supported codecs in the following messages:

{ACCESS-RIGHTS-REQUEST}, {ACCESS-RIGHTS-ACCEPT}

{LOCATE-REQUEST}, {LOCATE-ACCEPT}

The IE << CODEC-LIST>> shall contain at least ITU-T Recommendation G.722 [17] and ITU-T Recommendation G.726 [15] codecs.

7.3.2 Basic service wideband speech and default attributes

The attribute "wideband speech default" in Information Element <<Basic Service>> indicates that the default setup attributes for wideband speech shall be valid as indicated in annex E.2 of EN 300 175-5 [5], and that the mechanism for codec negotiation as described in clause 7.2.3 of the present document are valid.

7.3.3 Codec Negotiation during call establishment

Equipment supporting New Generation DECT wideband voice shall support the codec negotiation as described in the following.

A CC-SETUP that offers more codecs than ITU-T Recommendation G.726 [15] shall contain the basic service "wideband speech default setup attributes", instead of the basic service "basic speech default setup attributes".

The IE <<CODEC-LIST>> may be added in CC-SETUP if a new list of codec is needed on a call by call basis. This may be useful when requesting a new codec (codec different from the location/registration phase) or changing the priorities within the list of codecs. The IE << CODEC-LIST>> shall contain at least ITU-T Recommendation G.722 [17] and ITU-T Recommendation G.726 [15] codecs.

Sending the IE <<CODEC-LIST>> in CC-SETUP is not necessary in case the most recent list sent during registration/location registration is still the valid one.

The receiving side chooses the codec in a response message, which does not have to be the first response message. The codec has at latest to be chosen in a CC-INFO that connects the U-Plane using the IE <<PROGRESS-INDICATOR>> or with CC-CONNECT. The response message which chooses the codec uses the same IE <<CODEC-LIST>>, but only one codec shall be in the list.

Setup parameters which are not mentioned in the IE <<CODEC-LIST>> shall have default values as given in EN 300 175-5 [5] of this document. The IEs <<IWU-attributes>>, <<CALL-attributes>> and <<CONNECTION-attributes>> shall not be used in the CC-SETUP or in the response message.

After a codec was chosen, the call initiating side initiates a slot type modification at MAC layer if necessary.

In the case where the slot type modification is necessary and fails, the initiating side shall switch to a mandatory codec supporting the current slot format and indicate so by sending {IWU-INFO} including the IE <<CODEC-LIST>> with the required codec. On receiving this message, the receiving side shall also switch back to the required codec and indicate so by sending {IWU-INFO} including the IE <<CODEC-LIST>> with the required codec.

In the case where no slot type modification is necessary or the slot type modification is successful, {IWU-INFO} messages are not exchanged.

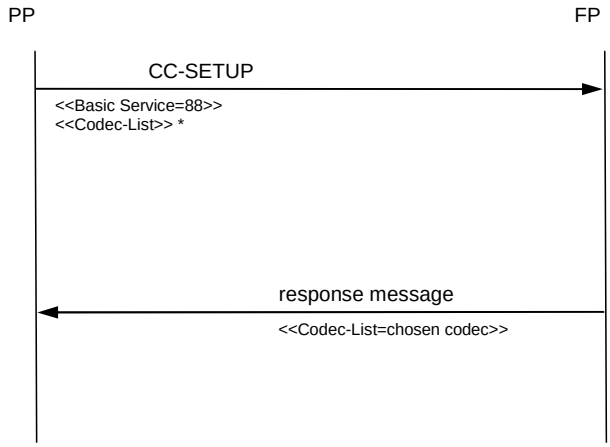

Figure 4: Codec Negotiation during call setup

7.3.4 Codec Change

Equipment supporting New Generation DECT wideband voice shall support the codec change as described in the following during call establishment after the codec negotiation is finished and in CC-state ACTIVE.

To switch the codec the initiating side sends a CC-SERVICE-CHANGE including the IE<<CODEC-LIST>> and the IE<<SERVICE-CHANGE-INFO>>.

The IE <<CODEC-LIST>> shall contain only one codec.

The IE <<SERVICE-CHANGE-INFO>> shall indicate that an audio codec change is attempted and that the sending side is master of the change.

The receiving side shall either accept or reject the change.

Both CC-SERVICE-ACCEPT and CC-SERVICE-REJECT shall not contain the IE <<CODEC-LIST>>.

In case the change is accepted, the initiator of the service change also initiates a slot type modification at MAC layer if necessary.

Having switched to the new codec and performed slot type modification if necessary, both sides shall indicate so by sending {IWU-INFO} including the IE <<CODEC-LIST>> with the new codec.

In case the slot type modification fails the initiating side shall switch back to the old codec and indicate so by sending {IWU-INFO} including the IE <<CODEC-LIST>> with the old codec. On receiving this message, the receiving side shall also switch back to the old codec and indicate so by sending {IWU-INFO} including the IE <<CODEC-LIST>> with the old codec.

Each side shall mute its receiving path at sending/receiving CC-SERVICE-ACCEPT.

Receiving {IWU-INFO} shall be a trigger for each side that it may unmute its receiving path.

{IWU-INFO} shall also be sent in case the service change is performed before CONNECT, although the U-Plane will not be connected before CONNECT.

The service change for audio codec change is always followed with sending {IWU-INFO} from both sides. A new service change shall not be initiated until both sides have sent {IWU-INFO}.

7.3.4.1 Service change info

In order to change the codec, the value "Audio Change codec" (see clause 7.7.38 of EN 300 175-5 [5]) shall be inserted in the IE <<Service Change Info>>.

7.3.5 Slot type modification

If the codec change requires a modification in the slot type, the MAC slot change procedure shall be executed as described in EN 300 175-3 [3], clause 10.3.2.

The initiating side of the Network Layer procedure shall also initiate the slot type modification at MAC layer in order to change the audio codec.

7.3.5.1 Failure of slot type modification

On failure of the slot type modification the initiating side shall not release the call but switch back to the previously active codec and indicate so to the receiving side by sending {IWU-INFO} including the IE <<CODEC-LIST>> with the old codec. On receiving this message, the receiving side shall also switch back to the old codec and indicate so by sending {IWU-INFO} including the IE <<CODEC-LIST>> with the old codec.

This can happen both after Service Negotiation and after Service Change. After Service Change the previously active codec shall be restored. After Service Negotiation a mandatory codec shall be used fitting to the previous slot format.

7.3.6 Internal call setup

NOTE 1: This procedure description replaces clause 8.18 of GAP [13].

The following text together with the associated clauses define the mandatory requirements with regard to the present document.

For the initiation of this procedure the "outgoing call request" procedure defined in GAP (clause 8.2 of [13]) shall be used, with the following replacement to the {CC-SETUP} message:

Table 13: Values used within the outgoing {CC-SETUP} message for internal call

Information element	Field within the information element	Standard values within the field/information element	Normative action/comment
<<Basic service>>	<Call class>	9	Internal call

For the termination of this procedure the "incoming call request" procedure defined in GAP (clause 8.12 of [13]) shall be used.

However, if the Portable Part is an NG DECT PP, the NG DECT FP shall use the following replacement to the {CC-SETUP} message:

Table 14: Values used within the incoming {CC-SETUP} message to a New Generation DECT PP for internal call

Information element	Field within the information element	Standard values within the field/information element	Normative action/comment
<<Basic service>>	<Call class>	9	Internal call

NOTE 2: A New Generation DECT PP is identified by the support in the "Terminal capability indication" procedure (clause 7.3.5).

For backward compatibility reasons, New Generation DECT FPs shall use the "external call" call class if the PP is a GAP PP.

NOTE 3: A New Generation DECT PP is identified by the support in the "Terminal capability indication" procedure (clause 7.3.5).

7.3.7 Terminal capability indication

NOTE: This procedure description replaces clause 8.17 of GAP [13].

The PP shall be able to send the <<Terminal capability>> information element and the FP shall be able to receive it at least in {ACCESS-RIGHTS-REQUEST} and when location registration is supported in the {LOCATE-REQUEST}. The following text together with the associated clauses define the mandatory requirements with regard to the present document.

Table 15: Values used within the <<TERMINAL CAPABILITY>> information element

Information element	Field within the information element	Standard values within the field/information element	Normative action/comment
<<Terminal capability>>			
	<Tone capability>	All	
	<Display capability>	All	If PT supports feature (N.24) it shall indicate in this field value which is equal to or higher than 2
	<Profile indicator_1>	"xxxxx1x"B	GAP and/or PAP supported
	<Profile indicator_7>	"xxxxx1x"B	New Generation DECT Wideband speech supported
	<Control codes>	All	If PT supports feature (N.25) it shall indicate in this field value which is equal to or higher than 2

The capabilities in table 16 shall be assumed as default if the following fields in the <<TERMINAL CAPABILITY>> information element are not present.

Table 16: Values assumed as terminal capabilities

Information element	Field within the information element	Standard values within the field/information element	Normative action/comment
<<Terminal capability>>			
	<Echo parameters>	1	Minimum Telephone Coupling Loss (TCL) (> 34 dB)
	<N-REJ>	1	No noise rejection
	<A-VOL>	1	No PP adaptive volume control
	<Slot type capability>	"xxx1x1x"B	Full slot and Long slot (j=640) supported

No echoing of characters is allowed in the FT and therefore the PT would be responsible for displaying dialled digits. All display information from the FT would be assumed to be additional information that the PT shall display in addition. The PT shall logically separate display information originating at the FT and PT. This could be achieved, for example, by one physical display and two logical displays or two physical displays and two logical displays. The key point is that display characters from the PT and FT shall not be simultaneously interleaved/mixed on the same physical display.

7.4 Implementation examples of specific procedures

For detailed examples please refer to the informative Annex D. These diagrams are strongly recommended to be used as implementation guidelines as they are best practice cases and respect all mandatory requirements of the current standard.

7.5 Data Link Control (DLC) layer procedures

This clause specifies the additional DLC layer procedures, messages and information elements required in New Generation DECT wideband speech not described in GAP [13], or incorporating modifications to the GAP specification.

7.5.1 FU1 frame operation

The procedure shall be performed as defined in EN 300 175-4 [4], clauses 12.1 and 12.2. The following text together with the associated clauses define the mandatory requirements with regard to the present document.

Figure 5: Sending a FU1 frame

NOTE: The case when FT initiates differs only in the notations.

The length of a FU1 frame will be k = 40 (full slot) for 32 kbit/s services and k = 80 octets (long slot) for 64 kbit/s services.

One complete frame shall be submitted to/from MAC layer included in a MAC_CO_DATA-req(ind) primitive.

7.6 Medium Access Control (MAC) layer procedures

This clause specifies the additional MAC layer procedures, messages and information elements required in New Generation DECT wideband speech not described in GAP [13], or incorporating modifications to the GAP specification.

7.6.1 MAC services

The FT and PT shall support I_N_minimum_delay symmetric service as defined in EN 300 175-3 [3], clause 5.6.2.1 and clause 10.8.3.1.

The FT and PT may support I_N_normal_delay symmetric service as defined in EN 300 175-3 [3], clause 5.6.2.1 and clause 10.8.3.2.

The FT and PT may support I_{PQ}_error_detection symmetric service as defined in EN 300 175-3 [3], clause 5.6.2.1 and clause 10.8.3.2.

7.6.2 Frame formats and multiplexers

The FT and PT shall support the following frame formats:

- D-field mapping for the full slot structure (physical packet P32), as defined in EN 300 175-3 [3], clause 6.2.1.1.2.
- D-field mapping for the variable slot structure (physical packet P00j), as defined in EN 300 175-3 [3], clause 6.2.1.1.3, with a j value of j = 640.

The FT and PT may support frame format as follows:

- D-field mapping for the variable slot structure (physical packet P00j), as defined in EN 300 175-3 [3], clause 6.2.1.1.3, with a j value of j = 672.
- D-field mapping for the double slot structure (physical packet P80), as defined in EN 300 175-3 [3], clause 6.2.1.1.1.

The FT and PT shall support A-field mapping A-MAP.

The FT and PT shall understand all A field tail identifications (a0, a1 and a2) in the header field as defined in EN 300 175-3 [3], clauses 6.2.1.2 and 7.1.2.

The FT and PT shall support the following B-field field identifications (a4, a5 and a6) as defined in EN 300 175-3 [3], clause 7.1.4:

- U-type: In, "000"B.
- No B-field, "111" B (shall only be used for dummy bearers).
- Long slot required, "101"B.

The FT and PT shall support T-MUX as defined in EN 300 175-3 [3], clause 6.2.2.1.

The FT and PT shall support B-field multiplex E/U MUX type U32a and U64a.

The FT and PT shall support scrambling as defined in EN 300 175-3 [3], clause 6.2.4.

The FT and PT shall provide R-CRC generation and checking as defined in EN 300 175-3 [3], clause 6.2.5.2. The FT and PT shall provide X-CRC generation and checking as defined in EN 300 175-3 [3], clauses 6.2.5.3 and 6.2.5.4.

The PT shall support the normal duty cycle idle_locked mode as defined in EN 300 175-3 [3], clauses 11.3 and 4.3.1.

The FT and PT shall support primary scan procedure as defined in EN 300 175-3 [3], clause 11.8.

All requirements specified in EN 300 444 (GAP) [13], clause 10, shall apply.

7.7 Physical layer (PHL) requirements

7.7.1 Modulation

The FT and PT shall support 2 level Gaussian Frequency Shift Keying (GFSK) modulation as defined by EN 300 175-2 [2] clause 5.

7.7.2 Slot type (Physical packets)

The FT and PT shall support Physical packet P32 (full slot) as defined by EN 300 175-2 [2] clause 4.4.2.

The FT and PT shall support Physical packet P00j (variable slot) as defined by EN 300 175-2 [2] clause 4.4.3, with a j value of j = 640.

The FT and PT may support Physical packet P00j (variable slot) as defined by EN 300 175-2 [2] clause 4.4.3, with a j value of j = 672.

The FT and PT may support Physical packet P80 (double slot) as defined by EN 300 175-2 [2] clause 4.4.4.

All requirements specified in EN 300 444 (GAP) [13], clause 11, shall apply.

All requirements specified in EN 300 175-2 [2], and EN 301 406 [10] (replacing TBR 006 [25]) for 2 level GFSK modulation shall apply.

7.8 Requirements regarding the speech transmission

7.8.1 General

The applicable requirements specified in EN 300 175-8 [8] and EN 300 176-2 [9] (previously covered by TBR 010 [26]) shall be applied.

7.8.2 Speech codecs

The FT and PT shall support ITU-T Recommendation G.726 [15] ADPCM narrow band codec [15], operating at 32 kbit/s rate, as defined by EN 300 175-8 [8] clause 5.1.

The FT and PT shall ITU-T Recommendation G.722 wideband SB-ADPCM 7 kHz codec [17], operating at 64 kbit/s rate, as defined by EN 300 175-8 [8] clause 5.3.

The FT and PT may support ITU-T Recommendation G.711 PCM narrow band codec [16], operating at 64 kbit/s rate, as defined by EN 300 175-8 [8] clause 5.2.

The FT and PT may support ITU-T Recommendation G.729.1 wideband codec [18], operating at 32 kbit/s rate, as defined by EN 300 175-8 [8] clause 5.4.

The FT and PT may support MPEG-4 ER AAC-LD codec [20], operating at 32 kbit/s or 64 kbit/s rate, as defined by EN 300 175-8 [8] clause 5.5.

7.8.3 Audio performance requirements

7.8.3.1 Audio performance requirements for narrowband 3,1 kHz speech

New Generation DECT handsets shall fulfil the audio performance requirements described in EN 300 175-8 [8], clause 7.

7.8.3.2 Audio performance requirements for wideband 7 kHz speech

New Generation DECT handsets shall fulfil the audio performance requirements described in ITU-T Recommendation P.311 [21], as described in EN 300 175-8 [8], clause 9.

7.9 Management procedures

All procedures described in GAP [13] clause 13 shall be supported.

7.10 Application procedures

All procedures described in GAP [13] clause 14 shall be supported.

Annex A (informative): Audio codecs

A.1 Speech and audio coding

A.1.1 Overview

The basic codec for speech in the DECT standard is the "Adaptive Differential Pulse Code Modulation" (ADPCM) with 32 kbit/s as defined in ITU-T Recommendation G.726 [15]. It is of low complexity, offers a bandwidth of 3,1 kHz, introduces a very low delay of 0,125 ms and a quality slightly below the PSTN quality (ITU-T Recommendation G.711 [16] encoding) at 64 kbit/s.

Increasing the bandwidth from narrow band (300 Hz to 3 400Hz) to at least to 150 Hz to 7 000 Hz range ("wide band") will allow to increase decisively the speech quality: voice better encoded on all its frequencies, with a feeling of more transparent communication, a greatly improved sensation of presence and an increased intelligibility and listening comfort.

The following table reviews some speech and audio codecs.

Table 17: Codec overview

	Type	Bandwidth [kHz]	Sampling rate [kHz]	Bit rate [kbit/s]	Frame [ms]
ITU-T Recommendation G.711 [16]	LOG PCM	0,3 to 3,4	8	64	0,125
ITU-T Recommendation G.726 [15]	ADPCM	0,3 to 3,4	8	16, 24, 32, 40	0,125
ITU-T Recommendation G.722 [17]	Sub-Band ADPCM	0,05 to 7	16	64, 56, 48	0,125
ITU-T Recommendation G.729.1 [18]	EV-CELP Time Domain Bandwidth Extension (TDBWE) Transform Coding (MDCT)	0,05 to 7	16	8, 12, 14, 16, 18, 20, 22, 24, 26, 28, 30, 32	20
ISO/IEC 14496-3 [19]	MPEG-4 ER AAC-LD Advanced Audio Coding Low Delay	up to 20	up to 48	range of bit rates (around 24 to 96)	10 to 20 (depends on sampling rate)

A.1.2 Narrow band speech coding

ITU-T Recommendation G.726 narrow band codec [15] is mandatory for New Generation DECT in order to ensure interoperability with existing DECT systems.

ITU-T Recommendation G.711 narrow band codec [16] is optional for New Generation DECT in order to improve the quality of narrow band communications: slightly higher intrinsic voice quality, better robustness to transmission errors and no transcoding for PSTN calls.

Table 18: ITU-T Narrow band Speech codec for New Generation DECT

Standard	**ITU-T Recommendation *G.726***	**ITU-T Recommendation *G.711***
	ADPCM	LOG PCM
Date	*1990*	*1972*
Bandwidth	*300 Hz to 3 400 kHz*	*300 Hz to- 3 400 kHz*
Sampling rate	*8 kHz*	*8 kHz*
Bit rate(kbit/s)	*16, 24, 32, 40*	*64*
Embedded Scalability	*No*	*No*
Type	*ADPCM*	*LOG PCM*
Frame size	*0,125 ms*	*0,125 ms*
Algorithmic Delay	*0,125 ms*	*0,125 ms*
Complexity	*12 MIPS*	*0,01 MIPS*
RAM (kBytes)	*1*	*≈0*

A.1.3 Wideband Speech coding

ITU-T Recommendation G.722 codec [17] is chosen as mandatory wideband codec for New Generation DECT in order to greatly increase the voice quality by extending the bandwidth from narrow band to wideband.
ITU-T Recommendation G.722 [17] provides a high wideband quality at a bit rate of 64 kbit/s with low complexity and very low delay.

In addition, ITU-T Recommendation G.729.1 codec [18] is recommended as an optional codec for wideband speech to provide even higher wideband quality and better robustness to frames/packets losses than
ITU-T Recommendation G.722 [17] at much lower dynamically adaptable bit rates. This allows a better transport efficiency on the network side and over the DECT air interface (fits in one single current DECT slot). In addition, it is seamless interoperable with largely deployed ITU-T Recommendation G.729 based VoIP networks and terminals.
ITU-T Recommendation G.729.1 [18] encodes signals in frames of 20 ms. It is a scalable codec operating at bitrates of 8 kbit/s and from 12 kbit/s to 32 kbit/s per steps of 2 kbit/s, both in narrowband or in wideband from 14 kbit/s.

Table 19: ITU-T Wideband Speech codec for New Generation DECT

Standard	**ITU-T Recommendation *G.722***	**ITU-T Recommendation *G.729.1***
	SB-ADPCM	*G.729 EV*
Date	*1988*	*2006*
Bandwidth	*50 Hz to 7 kHz*	*50 Hz to 4 kHz* *50 Hz to 7 kHz (bit rates ≥ 14 kbit/s)*
Sampling rate	*16 kHz*	*8kHz / 16kHz*
Bit rate(kbit/s)	*64, 56, 48*	*8, 12, 14, 16, 18, 20, 22, 24, 26, 28, 30, 32*
Embedded Scalability	*Yes*	*Yes (interoperable at 8 kbit/s with G.729)*
Type	*Sub-Band ADPCM*	*EV-CELP* *Time Domain Bandwidth Extension (TDBWE)* *Transform Coding (MDCT)*
Frame size	*0,125 ms*	*20 ms*
Algorithmic Delay	*1,625 ms*	*48,9375 ms*
Complexity	*10 MIPS*	*35,8 WMOPS based on new STL2005* *(34,7 WMOPS based on STL2000)*
RAM (kBytes)	*1*	*17,4*

PLC (Packet loss Concealment) ITU-T Recommendation G.722 [17] Appendix III and IV (NG1.S7): Appendices III and IV describe packet loss concealment solutions extending ITU-T Recommendation G.722 [17] decoder. These algorithms may be optionally implemented to improve voice quality in degraded transmission conditions where packets/frames may be lost (in the IP network or on the DECT air interface). Both appendices meet the same quality requirements but address two different quality/complexity trade offs:

- Appendix III aims at maximizing the robustness at a price of additional complexity (+0,1 to 0,2 MOS in comparison with appendix IV in degraded conditions).
- Appendix IV proposes an optimized complexity/quality trade off with almost no additional complexity compared with ITU-T Recommendation G.722 [17] normal decoding (0,07 WMOPS).

Since ITU-T Recommendation G.722 [17] does not incorporate any mechanism to cope with lost frames/packets, the use of a PLC algorithm is strongly recommended to avoid annoying effects in case of packet/frame losses.

NOTE: ITU-T Recommendation G.729.1 [18] already incorporates a high efficiency packet loss concealment mechanism.

Table 20: ITU-T Recommendation G.722 [17] PLC Appendices for New Generation DECT

PLC	*Appendix III*	*Appendix IV*
Date	*2006*	*2006*
Type	*Full band waveform extrapolation Re-encoding and signal monitoring, re-phasing and time warping*	*Split-band waveform extrapolation Partial state reset, cross fading*
Packetization size/ms		
Complexity Observed Worst Case in WMOPS (based on STL2005)	*5,87 WMOPS (10 ms packets) 5,60 WMOPS (20 ms packets)*	*3,18 WMOPS (10 ms packets) 3,15 WMOPS (20 ms packets)*
Total RAM (10 ms packets) (Static + Scratch) Total RAM (20 ms packets) (Static + Scratch) In 16 bits Words	*2 184 (10 ms packets) (1 118 + 826) 1 944 (20 ms packets) (1 118 + 1066)*	*1 659 (10 ms packets) (967 + 692) 1 659 (20 ms packets) (967 + 963)*
Program ROM (in number of basic-ops and function calls)	*2 410*	*1 061*
Table ROM (in 16 bits Words	*1 414*	*882*

To handle several codecs (at least ITU-T Recommendation G.726 [15] and ITU-T Recommendation G.722 [17]), New Generation DECT will support a codec selection and switching mechanism. This may consequently allow the use of other codecs that could be recommended in next releases as additional optional codecs according to future application or interoperability needs.

A.1.4 Super-wideband speech and audio coding

The MPEG-4 ER AAC-LD 64 kbit/s audio codec [20] is recommended as an optional codec for super-wideband speech. In order to provide high quality for music streaming or other multimedia applications mixing speech and music, the bandwidth can be further extended to superwideband (50 Hz to 14 kHz) and above (up to full audio bandwidth (20 Hz to 20 000 Hz). The codec may be also suitable for wideband speech.

MPEG-4 ER AAC-LD is designed for high quality communication application including all kind of audio signals e.g. speech and music. It provides an audio bandwidth of 14 kHz at a bitrate of 64 kbit/s. MPEG 4 ER AAC-LD is standardized in ISO/IEC 14496-3 [20]. The frame size is 10 ms and the algorithmic delay 20 ms. It may also be optionally used in 32 kbit/s mode. It provides a recommended bandwidth of 11,5 kHz. The frame size is 20 ms and the algorithmic delay 40 ms.

Table 21: MPEG-4 ER AAC-LD Audio codec for NG DECT

Standard	*MPEG-4 ER AAC- LD 32 kbit/s*	*MPEG-4 ER AAC-LD 64 kbit/s*
Date	*2000/2006*	*2000/2006*
recommended Bandwidth	*11,5 kHz*	*14 kHz*
Sampling rate	*24 kHz*	*48 kHz*
Bit rate(kbit/s)	*32*	*64*
Embedded Scalability	*no*	*no*
Type	*perceptual audio codec*	*perceptual audio codec*
Frame size	*20 ms (480 samples)*	*10 ms (480 samples)*
Algorithmic Delay	*40 ms*	*20 ms*
example Complexity	*~13 MIPS (encoder) ~5 MIPS (decoder)*	*~25 MIPS (encoder) ~10 MIPS (decoder)*
example RAM (kBytes)	*~28 kbyte (encoder) ~13 kbyte (decoder) IO Buffer not included*	*~28 kbyte (encoder) ~13 kbyte (decoder) IO Buffer not included*

As for wideband speech codec, the codec selection and switching mechanism may allow the use of other configurations or other optional super-wideband speech and audio codecs according to the applications or interoperability needs.

Annex B (normative): Audio patterns to indicate IP packet losses on the DECT link

B.1 Audio patterns to indicate IP packet losses.

The following annex is applicable for:

- New Generation DECT FP connected to a VoIP network (directly or through a gateway) with audio frames coming from the VoIP network decoded in the PT (no transcoding done between the network and the DECT link).

B.1.1 Insertion of audio patterns

Upon detection of a packet loss or a corrupted IP packet, the FP shall insert an appropriate audio pattern on the DECT link in direction of the Portable part.

These patterns may be repeated as many times as necessary on the DECT link to replace the faulty IP packet. For example if a 20 ms RTP VoIP packet is lost. The pattern shall be inserted twice on the DECT link.

B.1.2 Reception of audio patterns

Upon reception of these patterns in the PT:

- If a PLC is available on the PT with the current activated codec, the PT should activate it.
- If no PLC is available on the PT, the PT should decode this pattern as normal audio frame in the decoder.

However, it is recommended to use standardized PLC mechanism in order to improve audio robustness to packet losses.

It is not recommended to use these patterns in the PT to FT direction as the PT should always be able to provide correct audio frames as soon as the U-plane is established on the DECT link.

B.1.3 Contents of the audio patterns

The following patterns were chosen because:

- They correspond to 10 ms of audio decoded signal (20 ms for 20 ms audio framed codec).
- They generate a very low energy decoded signal if no PLC mechanism is available on the terminal.

Their occurrence in normal audio encoded bitstream is quite impossible.

For MPEG-4 ER AAC-LD, the pattern is a standard conform MPEG-4 ER AAC-LD frame. If no PLC mechanism is available on the terminal side, the pattern forces the decoder to fade out smoothly within 10 ms (20 ms at 32 kbps). The same pattern can be used for both, the 64 kbit/s and 32 kbit/s service. The transport format is a MPEG-4 Access Unit.

B.1.4 Packet loss patterns for ITU-T Recommendation G.722

ITU-T Recommendation G.722 [17] law packet loss pattern. 640 bits. To be inserted in 1 long slot.

```
/* Pattern is 0xFF (repeated 80 times) */
Pattern[80] ={
0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF,
0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF,
0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF,
0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF,
0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF,
0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF,
0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF,
0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF,
0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF,
0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF};
```

B.1.5 Packet loss patterns for ITU-T Recommendation G.711

ITU-T Recommendation G.711 [16] law packet loss. 640 bits. To be inserted in 1 long slot.

```
/* Pattern is 0xD5 (repeated 24 times), 0x55 (repeated 32 times), 0xD5 (repeated 24 times) */
Pattern[80] ={
0xD5, 0xD5, 0xD5, 0xD5, 0xD5, 0xD5, 0xD5, 0xD5,
0xD5, 0xD5, 0xD5, 0xD5, 0xD5, 0xD5, 0xD5, 0xD5,
0xD5, 0xD5, 0xD5, 0xD5, 0xD5, 0xD5, 0xD5, 0xD5,
0x55, 0x55, 0x55, 0x55, 0x55, 0x55, 0x55, 0x55,
0x55, 0x55, 0x55, 0x55, 0x55, 0x55, 0x55, 0x55,
0x55, 0x55, 0x55, 0x55, 0x55, 0x55, 0x55, 0x55,
0x55, 0x55, 0x55, 0x55, 0x55, 0x55, 0x55, 0x55,
0xD5, 0xD5, 0xD5, 0xD5, 0xD5, 0xD5, 0xD5, 0xD5,
0xD5, 0xD5, 0xD5, 0xD5, 0xD5, 0xD5, 0xD5, 0xD5,
0xD5, 0xD5, 0xD5, 0xD5, 0xD5, 0xD5, 0xD5, 0xD5};
```

ITU-T Recommendation G.711 [16] law packet loss. 640 bits. To be inserted in 1 long slot.

```
/* Pattern is 0xFF (repeated 24 times), 0x7F (repeated 32 times), 0xFF (repeated 24 times) */
Pattern[80] ={
0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF,
0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF,
0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF,
```

```
0x7F, 0x7F, 0x7F, 0x7F, 0x7F, 0x7F, 0x7F, 0x7F,
0x7F, 0x7F, 0x7F, 0x7F, 0x7F, 0x7F, 0x7F, 0x7F,
0x7F, 0x7F, 0x7F, 0x7F, 0x7F, 0x7F, 0x7F, 0x7F,
0x7F, 0x7F, 0x7F, 0x7F, 0x7F, 0x7F, 0x7F, 0x7F,
0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF,
0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF,
0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF, 0xFF};
```

B.1.6 Packet loss patterns for ITU-T Recommendation G.726

ITU-T Recommendation G.726 [15].

No pattern is proposed for the reason that a transcoding is done in the Fixed part, so the PLC is not done in the PP.

B.1.7 Packet loss patterns for ITU-T Recommendation G.729.1

ITU-T Recommendation G.729.1 [18] packet loss. 640 bits. To be inserted in 2 full slots. Audio frames are 20ms.

The payload format described in Annex C.1 "transport of the ITU-T Recommendation G.729.1 [18] audio frame in full-slot mode" shall be used. The following patterns must replace the faulty packets in the coded bitstream in case of packet loss:

First full slot

In the first full slot, bad frame indicator is set BFI=1, First frame part: FPA1=0 FPA2=0, Parity even is set PA=1.

Pattern[40] ={ **in full slot1**

```
0x81, 0x00, 0x00, 0x00, 0x00, 0x00, 0x00, 0x00,
0x00, 0x00, 0x00, 0x00, 0x00, 0x00, 0x00, 0x00,
0x00, 0x00, 0x00, 0x00, 0x00, 0x00, 0x00, 0x00,
0x00, 0x00, 0x00, 0x00, 0x00, 0x00, 0x00, 0x00,
0x00, 0x00, 0x00, 0x00, 0x00, 0x00, 0x00, 0x00};
```

Second full slot

In the second full slot, bad frame indicator is set BFI=1, second frame part: FPA1=0 FPA2=1, Parity even is not set PA=0.

Pattern[40] ={ **in full slot2**

```
0x03, 0x00, 0x00, 0x00, 0x00, 0x00, 0x00, 0x00,
0x00, 0x00, 0x00, 0x00, 0x00, 0x00, 0x00, 0x00,
0x00, 0x00, 0x00, 0x00, 0x00, 0x00, 0x00, 0x00,
0x00, 0x00, 0x00, 0x00, 0x00, 0x00, 0x00, 0x00,
0x00, 0x00, 0x00, 0x00, 0x00, 0x00, 0x00, 0x00};
```

B.1.8 Packet loss patterns for MPEG-4 ER AAC-LD

MPEG-4 ER AAC-LD, 64 kbit/s

MPEG-4 ER AAC-LD packet loss pattern. 640 bits. To be inserted in 1 long slot (64 kbit/s). Audio frames are 10 ms.

Pattern[80]={

```
0x00,0x00,0x00,0x00,0x00,0x00,0x00,0x00,0x02,0x32,
0x92,0x0A,0x6A,0x2A,0xFA,0x62,0x7A,0x9A,0x9D,0x2D,
0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,
0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,
0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,
0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,
0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,0x28,0x00
};
```

MPEG-4 ER AAC-LD, 32 kbit/s

MPEG-4 ER AAC-LD packet loss pattern. 640 bits. To be inserted in 2 full slots (32 kbit/s).Audio frames are 20 ms.

First full slot

Pattern[40]={**in full slot1**

```
0x00,0x00,0x00,0x00,0x00,0x00,0x00,0x00,0x02,0x32,
0x92,0x0A,0x6A,0x2A,0xFA,0x62,0x7A,0x9A,0x9D,0x2D,
0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,
0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,0x2D
};
```

Second full slot

Pattern[40]={**in full slot2**

```
0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,
0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,
0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,
0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,0x2D,0x28,0x00
};
```

Annex C (normative): Configuration signalling for specific codecs

C.1 MPEG-4 ER AAC-LD configuration signalling

If the MPEG-4 ER AAC-LD voice service is used as a communication service, some out of band signalling increases the interoperability between FP and the IP world. Therefore the following two <<IWU to IWU>> elements shall be used to signal the available capabilities. The first <<IWU to IWU>> element shall be used to signal the supported capabilities of the device (MPEG4CapabilityElement) and the second element shall be used to signal the selected configuration (MPEG4ConfigurationElement).

Both elements contain level and transport format information whereas AudioSpecificConfig (ASC) is transmitted only within the MPEG4ConfigurationElement.

- **Level:** Used Low Delay AAC Profile level.
- **Transport format:** RFC3640 [21] transmits plain MPEG-4 access units whereas in RFC3016 [23] LATM transport format is used. Both formats can be converted into each other. To avoid the conversion process transmission of both formats over the New Generation DECT link is possible.
- **AudioSpecificConfig:** The usage of ER tools is signalled within ASC. In packet oriented IP transmission ER tools are normally not used up to now. This has to be signalled to the decoder. The AudioSpecificConfig is included in the RTP Payload format description RFC 3640 [22]/RFC 3016 [23]. With it, the FP can directly transmit the AudioSpecificConfig to the IP world and back to the PP. Thus the transportation of the AudioSpecificConfig is possible.

The <<IWU to IWU>> Elements has to be transmitted in certain messages if the IE <<Codec List>> [5] includes MPEG-4 ER AAC-LD. The format of both <<IWU to IWU>> Elements differs only in the occurrence of the AudioSpecificConfig. The ASC occurrence depends on the length of the corresponded <<IWU TO IWU>> Element. The length of the MPEG-4 Capability Element is 4 octets while the MPEG4 ConfigurationElement which includes an ASC exceeds 4 octets.

C.1.1 <<IWU to IWU>> element to signal the supported capabilities (MPEG4CapabilityElement)

If a new PP is registered at the FP side it is important for both, PP and FP, to get information about the MPEG-4 ER AAC-LD capability of the responding part. Therefore it is possible to determine the fitting configuration during a call establishment without any further negotiation process.

The following <<IWU to IWU>> Element will handle the signalling of the supported capability and shall be used in the MM-Messages <<LOCATE-REQUEST>>, <<LOCATE-ACCEPT>>, <<ACCESS-RIGHTS-REQUEST>>, <<ACCESS-RIGHTS-ACCEPT>>.

In case of default configuration according to table C.1, no MPEG4CapabilityElement shall be sent.

Information element coding:

Bit:	8	7	6	5	4	3	2	1	Octet:
	0	<< IWU to IWU>> (0x77)							**1**
	Length of Contents (L)								**2**
	1	S/R	Protocol Discriminator (0x25 MPEG-4 ER AAC-LD Configuration Description)						**3**
	Transport format capability				MPEG-4 ER AAC-LD Level capability				**4**

The "Transport format capability" field contains the supported transport formats and shall be interpreted as follows:

Bit:	8	7	6	5	Octet
	reserved	reserved (e.g. MPEG-4 LOAS AudioPointerStream())	MPEG-4 LOAS AudioSyncStream()	MPEG-4 Access Units (content of er_raw_data_block())	4

Whereas MPEG-4 LOAS AudioSyncStream() and MPEG-4 Access Unit capability is mandatory for a New Generation DECT device which supports MPEG-4 ER AAC-LD.

The content of the "MPEG-4 ER AAC-LD Level capability" field describes the supported Low Delay AAC Profile level [19]. Higher levels also include the support of lower levels:

MPEG-4 ER AAC-LD Level capability Coding (Octet 4):

Bits	**4 3 2 1**	**Meaning**
	0 0 0 0	reserved for ETSI use
	0 0 0 1	ISO/IEC 14496-3:2005 Low Delay AAC Profile level 1 [19]
all other values		reserved

Table C.1 Default Coding of MPEG4CapabilityElement

Octet	Information Element Field	Field Value
4	MPEG-4 ER AAC-LD Level capability	ISO/IEC 14496-3 [19]:2005 Low Delay AAC Profile level 1 (="0001"B)
4	Transport format capability	Bit 5 MPEG-4 Access Units = 1 Bit 6 MPEG-4 LOAS 1 AudioSyncStream() = 1

C.1.2 <<IWU to IWU>> element to signal the used Configuration (MPEG4ConfigurationElement)

During the connection establishment between PP, FP and the IP world, the selected transport format and the selected Low Delay AAC Profile level [19] is signalled. Furthermore the transport of the AudioSpecficConfig (detailed description can be found in [20]) is used to signal MPEG-4 ER AAC-LD error resilience tools. If the IE <<Codec List>> provides MPEG-4 ER AAC-LD, the following <<IWU to IWU>> element has to be used in the following messages:

Messages:

<<CC-SETUP>>, <<CC-CONNECT>>, <<CC-INFO>>, <<CC-SETUP-ACK>>, <<CC-CALL-PROC>>, <<CC-ALERTING>>,<<IWU-INFO>>,<<CC-SERVICE-CHANGE>>.

Information element coding:

Bit:	8	7	6	5	4	3	2	1	Octet:
	0	<< IWU to IWU>> (0x77)							**1**
	Length of Contents (L)								**2**
	1	S/R	Protocol Discriminator (0x25 MPEG-4 ER AAC-LD Configuration)						**3**
	Transport format				MPEG-4 ER AAC-LD Level				**4**
	content of AudioSpecificConfig								**5**
									…
									L+2

The "Transport format" field contains the selected transport format and should be interpreted as follows:

Bit:	8	7	6	5	Octet
	reserved	reserved (Audio Pointer stream)	MPEG-4 LOAS AudioSyncStream()	MPEG-4 Access Units (content of er_raw_data_block())	4

Whereas only one bit of these fields is set to signal the used transport format. The content of the "MPEG-4 ER AAC-LD Level" describes a value indicating which Low Delay AAC Profile Level [19] is used.

MPEG-4 ER AAC-LD Level Coding (Octet 4):

Bits	**4 3 2 1**	**Meaning**
	0 0 0 0	reserved for ETSI use
	0 0 0 1	ISO/IEC 14496-3:2005 Low Delay AAC Profile level 1 [19]
all other values		reserved

The Octets 5 to L+2 contains the AudioSpecificConfig [20].

Annex D (informative): Recommended implementation of procedures

D.1 Examples of implementation of specific procedures

D.1.1 General

In the following clauses, several examples are depicted.

It has to be noted that the sequences are only examples, it cannot be mandatory that the message flows shall always be exactly in the described way.

For example it should remain in the hand of each device whether a service is confirmed at the latest possibility with CC_CONNECT or in an earlier message with the consequence that a service negotiation might be more probable.

Also it should remain in the hand of each base station, whether CALL-PROCEEDING is sent or directly CC-CONNECT.

Also it should remain in the hand of each device, in which situation it establishes a long-slot connection or prefers to establish a full-slot connection, perhaps at the risk that connection modification will be more probable.

Therefore the diagrams can only be used as recommendations.

The connection of the U-Plane is not marked in the diagrams, but done as usually in DECT with sending/receiving of CC-CONNECT for outgoing calls and with sending/receiving CC-CONNECT-ACK for incoming calls. In addition to this, the IE <<Progress Indicator>> can be sent from FP to PP in order to connect the U-Plane.

Where the diagrams contain "paging for longslot", it should be kept in mind, that the FP are only paging for the establishment of the slot type "long slot" in case the PP indicated the support of the corresponding long slot format in the terminal capabilities.

D.1.2 Outgoing wideband call

D.1.2.1 Outgoing wideband call, no codec list, ITU-T Recommendation G.722 chosen

Use case: User requests a wideband call and the network supports it.

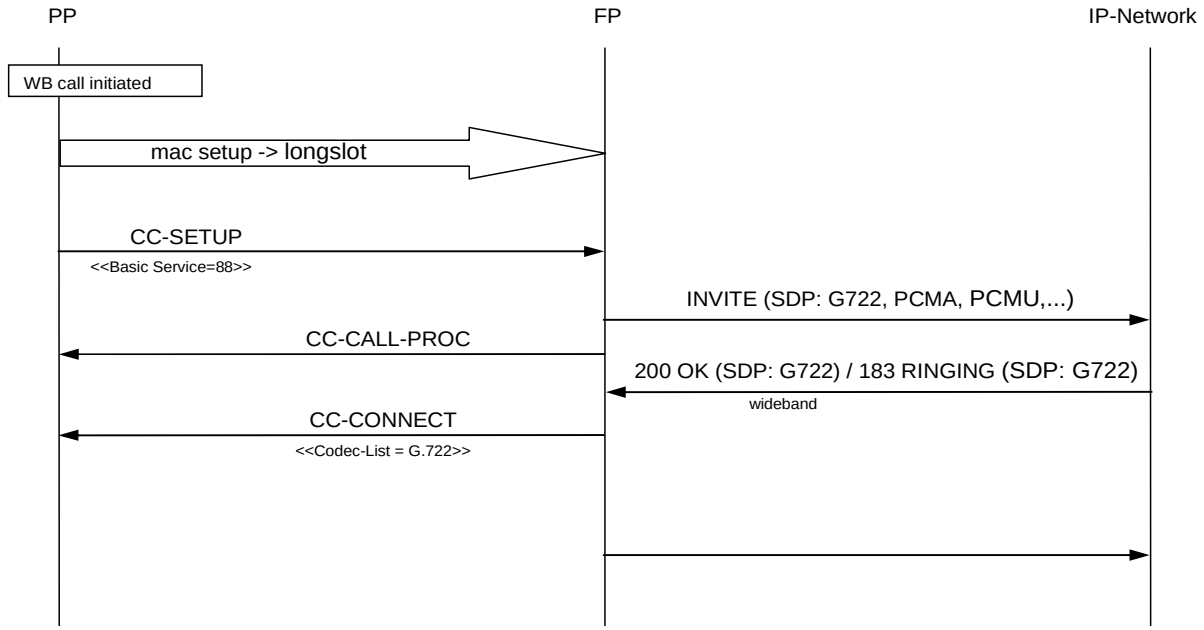

Figure D.1: Outgoing wideband call, no codec list, ITU-T Recommendation G.722 chosen

The use of the basic service "wideband speech default setup attributes" implies the offer of the codec-list indicated in the last (location) registration or at subscription registration. Since in this example no other Codec List shall be indicated, the IE <<Codec-List>> can be omitted in CC-SETUP.

In a response message (here CC-CONNECT), the peer entity confirms the chosen service with <<Codec-List>>.

The following tables are showing the IE codings for this example:

Table D.1: Values used within the {CC-SETUP} message

Information element	Information Element Coding	Field within the information element	Standard values within the field/information element	Normative action/comment
<< Basic Service >>	e0 88			
		<< Call class >>	1000	Normal call setup
		<< Basic Service >>	1000	Wideband speech default setup attributes

Table D.2: Values used within the {CC-CONNECT} message

Information element	Information Element Coding	Field within the information element	Standard values within the field/information element	Normative action/comment
<<Codec-List>>	7c 04 90 03 00 81			
		<< Negotiation indicator>>		
		<<1st codec identifier>>	0000011	ITU-T Recommendation G.722 [17]
		<< MAC service >>	0000	In_ min_delay
		<< C-Plane routing >>	000	Cf never
		<< Slot size >>	0001	long slot

D.1.2.2 Outgoing Call Wideband, codec list, negotiation results in Wideband

Use case: User requests a wideband call but specifies another NB codec in the SETUP (instead of ITU-T Recommendation G.726 [15]), but network only supports ITU-T Recommendation G.722 [17].

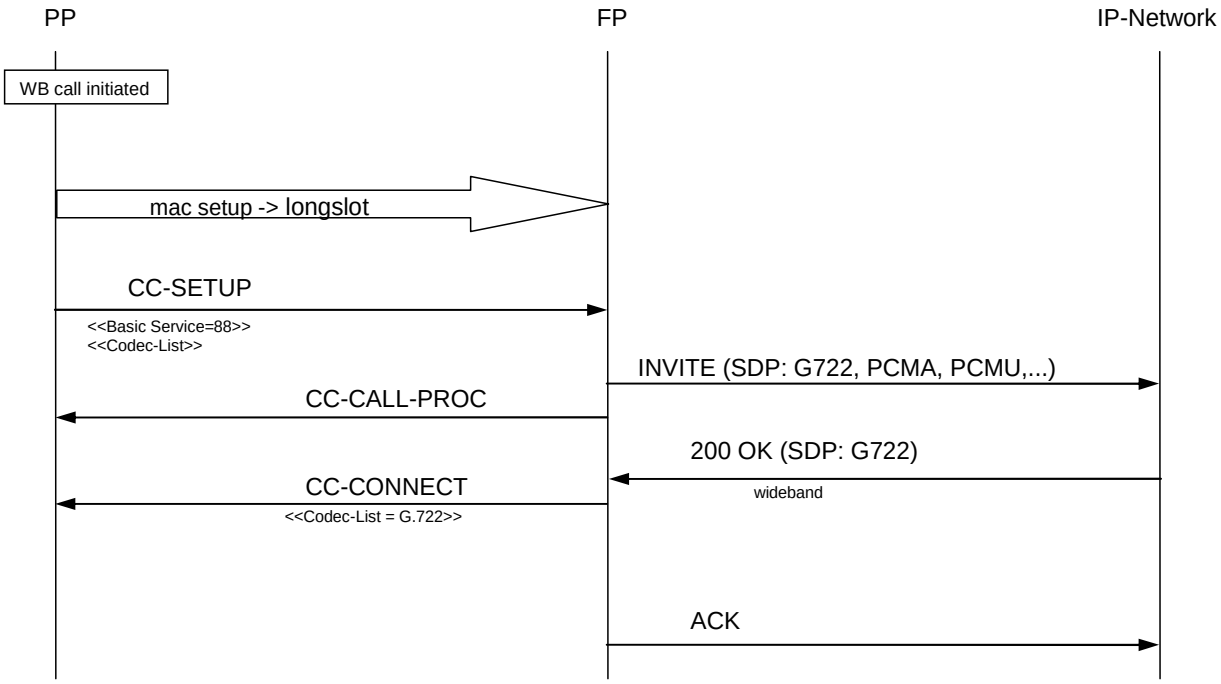

Figure D.2: Outgoing Call Wideband, codec list, negotiation results in Wideband

Table D.3: Values used within the {CC-SETUP} message

Information element	Information Element Coding	Field within the information element	Standard values within the field/information element	Normative action/comment
<< Basic Service >>	e0 88			
		<< Call class >>	1000	Normal call setup
		<< Basic Service >>	1000	Wideband speech default setup attributes
<<Codec-List>>	7c 07 90 03 00 01 02 00 84			
		<< Negotiation indicator>>		
		<<1st codec identifier>>	0000011	ITU-T Recommendation G.722 [17]
		<< MAC service >>	0000	In_ min_delay
		<< C-Plane routing >>	000	Cf never
		<< Slot size >>	0001	long slot
		<<2nd codec identifier>>	0000100	ITU-T Recommendation G.711 [16]
		<< MAC service >>	0000	In_ min_delay
		<< C-Plane routing >>	000	Cf never
		<< Slot size >>	0001	long slot
		<<3rd codec identifier>>	0000010	G.726
		<< MAC service >>	0000	In_ min_delay
		<< C-Plane routing >>	000	Cf never
		<< Slot size >>	0100	Full slot

Here, a new codec-list is offered in the CC-SETUP.

Again, in a response message (here CC-Connect), the peer entity confirms the chosen service with the IE <<Codec-List>>.

Table D.3 shows the IE codings for this example.

D.1.2.3 Outgoing call with progress indicator with negotiation results in CC-INFO

Use case: User requests a wideband call and Fixed Part uses Progress indicator messages.

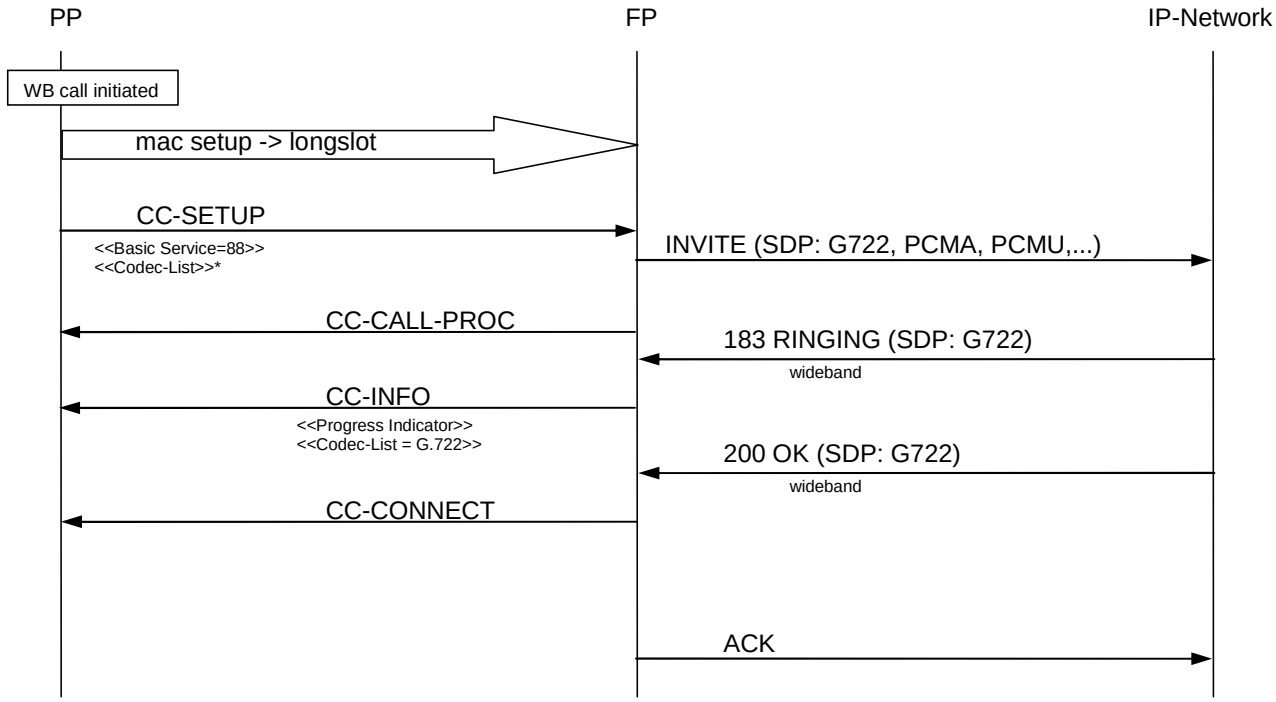

Figure D.3: Outgoing call with progress indicator with negotiation results in CC-INFO

In case the IE <<Progress Indicator>> is used to connect the U-Plane before {CC-CONNECT}, the service shall be confirmed at latest in the same message.

If the service negotiation via the network interface results in the need to change the codec in DECT again, this has to be done with the service change procedure (before or after CC-CONNECT).

D.1.2.4 Outgoing call with progress indicator; with negotiation results in CC-INFO codec change in 200 OK

Use case: User requests outgoing wideband call but the codec changes between RINGING and OK messages on the IP network.

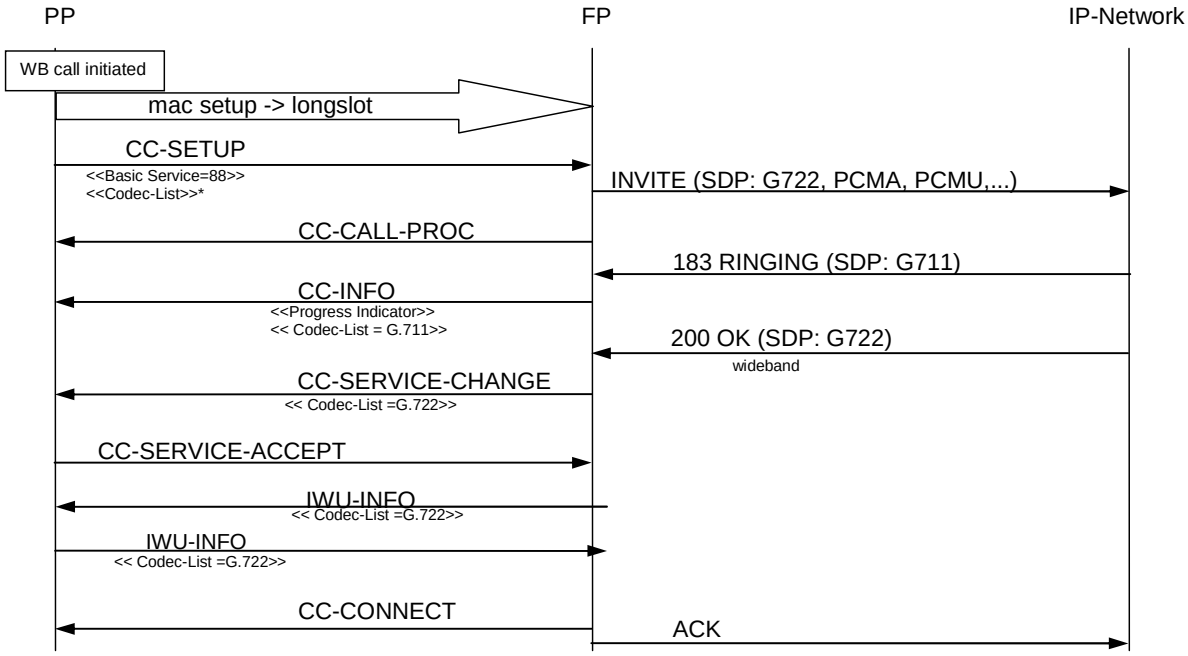

Figure D.4: Outgoing call with progress indicator; with negotiation results in CC-INFO codec change in 200 OK

In this case the codec is changed but the slot format remains unchanged. {IWU-INFO}is exchanged although before CONNECT.

D.1.2.5 Outgoing Call Wideband, negotiation results in Narrowband

Use case: user requests wideband outgoing call but the IP network does not support wide band

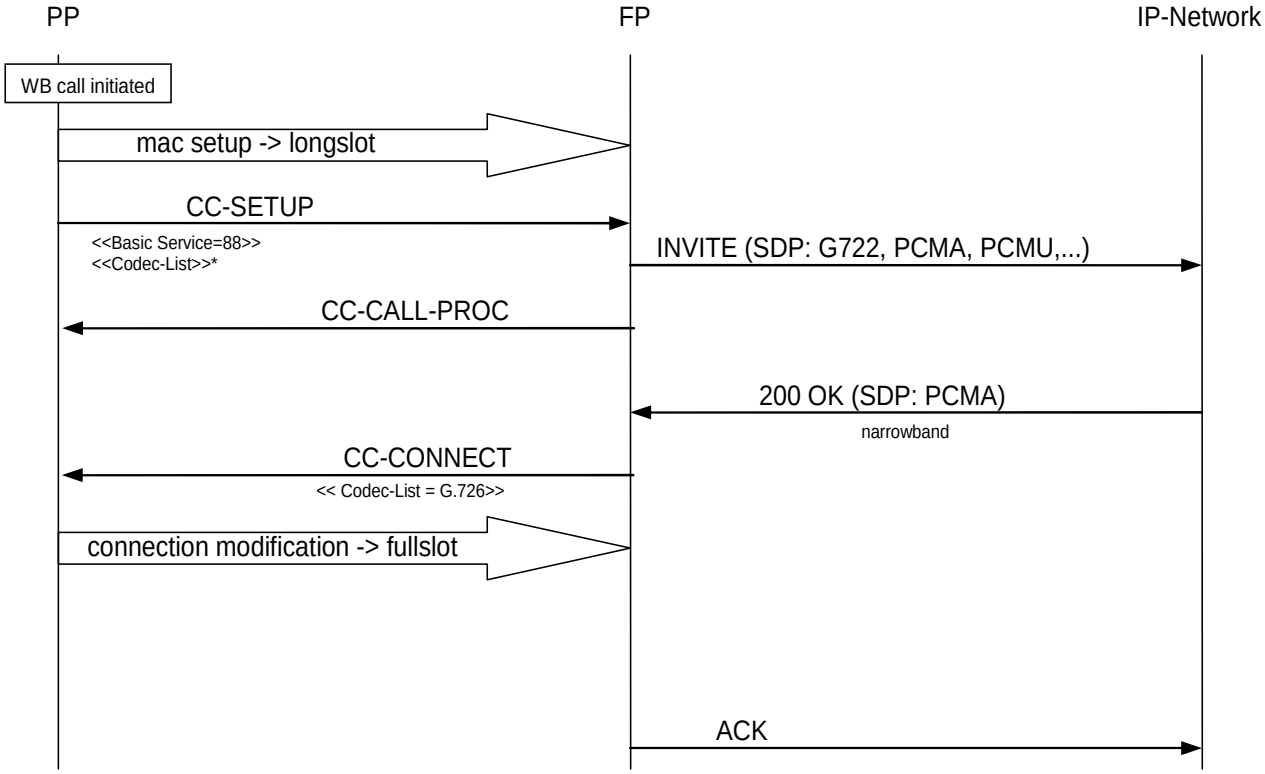

Figure D.5: Outgoing Call Wideband, negotiation results in Narrowband

D.1.2.6 Outgoing Call Wideband, negotiation results in longslot

Use case: User requests outgoing wideband call but establishes the radio link in full-slot.

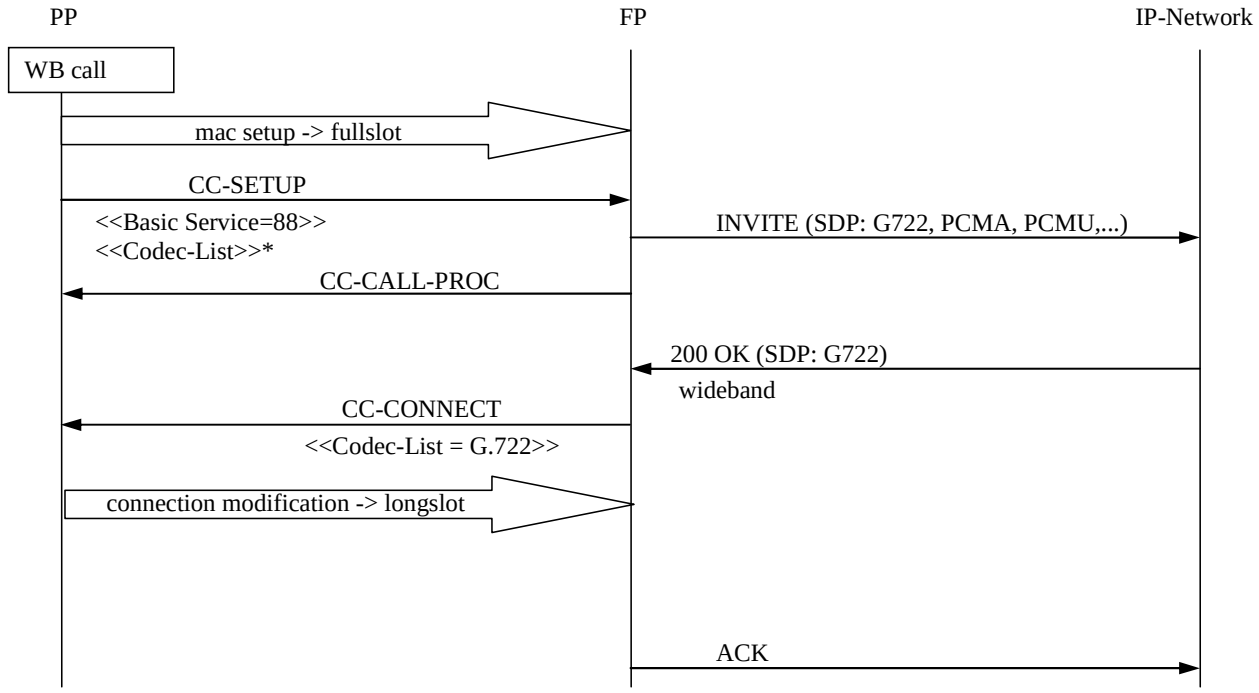

Figure D.6: Outgoing Call Wideband, negotiation results in longslot

It is also possible to establish a full slot connection during call establishment and modify it to a long slot connection after negotiation, if necessary. However it is not recommended, since it might be possible that modification from fullslot to longslot fails due to limited MAC resources (result would appear in the connection attributes of the CC-CONNECT message)

D.1.3 Incoming Call Wideband

D.1.3.1 Incoming Call Wideband, negotiation results in Wideband

Use case: Explicit in the figures title.

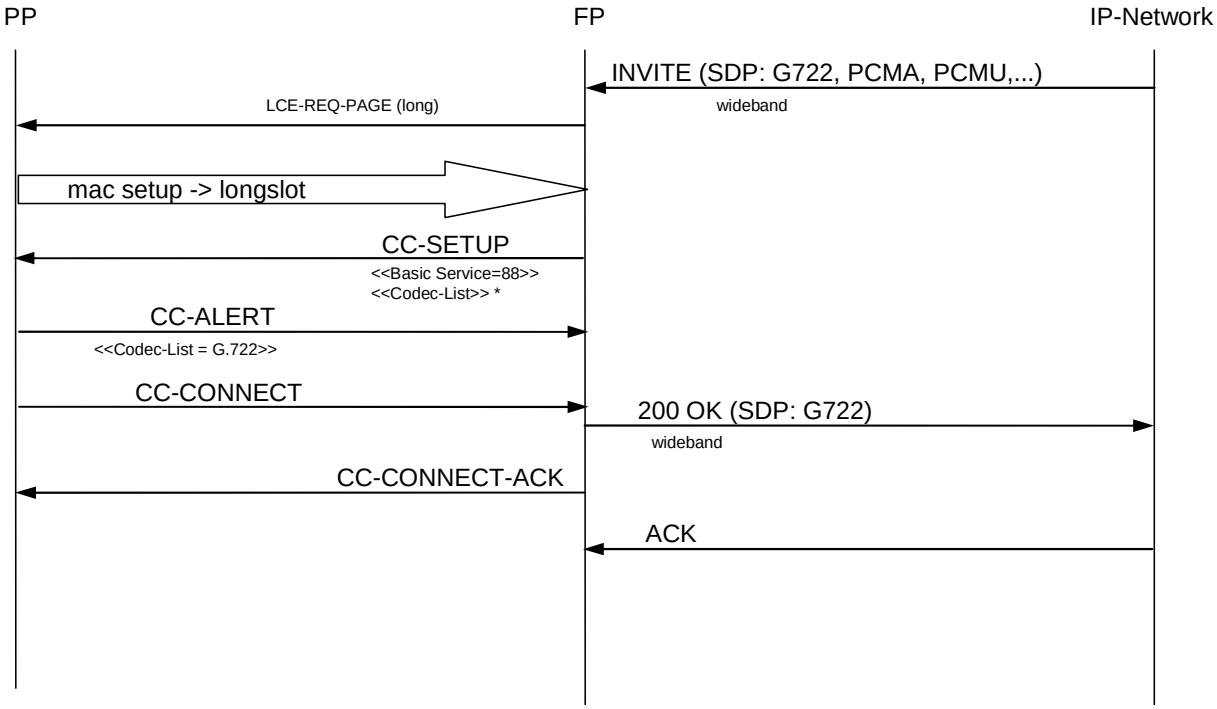

Figure D.7: Incoming Call Wideband, negotiation results in Wideband

D.1.3.2 Incoming Call Wideband, negotiation results in Narrowband

Use case: User receives incoming call in wideband preferred but a narrow band connection is set up (for example if we pickup the call on a NB handset)

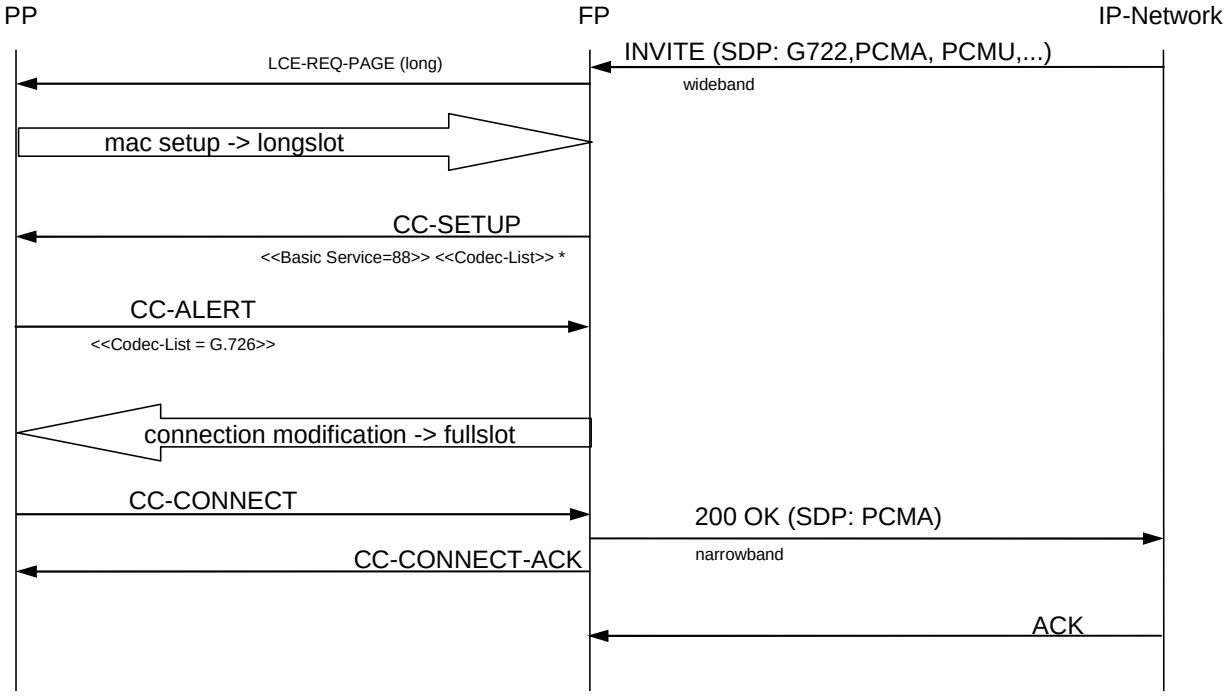

Figure D.8: Incoming Call Wideband, negotiation results in Narrowband

D.1.3.3 Incoming Call Wideband, No SDP Offer in Invite, negotiation results in Narrowband

Use case: User receives an incoming call, FP proposes to establish in WB but network forces narrow-band.

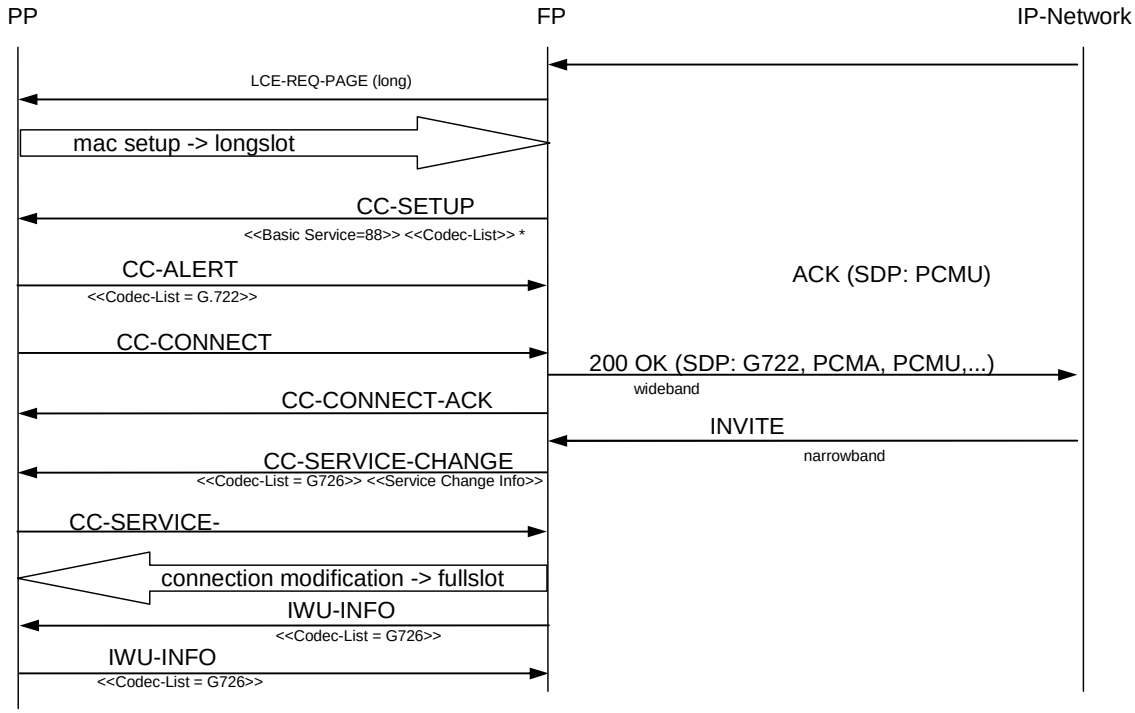

Figure D.9: Incoming Call Wideband, No SDP Offer in Invite, negotiation results in Narrowband

In this case the FP has to assume a service in order to be able to propose one to the PP in the {CC-SETUP} message.

D.1.4 Service Change

D.1.4.1 Service Change from Wideband to Narrowband; re-negotiation initiated from IP-Network

Use case: network requests a codec change (for example call waiting).

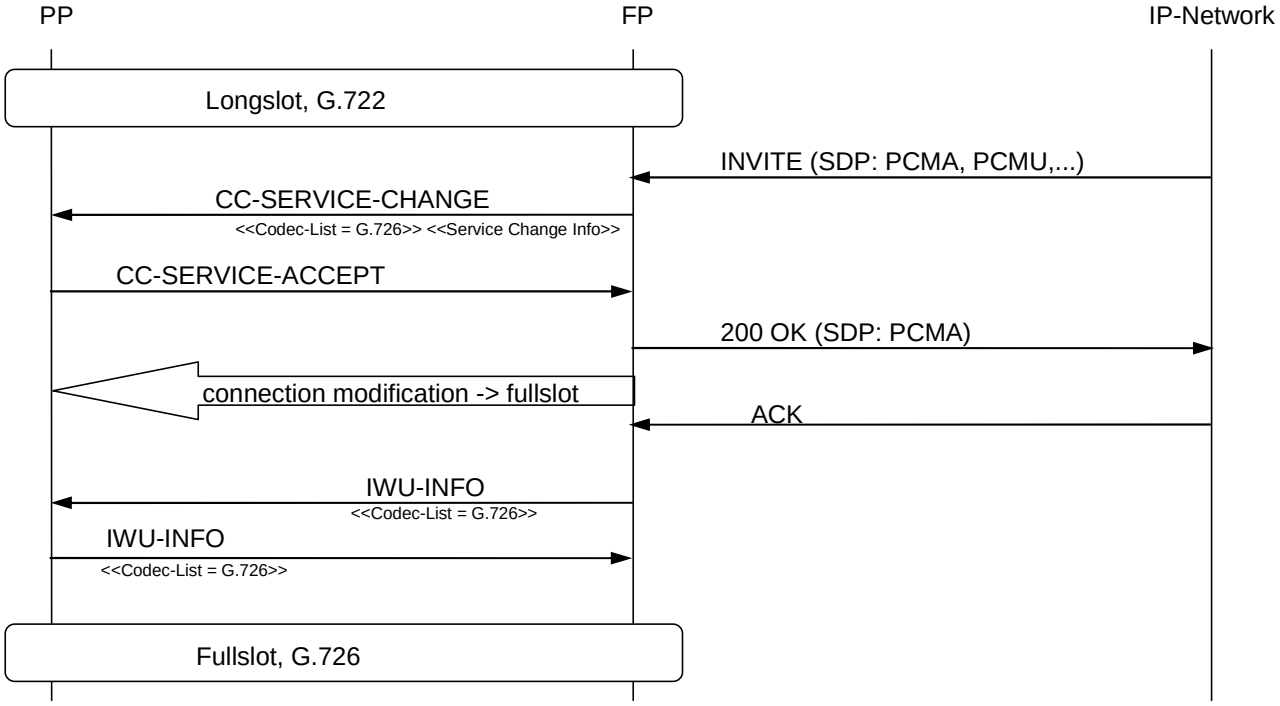

Figure D.10: Service Change from Wideband to Narrowband; re-negotiation initiated from IP-Network

The peer side can either accept the proposal by answering CC-SERVICE-ACCEPT or reject it with CC-SERVICE-REJECT. In the latter case there will be no changes. In the first case the side indicated as master in the IE <<Service change info>> will initiate the agreed changes.

Table D.3: Values used in the {CC-SERVICE-CHANGE} message

Information element	Information Element Coding	Field within the information element	Standard values within the field/information element	Normative action/comment
<<Codec-List>>	7c 04 90 02 00 84			
		<< Negotiation indicator>>		
		<<2nd codec identifier>>	0000010	ITU-T Recommendation G.726 [15]
		<< MAC service >>	0000	In_ min_delay
		<< C-Plane routing >>	000	Cf never
		<< Slot size >>	0100	Full slot
<< Service Change Info >>	16 01 9d	< coding standard >	00	Dect
		< Master >	1	Receiving side (always PP)
		< Change mode >	1101	Audio codec change

Table D.4: Values used within both {IWU-INFO} messages

Information element	Information Element Coding	Field within the information element	Standard values within the field/information element	Normative action/comment
<<Codec-List>>	7c 04 90 02 00 84			
		<< Negotiation indicator>>		
		<<2nd codec identifier>>	0000010	ITU-T Recommendation G.726 [15]
		<< MAC service >>	0000	In_ min_delay
		<< C-Plane routing >>	000	Cf never
		<< Slot size >>	0100	Full slot

D.1.4.2 Service Change from Wideband to Narrowband; re-negotiation initiated from FP

Use case: FP requests codec change on IP network in order to change the radio format on the DECT link (release radio resources for example).

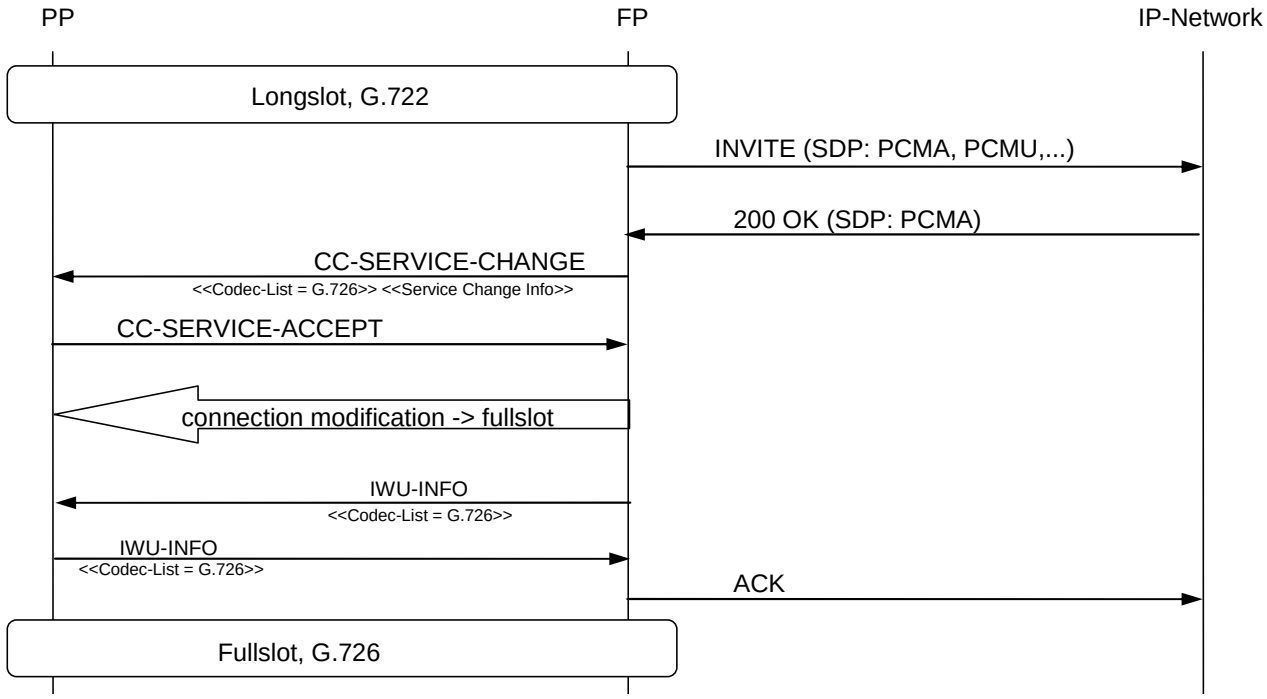

Figure D.11: Service Change from Wideband to Narrowband; re-negotiation initiated from FP

D.1.4.3 Service Change from Wideband to Narrowband; PP initiated; IP Network accepts Narrowband Codec

Use case (example): user connects a narrow-band headset on the PP during an established wideband call

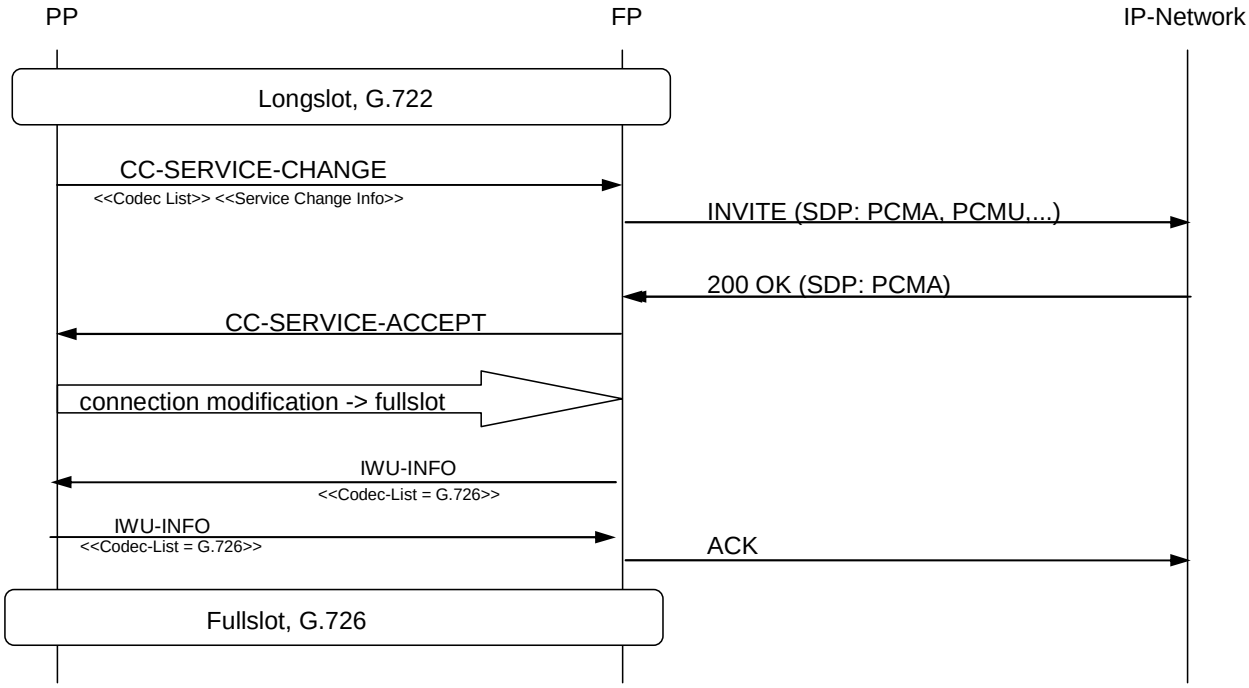

Figure D.12: Service Change from Wideband to Narrowband; PP initiated; IP Network accepts Narrowband Codec

D.1.4.4 Service Change from Wideband ITU-T Recommendation G.722 to Narrowband; PP initiated; IP Network does not accept Narrowband Codec

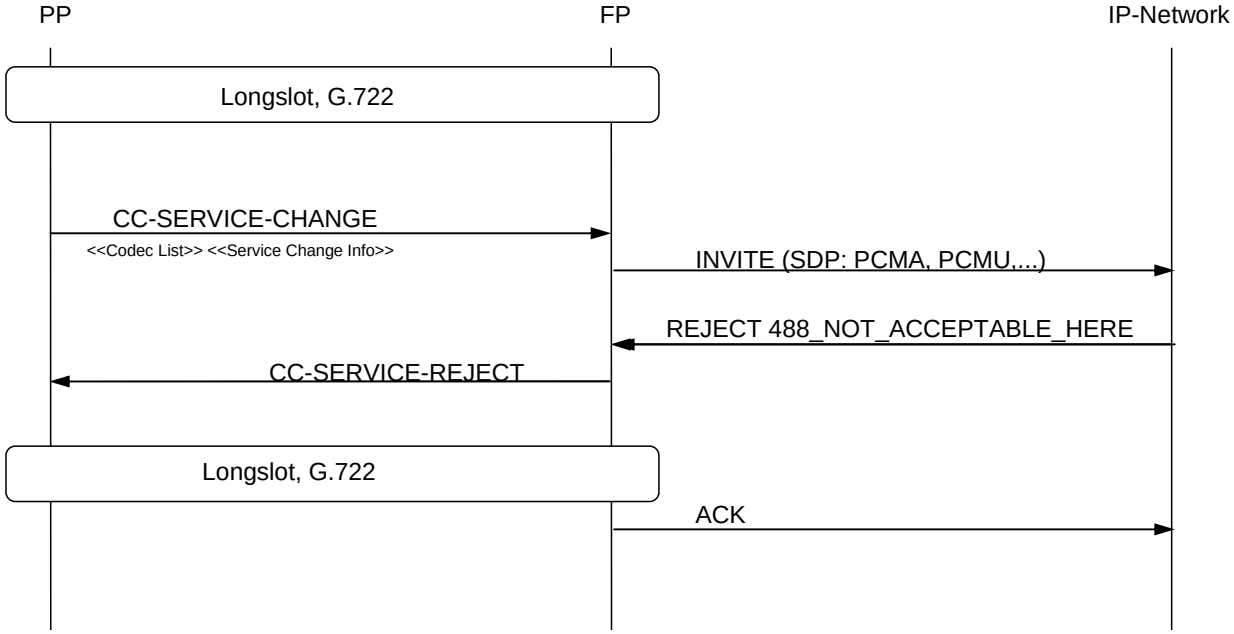

Figure D.13: Service Change from Wideband ITU-T Recommendation G.722 to Narrowband; PP initiated; IP Network does not accept Narrowband Codec

Use case: User connects a NB headset during established call on the PP, but the codec is refused by the network.

D.1.5 Internal Call

D.1.5.1 Intercom Call, PP2 confirms Wideband

Use case: user requests a wideband intercom call and is successful because the other PP supports wideband.

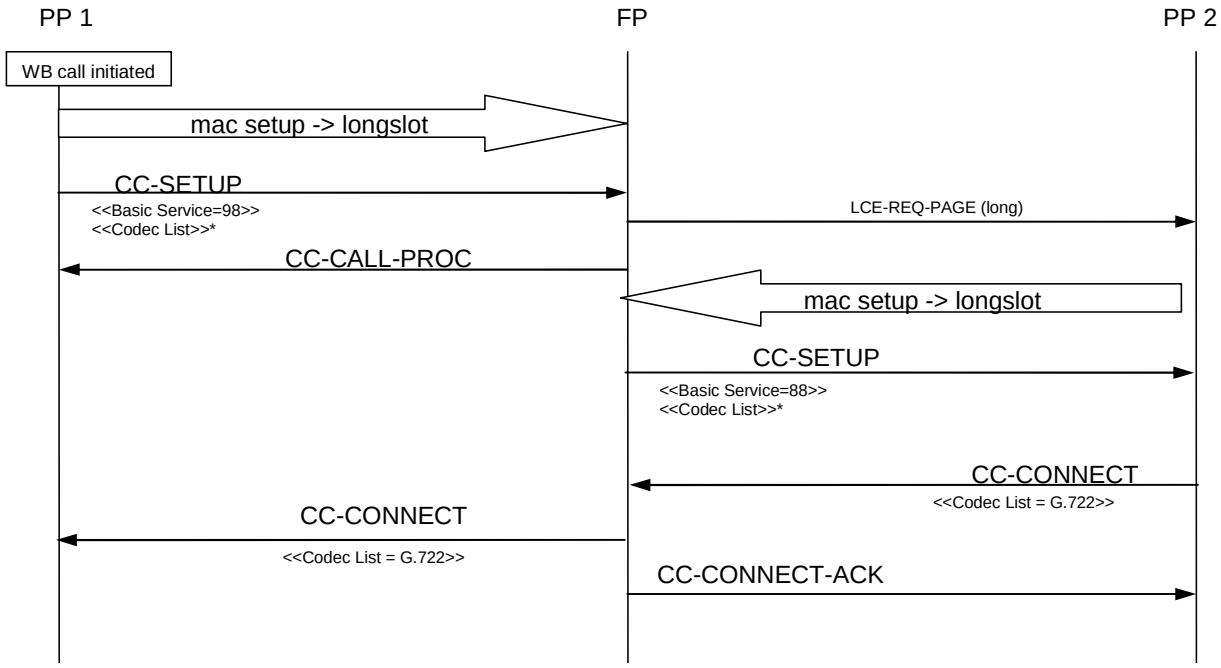

Figure D.14: Intercom Call, PP2 confirms Wideband

D.1.5.2 Intercom Call, PP2 confirms narrowband

Use case: user requests a wideband intercom call but other PP refuses wideband (narrowband headset connected to a wideband PP for example). Intercom call established in narrowband.

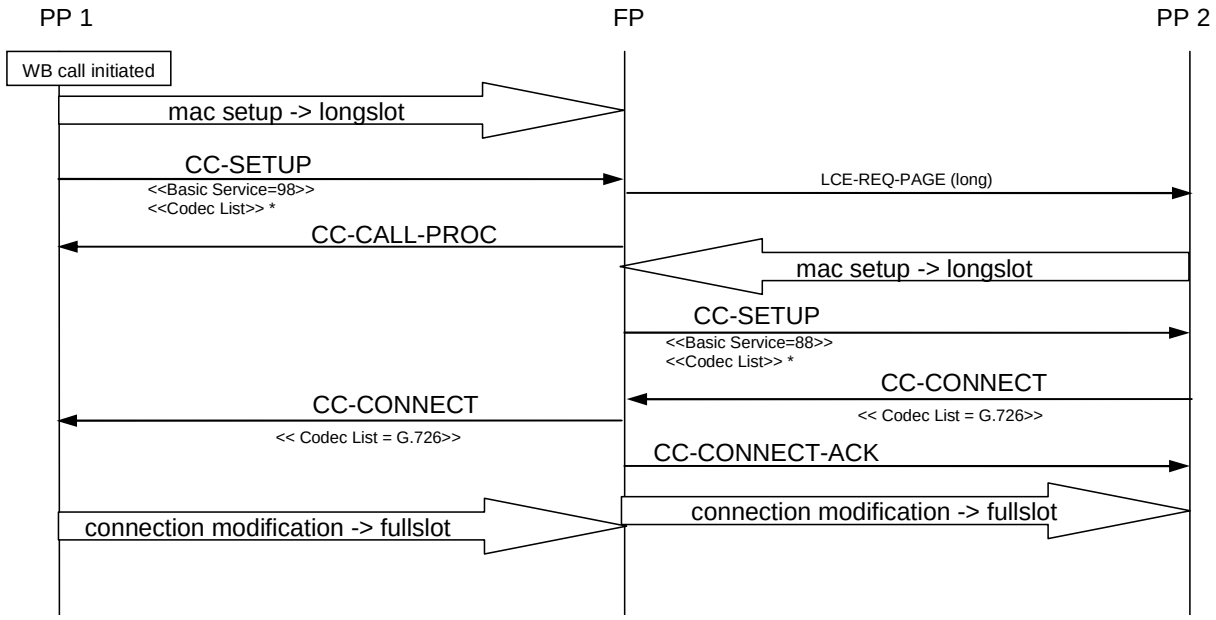

Figure D.15: Intercom Call, PP2 confirms narrowband

D.1.5.3 Intercom Call with Interworking: WB Handset -> NB Handset

Use case: User requests an intercom call between New Generation DECT PP1 to a standard DECT PP2 on the same FP.

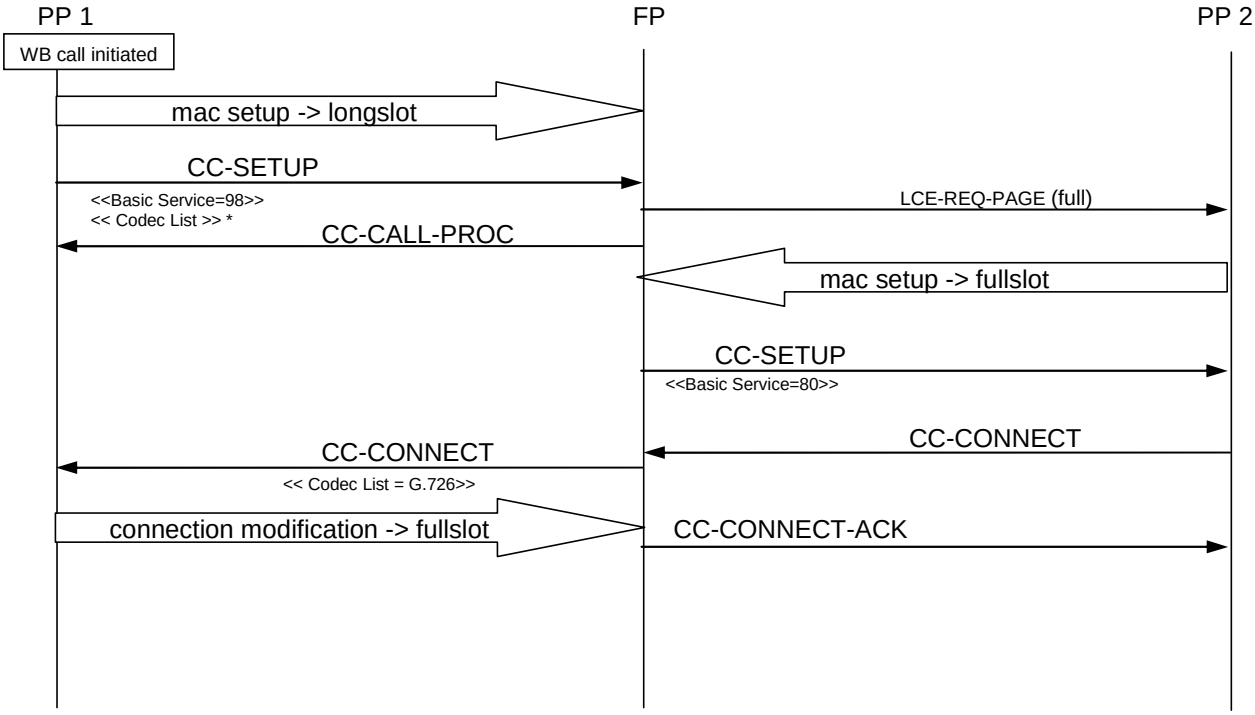

Figure D.16: Intercom Call with Interworking: WB Handset -> NB Handset

Other use case: User requests an intercom call between New Generation DECT PP1 to a standard DECT PP2 on the same FP, but FP requests to change radio format earlier than CONNECT message.

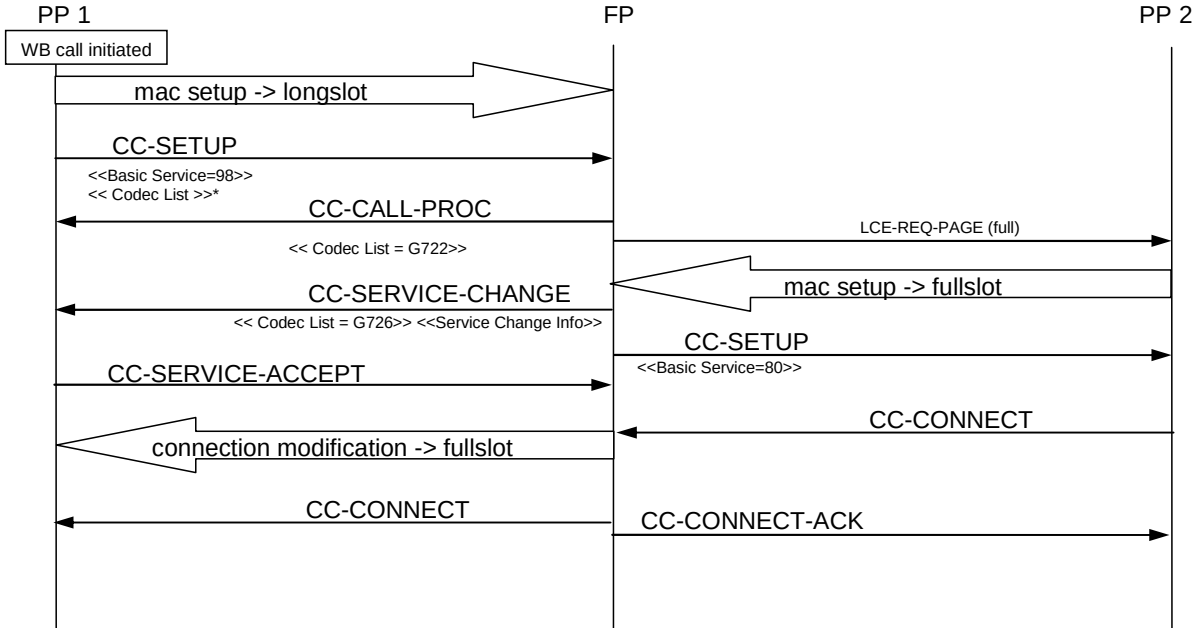

Figure D.17: Intercom Call with Interworking: WB Handset -> NB Handset, alternative procedure

D.1.5.4 Internal Call transfer, WB -> NB

Use case: New Generation DECT PP1 in communication, PP1 initiates an internal call with New Generation DECT PP2, PP2 switches to narrowband due to narrowband headset use.

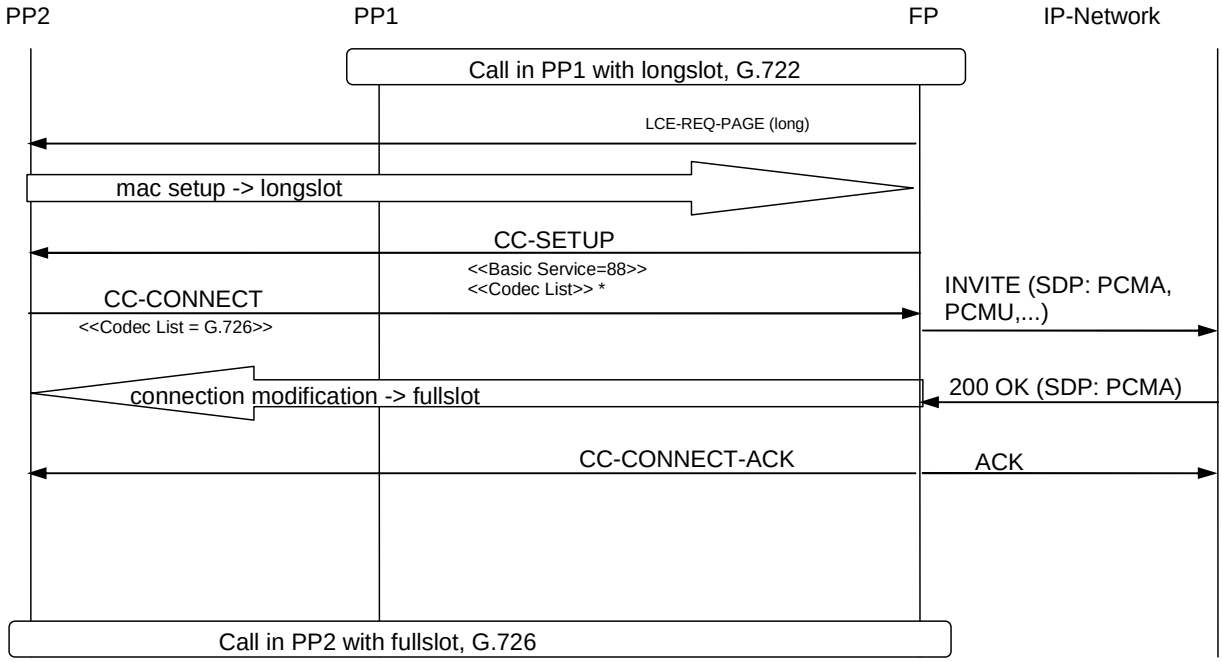

Figure D.18: Internal Call transfer between two NG PPs, initiated as WB and switched later to NB

Other use case: New Generation DECT PP1 in communication, PP1 initiates an internal call with standard DECT PP2, call established in narrowband.

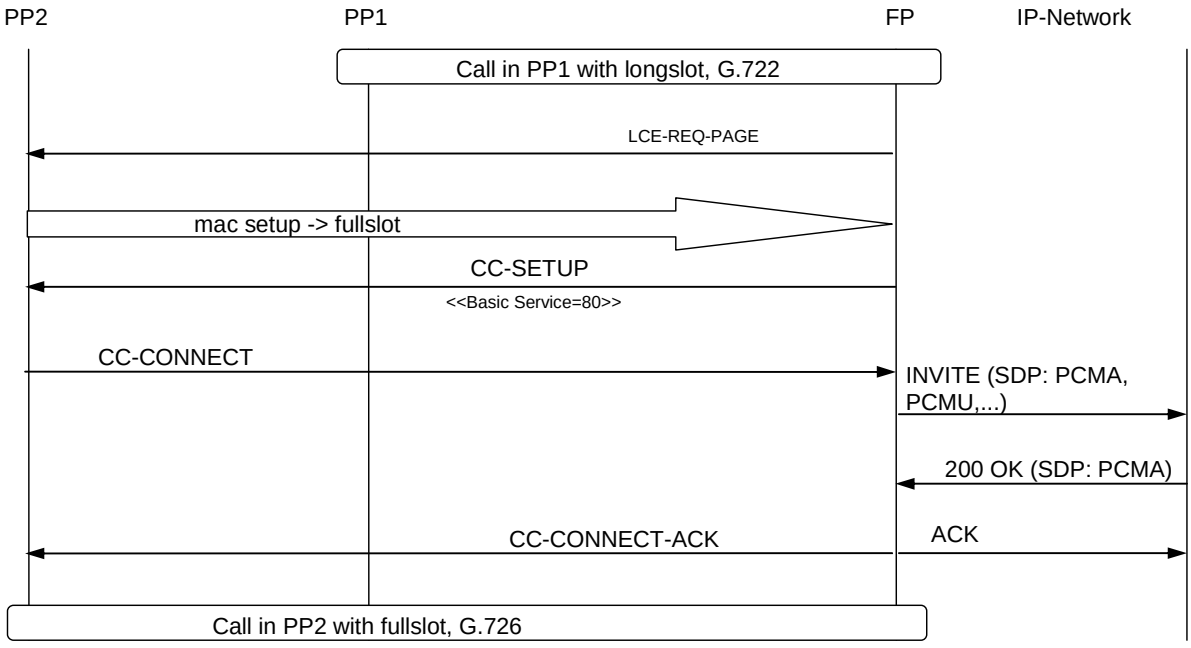

Figure D.19: Internal Call transfer from a NG PP to a PP which does not support wideband

D.1.5.5 Internal Call transfer, NB -> WB

Use case: New Generation DECT PP1 transfers a call to a New Generation DECT PP2.

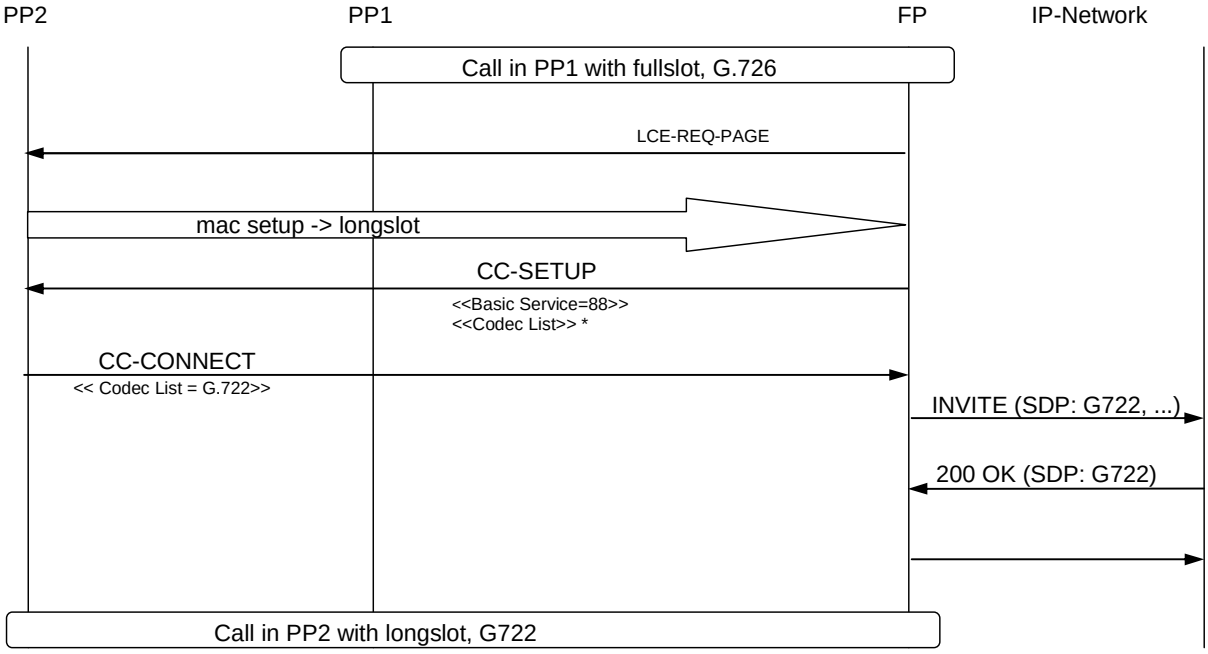

Figure D.20: Internal Call transfer, NB -> WB

D.1.5.6 Internal Call transfer, NB -> WB, IP negotiation results in NB

Use case: New Generation DECT PP1 transfers a narrowband external call to New Generation DECT PP2, requests on IP for wideband is refused by the network. External call is transferred in the same codec: narrowband.

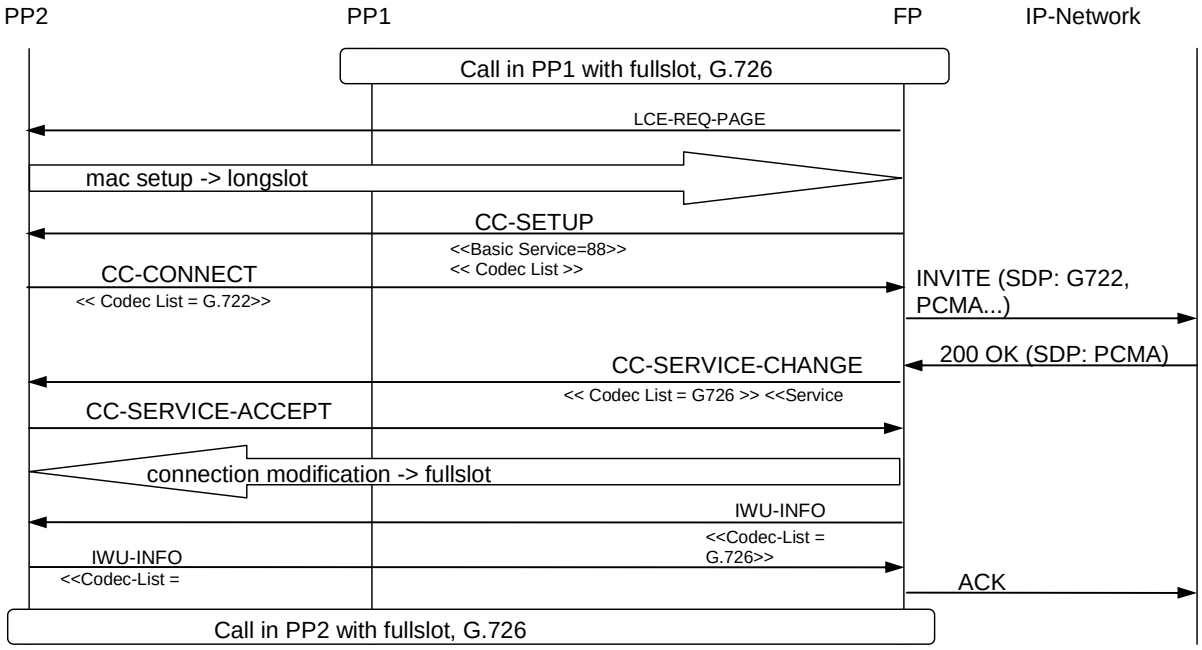

Figure D.21: Internal Call transfer, NB -> WB, IP negotiation results in NB

D.1.6 Special cases

D.1.6.1 Service Change from Wideband to Narrowband with Call Waiting

Use case: User accepts a call waiting from IP-Network.

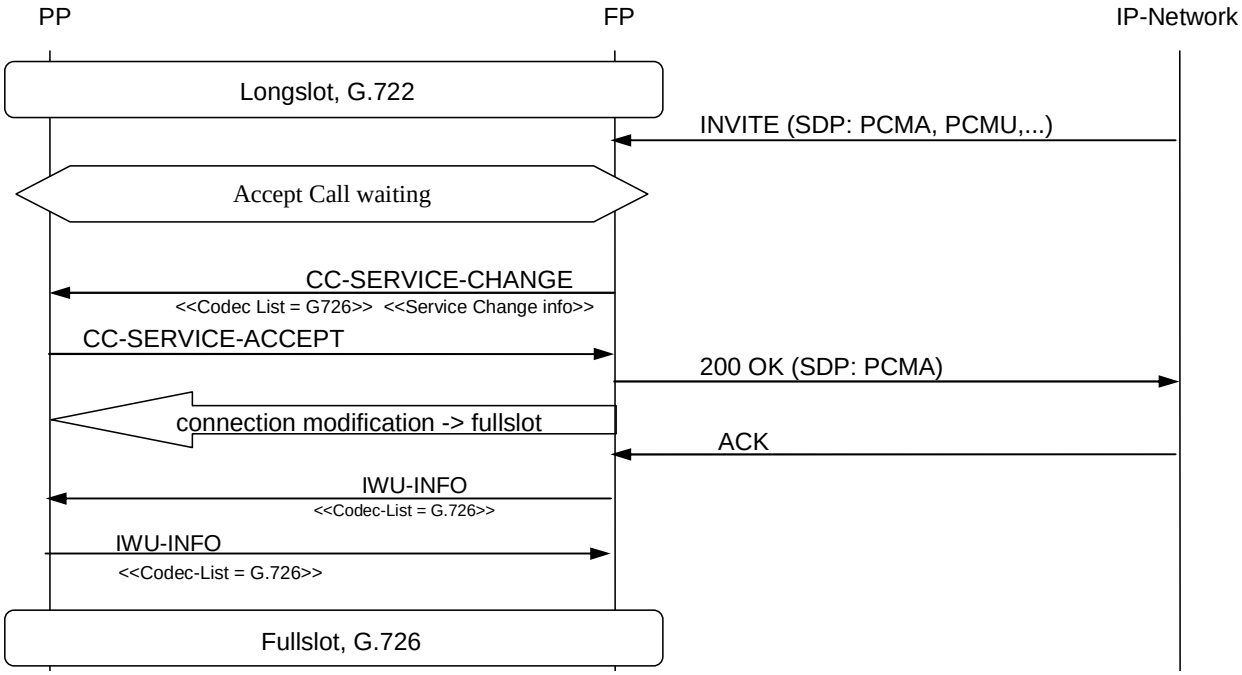

Figure D.22: Service Change from Wideband to Narrowband with Call Waiting

D.1.6.2 Service Change from Wideband to Narrowband with Call Hold

Use case: User requests a call hold during a wideband call and requests a new call setup in narrowband accepted by the IP network.

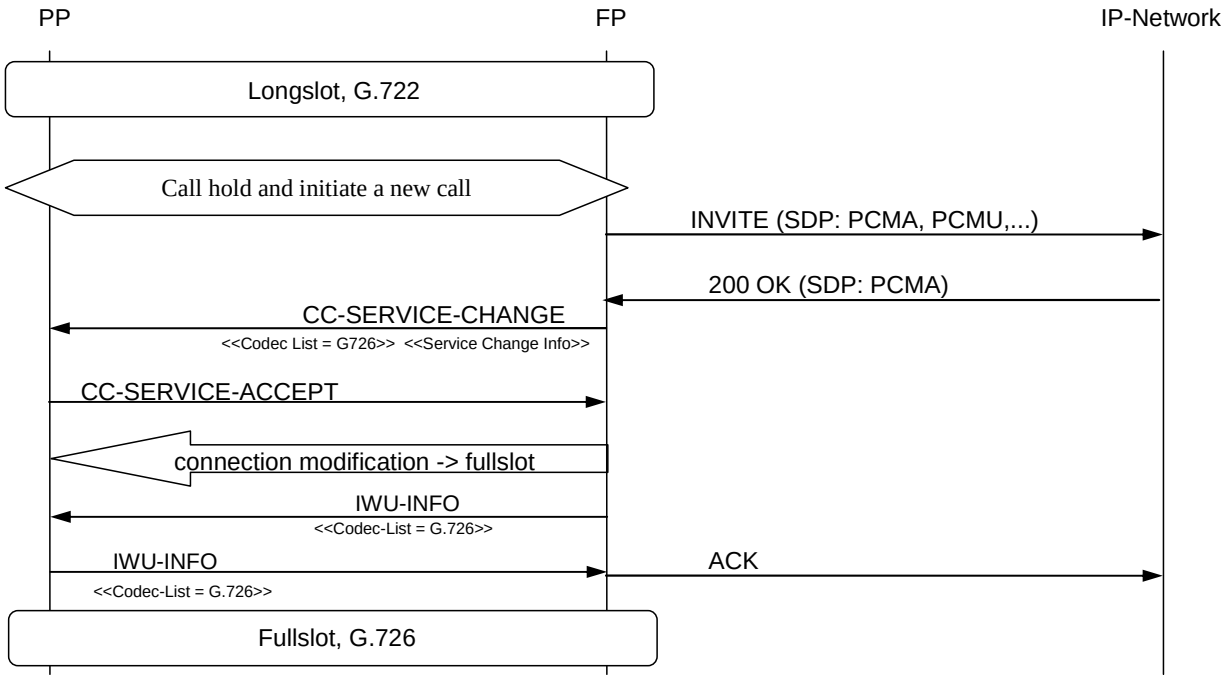

Figure D.23: Service Change from Wideband to Narrowband with Call Hold

D.1.6.3 Service Change from Wideband to Narrowband; Network layer Acknowledgment

Use case: Change codec or audio format during a communication without audio artefacts.

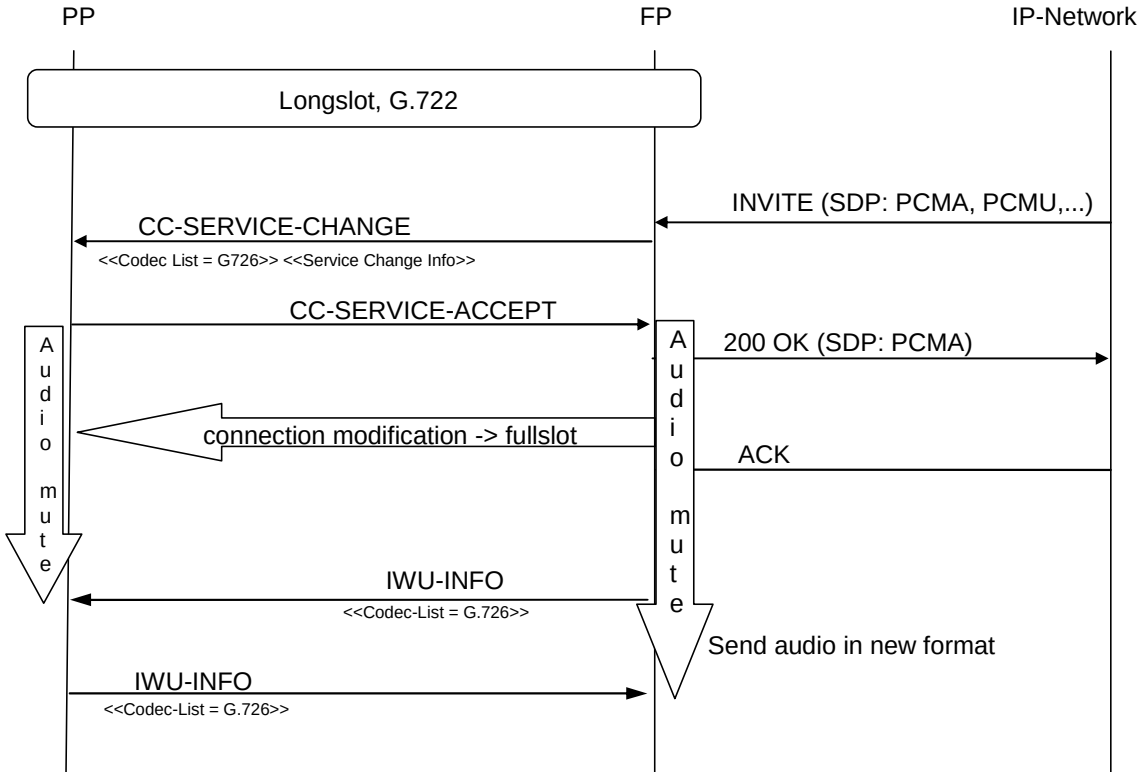

Figure D.24: Service Change from Wideband to Narrowband; Network layer Acknowledgment

The IWU-INFO is sent by both sides. The service change from Narrowband to Wideband is performed in the same way.

D.1.6.4 Service Change from Narrowband to Wideband fails; Network layer Acknowledgment

Use case: Failure in change of codec or audio format during a communication without audio artefacts.

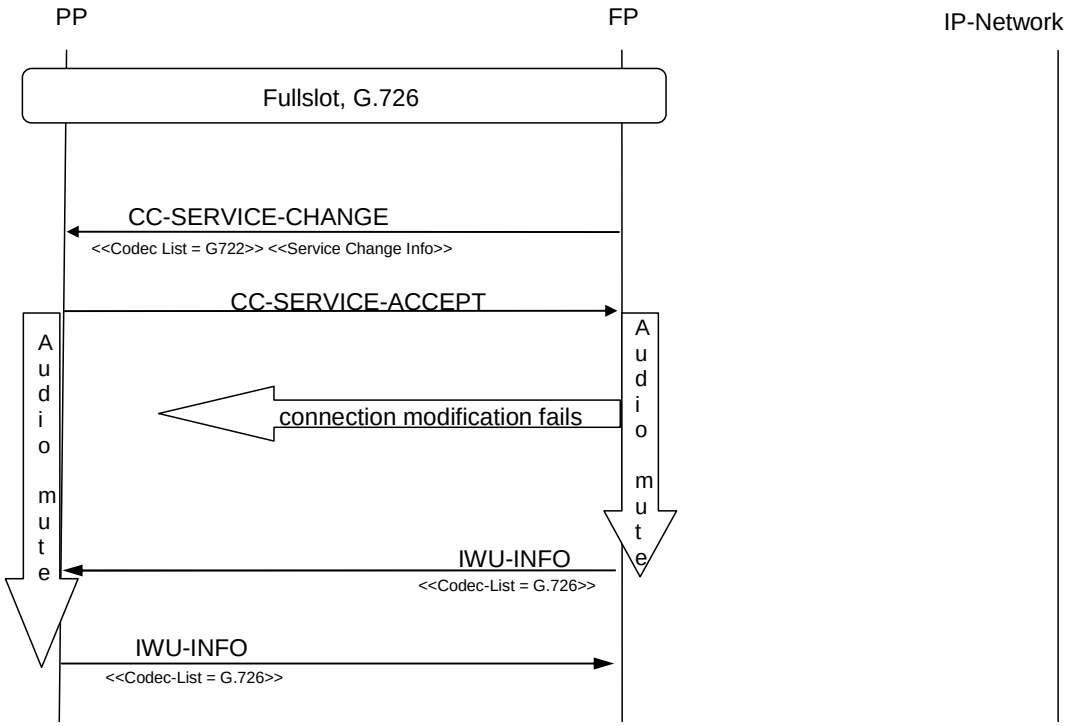

Figure D.25: Service Change from Narrowband to Wideband fails; Network layer Acknowledgment

The IWU-INFO is sent by both sides. The service change from Narrowband to Wideband is performed in the same way.

D.1.6.5 Outgoing Call, slot type modification fails

Use case: Slot type modification after negotiation fails.

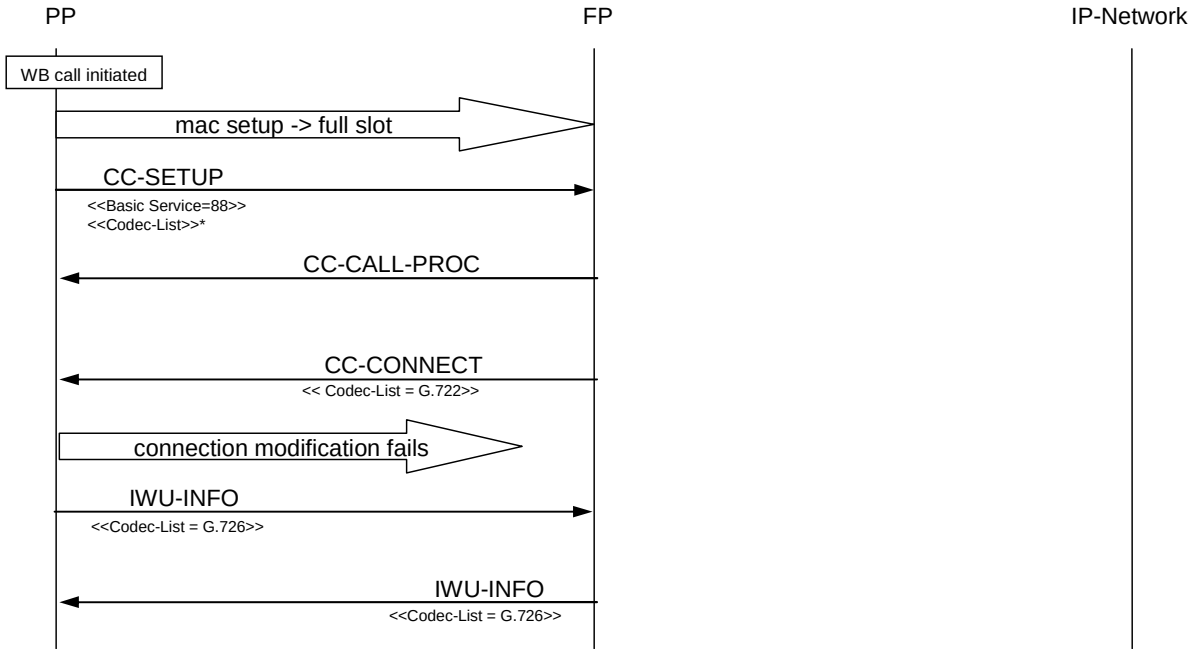

Figure D.26: Outgoing Call, slot type modification fails

D.2 Examples of implementation of procedures for MPEG-4 ER AAC-LD voice service

D.2.1 MPEG-4 ER AAC-LD voice service codec configuration and negotiation process

In annex C, the signalling of the MPEG-4 ER AAC-LD configuration using two <<IWU to IWU>> elements is defined. The following informative flowcharts describe the handling of the codec configuration and negotiation process in case of a MPEG-4 ER AAC-LD voice service selection. Furthermore the Session Initiation Protocol [27] for call establishment of the Voice over IP call is assumed.

D.2.1.1 Transmitting non default configuration using <<LOCATE-REQUEST>>, <<LOCATE-ACCEPT>> Message

Use case: Explicit in the figures title.

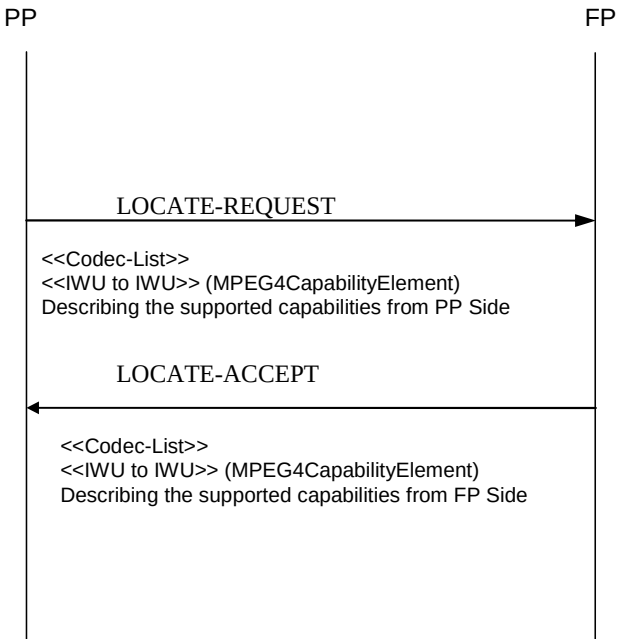

Figure D.27: Transmitting non default configuration using <<LOCATE-REQUEST>>, <<LOCATE-ACCEPT>> Message

D.2.1.2 Transmitting non default configuration using <<ACCESS-RIGHTS-REQUEST>>, << ACCESS-RIGHTS-ACCEPT>> Message

Use case: Explicit in the figures title.

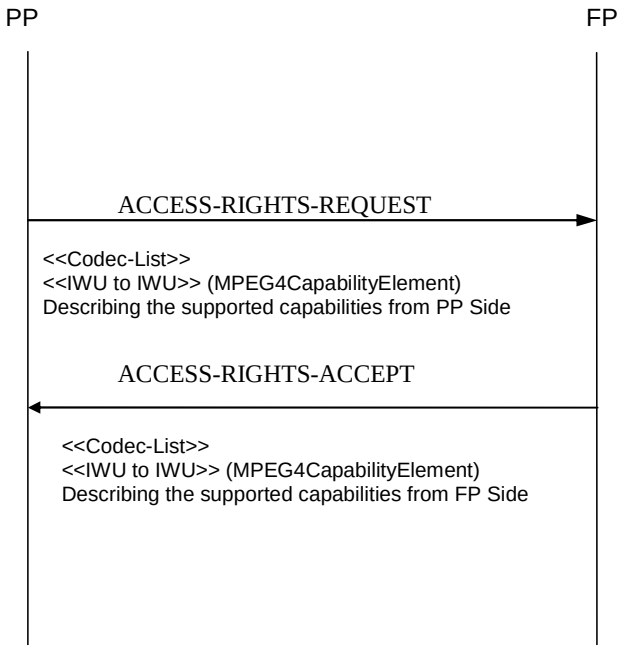

Figure D.28: Transmitting non default configuration using <<ACCESS-RIGHTS-REQUEST>>, << ACCESS-RIGHTS-ACCEPT>> Message

D.2.1.3 Outgoing Call Super Wideband, codec MPEG-4 ER AAC-LD

Use case: Explicit in the figures title.

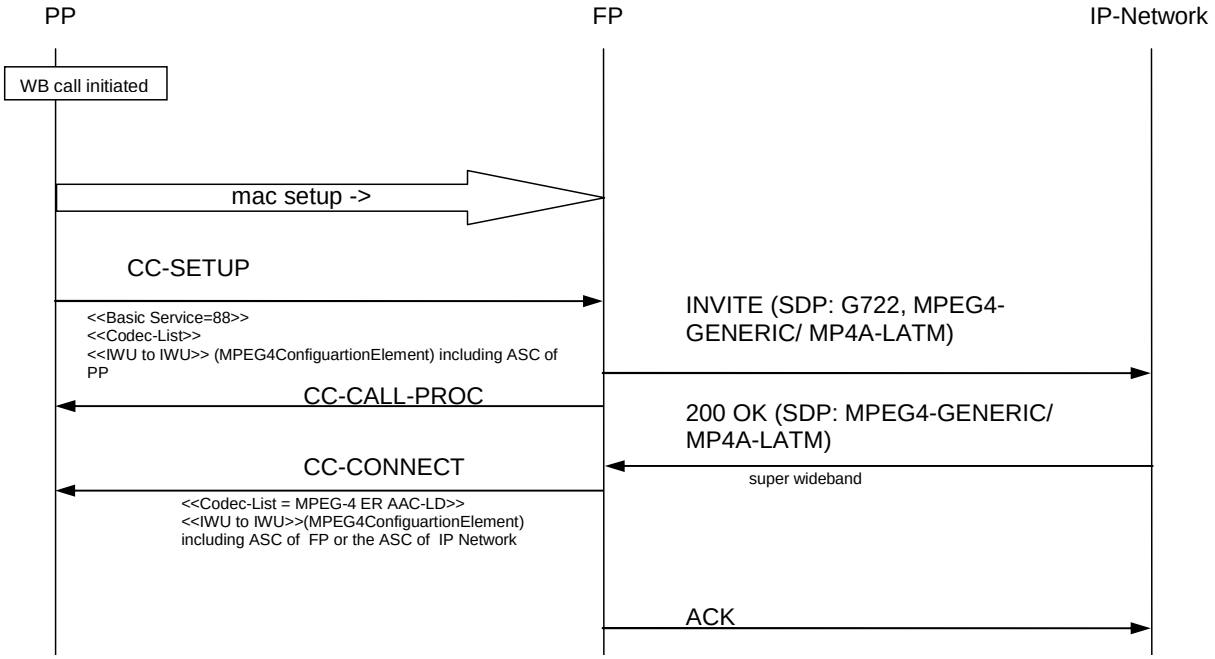

Figure D.29: Outgoing Call Super Wideband, codec MPEG-4 ER AAC-LD

D.2.1.3.1 Outgoing Call Super Wideband, INVITE command: AudioSpecificConfig()

In the MPEG4-GENERIC/MP4A-LATM part of the SDP content (during the INVITE command) the AudioSpecificConfig() (ASC) of the PP or the FP has to be transmitted to signal the used MPEG-4 ER AAC-LD configuration. The SDP content during the INVITE command can be described as follow.

G722 codec part:

```
m=audio 49230 RTP/AVP 9
a=rtpmap:9 G722/8000
```

The description for MPEG-4 ER AAC-LD can be transmitted using two different RFCs (3016 and 3640):

RFC 3640 [22]:

```
m=audio 49230 RTP/AVP 96
a=rtpmap:96 mpeg4-generic/48000/1
a=fmtp:96 streamtype=5; profile-level-id=52; mode=AAC-hbr;
config=ASC; sizeLength=13; indexLength=3; indexDeltaLength=3;
constantDuration=480;
```

RFC 3016 [23]:

```
m=audio 49230 RTP/AVP 96
a=rtpmap:96 MP4A-LATM/48000
a=fmtp:96 profile-level-id=52; bitrate=64000; cpresent=0;
config=ASC;
```

D.2.1.3.2 Outgoing Call Super Wideband, OK command: AudioSpecificConfig()

In the MPEG4-GENERIC/MP4A-LATM part of the SDP content (during the OK command) the AudioSpecificConfig() (ASC) of the IP remote station has to be transmitted to signal the used MPEG-4 ER AAC-LD configuration. The SDP content during the OK command can be described as follow.

RFC 3640 [22]:

```
m=audio 49230 RTP/AVP 96

a=rtpmap:96 mpeg4-generic/48000/1

a=fmtp:96 streamtype=5; profile-level-id=52; mode=AAC-hbr;

config=ASC; sizeLength=13; indexLength=3; indexDeltaLength=3;
constantDuration=480;
```

RFC 3016 [23]:

```
m=audio 49230 RTP/AVP 96

a=rtpmap:96 MP4A-LATM/48000

a=fmtp:96 profile-level-id=52; bitrate=64000; cpresent=0;

config=ASC;
```

D.2.1.4 Incoming Call Super Wideband, codec MPEG-4 ER AAC-LD

Use case: Explicit in the figures title.

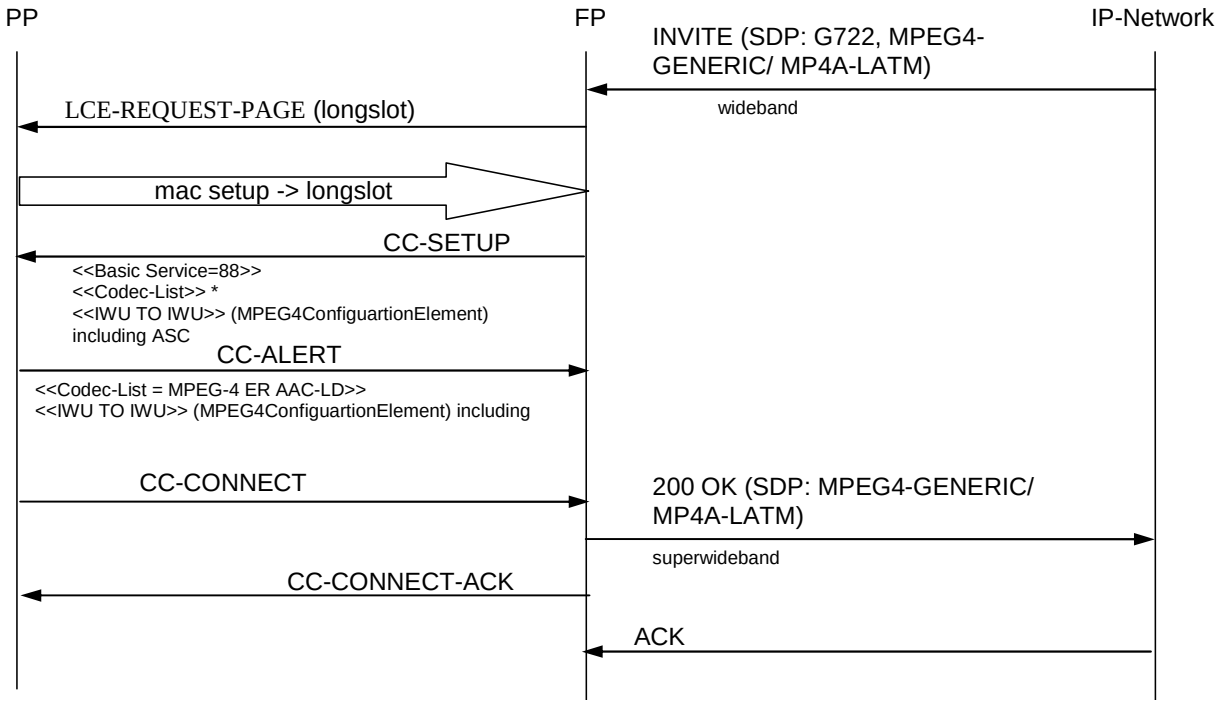

Figure D.30: Incoming Call Super Wideband, codec MPEG-4 ER AAC-LD

D.2.1.4.1 Incoming Call Super Wideband, INVITE command: AudioSpecificConfig()

In the MPEG4-GENERIC/MP4A-LATM part of the SDP content (during the INVITE command), the `AudioSpecificConfig()` (ASC) of the IP remote station or the FP has to be transmitted to signal the used MPEG-4 ER AAC-LD configuration. The SDP content during the INVITE command can be described as follow.

ITU-T Recommendation G.722 [17] codec part:

```
m=audio 49230 RTP/AVP 9
a=rtpmap:9 G722/8000
```

The description for MPEG-4 ER AAC-LD can be transmitted using two different RFCs (3016 and 3640):

RFC 3640[22]:

```
m=audio 49230 RTP/AVP 96
a=rtpmap:96 mpeg4-generic/48000/1
a=fmtp:96 streamtype=5; profile-level-id=52; mode=AAC-hbr;
config=ASC; sizeLength=13; indexLength=3; indexDeltaLength=3;
constantDuration=480;
```

RFC 3016 [23]:

```
m=audio 49230 RTP/AVP 96
a=rtpmap:96 MP4A-LATM/48000
a=fmtp:96 profile-level-id=52; bitrate=64000; cpresent=0;
config=ASC;
```

D.2.1.4.2 Incoming Call Super Wideband, OK command: AudioSpecificConfig()

In the MPEG4-GENERIC/MP4A-LATM part of the SDP content (during the OK command) the `AudioSpecificConfig()` (ASC) of the PP has to be transmitted to signal the used MPEG-4 ER AAC-LD configuration. The SDP content during the OK command can be described as follows:

RFC 3640 [22]:

```
m=audio 49230 RTP/AVP 96

a=rtpmap:96 mpeg4-generic/48000/1

a=fmtp:96 streamtype=5; profile-level-id=52; mode=AAC-hbr;

config=ASC; sizeLength=13; indexLength=3; indexDeltaLength=3;
constantDuration=480;
```

RFC 3016 [23]:

```
m=audio 49230 RTP/AVP 96

a=rtpmap:96 MP4A-LATM/48000

a=fmtp:96 profile-level-id=52; bitrate=64000; cpresent=0;

config=ASC;
```

Annex E (normative): Frame formats

E.1 Transport of the ITU-T Recommendation G.729.1 audio frame in full-slot mode

This clause specifies the format of the ITU-T Recommendation G.729.1 [18] audio frame in the B field. This format shall be used in **both** directions (from PT to FT and from FT to PT).

The same format may be re-used for any frame oriented codec using a bit rate up to 30,4 kbit/s.

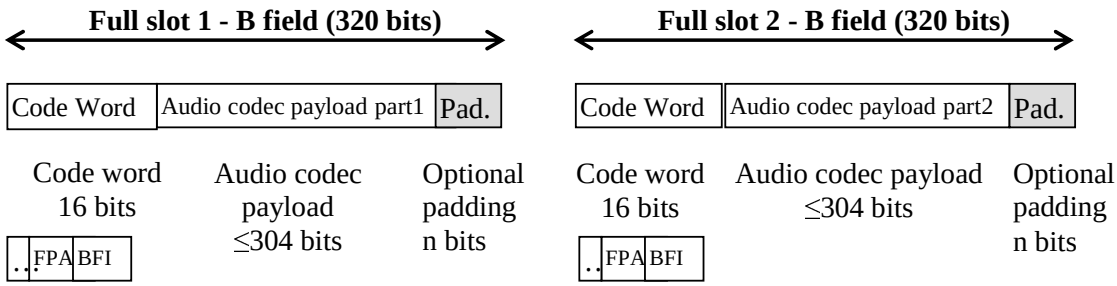

Figure E.1: Transport of framed audio codec payload in full slot mode

The code word is generic for any framed codec. The audio codec payload format is codec dependent.

Audio codec payload part 1 and 2 are simply concatenated to obtain the complete audio payload format described in 2) below.

1) The code word is coded as follows:

Bit:	8	7	6	5	4	3	2	1	Octet
	PA	X	X	X	X	FPA2	FPA1	BFI	1
	X	X	X	X	X	X	X	X	2

Figure E.2: Code word coding

Bad frame indicator (bit 1):

This bit indicates whether the current audio frame is valid or not. It may be set to 1 if an IP packet loss is detected in the FP, or if any other transmission problem has occurred which damages the integrity of the frame. It is not recommended to use it in the PP to FP direction.

Bit	**1**	**Meaning**
	0	The audio frame is valid
	1	The audio frame is not valid (packet loss detected in the network above the FP)

Frame Part (FPA1, FPA2):

Bits	**3 2**	**Meaning**
	0 0	First audio frame part
	0 1	Second audio frame part
	1 0	Third audio frame part (not used for ITU-T Recommendation G.729.1)
	1 1	Fourth audio frame part (not used for ITU-T Recommendation G.729.1)

Parity bit (PA):

Bit 8

x Even parity. Total number of "1" (including this bit) must be even in the first octet.

Other bits (octet 1 bits 4 to 8, octet 2 bits 1 to 8):

For future use. For example, additional protection of payload bits (redundancy, CRC etc.).

2) The audio payload format coding is codec-dependent:

Case 1: ITU-T Recommendation G.729.1 [18] payload format

The same coding is used as in RFC 4749 [24].

ITU-T Recommendation G.729.1 [18] is used at the maximum bit rate of 30 kbit/s in order to fit into the B field size (≤30,4 kbit/s). The audio codec frame is defined as in RFC:

G.729.1 payload format : 608 bits max (case 30 kbits/s)

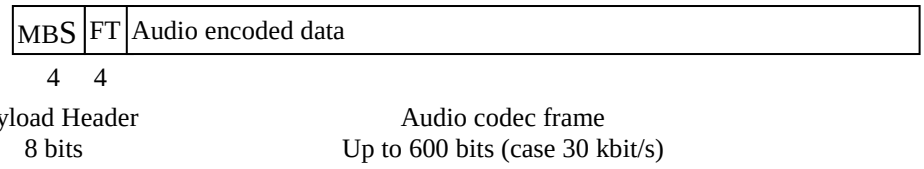

Figure E.3: G.729.1 payload format

Coding of the Payload header:

Bit:	8	7	6	5	4	3	2	1
	MBS4	MBS3	MBS2	MBS1	FT4	FT3	FT2	FT1

Figure E.4: Payload header coding

FT field (4 bits):

Frame type of the audio encoded data. The value of the FT field is set according to the following table:

FT4 to FT1:

```
0000  |  0 |   8 kbit/s | 20 octets |
0001  |  1 |  12 kbit/s | 30 octets |
0010  |  2 |  14 kbit/s | 35 octets |
0011  |  3 |  16 kbit/s | 40 octets |
0010  |  4 |  18 kbit/s | 45 octets |
0101  |  5 |  20 kbit/s | 50 octets |
0110  |  6 |  22 kbit/s | 55 octets |
0111  |  7 |  24 kbit/s | 60 octets |
0100  |  8 |  26 kbit/s | 65 octets |
1001  |  9 |  28 kbit/s | 70 octets |
1010  | 10 |  30 kbit/s | 75 octets |(recommended default value for G.729.1)
1011  | 11 |  32 kbit/s | 80 octets |(value not possible for G.729.1 on DECT
                                        link in full-slot mode)
1100  | 12 | (reserved) |
1101  | 13 | (reserved) |
1110  | 14 | (reserved) |
1111  | 15 |  NO_DATA   |     0     |
```

The FT value 15 (NO_DATA) indicates that there is no audio data in the payload. This MAY be used to update the MBS value when there is no audio frame to transmit. The payload will then be reduced to the payload header.

If a payload with a reserved FT value is received, the whole payload MUST be ignored.

MBS field (4 bits) = Maximum bit rate supported.

It indicates a maximum bit rate to the encoder at the site of the receiver of this payload. The MBS is used to tell the other party the maximum bit rate one can receive. The encoder MUST NOT exceed the sending rate indicated by the received MBS.

NOTE: Due to the embedded property of the coding scheme, the encoder can send frames at the MBS rate or any lower rate. As long as it does not exceed the MBS, the encoder can change its bit rate at any time without previous notice.

The value of the MBS field is set according to the following table:

MBS 4 to MBS1

0000	0	8 kbit/s	
0001	1	12 kbit/s	
0010	2	14 kbit/s	
0011	3	16 kbit/s	
0100	4	18 kbit/s	
0101	5	20 kbit/s	
0110	6	22 kbit/s	
0111	7	24 kbit/s	
1000	8	26 kbit/s	
1001	9	28 kbit/s	
1010	10	30 kbit/s	(recommended default value for G.729.1)
1011	11	32 kbit/s	(value not possible for G.729.1 on DECT link in full-slot mode)
1100	12	(reserved)	
1101	13	(reserved)	
1110	14	(reserved)	
1111	15	NO_MBS	

So the default normative recommended value for MBS is 10 and for FT is 10, for direction PP to FP. For the direction FP to PP of course MBS and FT are dependent of what is really received from the distant but MUST NOT be set to 11 which is a non authorized value on DECT link.

For FT (Frame Type) below 10 (bit rate below 30 kbit/s) padding are used as necessary on B field part.

Example 1: with a FT of 7 (24 kbit/s) that corresponds to ITU-T Recommendation G.729.1 [18] payload format frame of 8 (header) + 480 (audio frame) = 488 bits. Full slot 1 is then composed of a code word of value 0x0100 followed by the first 304 bits of ITU-T Recommendation G.729.1 [18] payload format and Full slot 2 is composed of a code word of value 0x0300 followed by last 184 bits ITU-T Recommendation G.729.1 [18] payload format, followed by 120 bits at zero (the padding bits).

Example 2: with a FT of 1 (12 kbit/s) so corresponding to a complete ITU-T Recommendation G.729.1 [18] payload format frame of 8 (header) + 240 (audio frame) = 248 bits. Full slot 1 is then composed of a code word of value 0x0100 followed by the full 248 bits of ITU-T Recommendation G.729.1 [18] payload format, followed by 56 bits at zero (padding bits) and Full slot 2 is composed of a code word of value 0x0300 followed by 304 bits at zero (only padding bits).

In case of BFI frame (indicated with bfi = 1 in "code Word" part), the ITU-T Recommendation G.729.1 [18] payload format should be adapted with a FT of 15 (NO_DATA), a MBS of 15 (NO MBS) and a complete frame of 0 (only padding).

Case 2: other codec payload format

To be defined for other codecs.

3) Optional padding:

0 to n bits. Not used for ITU-T Recommendation G.729.1 [18] at 30 kbit/s.

Annex F (normative): Amendments to other DECT specifications

F.1 Amendments to EN 300 444 (Generic Access Profile (GAP))

The following amendments to EN 300 444 [13] shall apply for the purpose of the present document.

F.1.1 Calling Line Identification Presentation (CLIP)

F.1.1.1 CLIP procedure description (add to clause 8 of EN 300 444)

A new clause with the following text shall be added to clause 8 of EN 300 444 [13]:

8.x Calling Line Identification Presentation (CLIP) Indication

The following text together with the associated clauses define the mandatory requirements with regard to the present document.

Calling Line Presentation Indication may be sent either by including the <<CALLING-PARTY-NUMBER>> information element in the {CC-SETUP} message or in a {CC-INFO} message. The FT is required to support one of the methods, the PT is required to support both methods.

For CLIP indication through the {CC-SETUP} see table 21, Values used within the {CC-SETUP} message.

For CLIP indication through {CC-INFO} consider the following.

Table x: Values used within the {CC-INFO} message

Information element	Field within the information element	Standard values within the field/information element	Normative action/comment
<<Calling party number>>			
	<Number type>	All	
	<Numbering plan id>	All	
	<Presentation indicator>	All	
	<Screening indicator>	All	
	<Calling party address>	All	

NOTE 1: To support the feature in the PP, it is sufficient that the PP is capable to display the IA5 characters given in the field <Calling party address> according to its display capabilities without consideration of the contents of octets 3 and 3a.

NOTE 2: In case both CLIP and CNIP are sent to the PP, it is sufficient to display CNIP. It is optional to display both.

F.1.2 Calling Name Identification Presentation (CNIP)

F.1.2.1 CNIP definition (add to clause 4.1 of EN 300 444)

The following definition shall be added to clause 4.1 "Network (NWK) features" of EN 300 444 [13]:

Calling Name Identification Presentation (CNIP) [N.34]: ability to provide the calling party name to the called party before accepting the call

F.1.2.2 CNIP NWK feature (add to clause 6.2 of EN 300 444)

The following entry shall be added to table 1 in clause 6.2 of EN 300 444 [13]:

Table 1: NWK features status

Feature supported					
			Status		
Item no.	Name of feature	Reference	PT	FT	
				R/B	P
N.34	Calling Name Identification Presentation (CNIP)	4.1	O	O	O

F.1.2.3 CNIP NWK feature to procedure mapping (add to clause 6.7 of EN 300 444)

The following entry shall be added to table 5 in clause 6.7 of EN 300 444 [13]:

Table 5: NWK feature to procedure mapping

Feature/Procedure mapping					
			Status		
Feature	Procedure	Reference	PT	FT	
				R/B	P
N.34 Calling Name Identification Presentation (CNIP)		4.1	O	O	O
	Calling Name Identification Presentation (CNIP) Indication	8.x	M	M	M

F.1.2.4 CNIP procedure description (add to clause 8 of EN 300 444)

A new clause with the following text shall be added to clause 8 of EN 300 444 [13]:

8.x Calling Name Identification Presentation (CNIP) Indication

The following text together with the associated clauses define the mandatory requirements with regard to the present document.

Calling Name Presentation Indication may be sent either by including the <<CALLING-PARTY-NAME>> information element in the {CC-SETUP} message or in a {CC-INFO} message. FT is required to support at least one of the methods, PT is required to support both.

For CNIP indication through the {CC-SETUP} see table 21, with the following additions:

Table x: Values added within the {CC-SETUP} message

Information element	Field within the information element	Standard values within the field/information element	Normative action/comment
<<Calling party name>>			
	<Presentation indicator>	All	
	<Screening indicator>	All	
	<Calling party name>	All	

For CNIP indication through {CC-INFO} consider the following:

Table x: Values used within the {CC-INFO} message

Information element	Field within the information element	Standard values within the field/information element	Normative action/comment
<<Calling party name>>			
	<Presentation indicator>	All	
	<Screening indicator>	All	
	<Calling party name>	All	

NOTE 1: To support the feature in the PP, it is sufficient that the PP is capable to display the IA5 characters given in the field <Calling party name> according to its display capabilities without consideration of the contents of octet 3.

NOTE 2: In case both CLIP and CNIP are sent to the PP, it is sufficient to display CNIP. It is optional to display both.

F.1.3 Internal Call CLIP and CNIP

F.1.3.1 Internal Call NWK feature to procedure mapping (modify clause 6.7 of EN 300 444)

The entry N.31 "Internal Call" in table 5, clause 6.7 of EN 300 444 [13] shall be modified as follows:

Table 5: NWK feature to procedure mapping

Feature/Procedure mapping					
			Status		
Feature	Procedure	Reference	PT	FT	
				R/B	P
N.31 Internal Call		4.1	O	O	O
	Internal call setup	8.18	M	M	M
	Internal call keypad	8.19	M	O	O
	Internal call CLIP	8.x (F.1.3.2)	O	O	O
	Internal call CNIP	8.x (F.1.3.3)	O	O	O

F.1.3.2 Internal Call CLIP procedure description (add to clause 8 of EN 300 444)

A new clause with the following text shall be added to clause 8 of EN 300 444 [13]:

8.x Internal Call Calling Line Identification Presentation (CLIP)

The following text together with the associated clauses define the mandatory requirements with regard to the present document.

Calling Line Identification Presentation (CLIP), shall be implemented for internal calls.

The general procedure for CLIP is described in clause 8.xx (F.1.1) "Calling Line Identification Presentation (CLIP) Indication" and it shall be used also for internal calls.

NOTE 1: The internal call CLIP indication can be used to convey internal handset number or any other number. External calling number could be used in case of call transfer.

NOTE 2: If internal call CLIP indication is used to indicate the handset number the following values for information element <<Calling Party Number>> should be used:

Table x: Suggested values for <<Calling Party Number>> IE for internal calls

Information element	Field within the information element	Standard values within the field/information element	Normative action/comment
<<Calling party number>>			
	<Number type>	3	Network specific number
	<Numbering plan id>	9	Private plan
	<Presentation indicator>	All	Presentation allowed
	<Screening indicator>	All	User-provided, verified and passed
	<Calling party address>	IA5 coding of Terminal Identity Number in decimal representation	Terminal Identity Number of the calling part - 0 for FP - 1 to n for PP

NOTE 3: See clause 14.4 for a description of the Terminal Identity Number and its use.

NOTE 4: To support the feature in the PP, it is sufficient that the PP is capable to display the IA5 characters given in the field <Calling party address> according to its display capabilities without consideration of the contents of octets 3 and 3a.

NOTE 5: In case both CLIP and CNIP are sent to the PP, it is sufficient to display CNIP. It is optional to display both.

F.1.3.3 Internal Call CNIP procedure description (add to clause 8 of EN 300 444)

A new clause with the following text shall be added to clause 8 of EN 300 444 [13]:

8.x Internal Call Calling Name Identification Presentation (CNIP)

The following text together with the associated clauses define the mandatory requirements with regard to the present document.

Calling Name Identification Presentation (CNIP), shall be implemented for internal calls.

The general procedure for CNIP is described in clause 8.x "Calling Name Identification Presentation (CNIP) Indication" (F.1.2.4) and it shall be used also for internal calls.

NOTE 1: The internal call CNIP indication can be used to convey a user-friendly identifier associated with the PT. For example, in a residential environment, this identifier could be an alias for a room ("kitchen", "living room"), a person, etc. that the PT has a strong relationship with.

NOTE 2: To support the feature in the PP, it is sufficient that the PP is capable to display the IA5 characters given in the field <Calling party name> according to its display capabilities without consideration of the contents of octet 3.

NOTE 3: In case both CLIP and CNIP are sent to the PP, it is sufficient to display CNIP. It is optional to display both.

F.1.4 Procedure to assign a simple identity number to each DECT entity in mono cell systems

F.1.4.1 Terminal identity number assignment in mono cell system definition (add to clause 4.2 of EN 300 444)

The following definition shall be added to clause 4.2 "Application features" of EN 300 444 [13]:

Terminal identity number assignment in mono cell system [A.4]: ability to assign to each PT a terminal identity number.

F.1.4.2 Terminal identity number assignment in mono cell system application feature (add to clause 6.6 of EN 300 444)

The following entry shall be added to table 4 in clause 6.6 of EN 300 444 [13]:

Table 4: Application features status

Feature supported					
			Status		
Item no.	Name of feature	Reference	PT	FT	
				R/B	P
A.4	Terminal identity number assignment in mono cell system	4.2	O	O	N/A

F.1.4.3 Terminal identity number assignment in mono cell system feature to procedure mapping (add to clause 6.8.3 of EN 300 444)

The following entry shall be added to table 8 in clause 6.8.3 of EN 300 444 [13]:

Table 8: Application feature to procedure mapping

Feature/Procedure mapping					
			Status		
Feature	Procedure	Reference	PT	FT	
				R/B	P
A.4 Terminal identity number assignment in mono cell system		4.2	O	O	N/A
	Terminal identity number assignment	14.x, (F.1.4.2)	O	O	N/A

F.1.4.4 Terminal identity number assignment in mono cell system procedure description (add to clause 14 of EN 300 444)

A new clause with the following text shall be added to clause 14 of EN 300 444 [13]:

14.x Terminal Identity number assignment in mono cell system

14.x.1 General

In a mono cell system for residential and small office applications, the terminal identity number is the number that can be:

- used by the FT to identify the subscription data related to each PT (i.e. 1 to the maximum number of PT subscribed). The subscription data includes IPUI, PARK, terminal capabilities, etc.;

- displayed by each PP in Idle Locked mode (for example, "DECT 4" if the PP is the 4th PT on the FT);
- used to select the called DECT entity (PP or FP) when initiating Internal Call (for example, "Internal call to PP number 4");
- used to display the calling handset when receiving Internal Call (for example, "Internal call from PP number 4");
- used to select the suppressed PT when removing subscription data related on the FT.

14.x.2 Procedure description

At the FP side

The terminal identity number value for the FT shall be 0. The identity number value for a PT should correspond to its subscription records number and should be in the limit (1, maximum number of subscription data on FT).

The terminal identity number shall be assigned by the FT to each PT during the location registration procedure (see clause 8.28), and shall be of 8 bits length. The terminal identity number shall be the least significant bits part of the individual assigned TPUI. The most significant bits shall follow the rules of EN 300 175-6 [6], clause 6.31.

The location registration procedure shall be used, (see clause 8.28) with the following replacement to the {LOCATE-ACCEPT} message.

Table x: Values used within the {LOCATE-ACCEPT} message

Information element	Field within the information element	Standard values within the field/information element	Normative action/comment
<<Portable-identity>>			
	<Type>	32	TPUI
	<Length of id value>	20	
	<Assignment type>	1	TPUI with number assigned
	<Identity-value>	Values in EN 300 175-6 [6] clause 6.3.1 are allowed	Only assigned individual TPUIs are allowed

At the PP side

The Identity Number in the PT shall be derived from the individual assigned TPUI received during the location registration procedure (see clause 8.28).

14.x.3 Related Procedures

Internal call "called party"

To select the called DECT entity, the dialled digit including in the <<MULTI-KEYPAD>> information element in the {CC-SETUP} message or in a {CC-INFO} message can be used with the following replacement:

Table xx: Values used within the {CC-INFO} or {CC-SETUP} message for Internal Call

Information element	Field within the information element	Standard values within the field/information element	Normative action/comment
<<Multi keypad>>	<Keypad information>	IA5 coding of terminal identity number in decimal. (NOTE2)	Terminal Identity Number of the called part - 0 for FP - 1 to n if PP
		2AH	Collective ringing (note 1)

NOTE 1: This value is used by the PP to request the FP to ring all the PP.

NOTE 2: Example for coding of the Terminal Identity Number in IA5:

- For terminal 1, terminal identity number is 0000 0001B, coded value is 31H.
- For terminal 14, terminal identity number is 0000 1110B, coded value is 31H 34H.

Calling Line Indication Presentation (CLIP) Indication

See internal call CLIP indication clause 8.x.x. The terminal identity number can be used in <<CALLING-PARTY-NUMBER>> information element.

Annex G (informative): Bibliography

ETSI TR 101 178: "Digital Enhanced Cordless Telecommunications (DECT); A high Level Guide to the DECT Standardization".

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DECT Packet Radio Service (DPRS)".

History

Document history		
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