ETSI TR 102 570 V1.1.1 (2007-03)

Technical Report

Digital Enhanced Cordless Telecommunications (DECT); New Generation DECT; Overview and Requirements

Reference
DTR/DECT-000238

Keywords
DECT, radio

ETSI

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Contents

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Foreword

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

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a rapidly changing area, and it is possible that some of the information contained in the present document may become outdated or incomplete within relatively short time-scales.

Introduction

The aim of the present document is to collect the requirements for "New Generation DECT" and to define the contents of the phase 1.

Clause 4 contains an overview and some background information about the "New Generation DECT" activities.

Clause 5 outlines the intended services and applications.

Clause 6 provides information about the support of voice and audio services.

Clause 7 deals with IP access.

Annex A provides information on use-cases and features as received from the DECT Forum.

1 Scope

The present document gives an overview of the "New Generation DECT" standardization activities and describes the architecture and the capabilities.

2 References

For the purposes of this Technical Report (TR) the following references apply:

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

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

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

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

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

[14] 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".

[15] ISO/IEC 14496-3:2005/Amd 1: "2007 Information technology - Coding of audio-visual objects - Part 3: Audio, AMENDMENT 1: Low delay AAC profile".

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

[17] IETF RFC 791: "Internet Protocol".

[18] IETF RFC 2460: "Internet Protocol version 6 (IPv6)".

[19] ITU-T Recommendation G.729 (01/2007): "Coding of speech at 8 kbit/s using conjugate-structure algebraic-code-excited linear prediction (CS-ACELP)".

3 Abbreviations

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

ADPCM	Adaptive Differential Pulse Code Modulation
CI	Common Interface
CLIP	Calling Line Identification Presentation
DECT	Digital Enhanced Cordless Telecommunications
DLC	Data Link Control
EN	European Standard
EV-CELP	Embedded Variable Code Excited Linear Prediction
T	Fixed radio Termination
IEC	International Electrotechnical Commission
IP	Internet Protocol
ISO	International Organization for Standardization
ITU	International Telecommunication Union
LAN	Local Area Network
MAC	Medium Access Control
MDCT	Modified Discrete Cosine Transform
MPEG	Moving Picture Experts Group
MTU	Maximum Transmission Unit
NG DECT	New Generation DECT
NWK	NetWorK
PCMA	Pulse Code Modulation A-law
PCMU	Pulse Code Modulation µ-law
PHL	PHysical Layer
PLC	Packet Loss Concealment
PP	Portable Part
PSTN	Public Switched Telephone Network
TDBWE	Time Domain Band-Width Extension
VoIP	Voice over Internet Protocol

4 Overview

The DECT Forum has approached TC DECT to develop the specification for "New Generation DECT" products, based on the provided market requirements. The DECT Forum requests to define and implement in the new standards full mandatory interoperability between bases/gateways and handsets, in particular since gateways and terminals from different vendors will be mixed.

To guarantee interoperability the DECT Forum intends to set up a certification program. Therefore the DECT Forum requests from ETSI to develop test specifications in addition to the system specifications.

5 Applications and features

Annex A contains a list of use cases/features, which have been proposed by DECT Forum for New Generation DECT. In order to meet the stringent time requirements, the specifications are developed in phases. The following features are included:

- Wideband speech.
- Internet access.

The present document investigates the features listed above and compiles some further information, which is relevant for the development of the specifications for "New Generation DECT".

For the support of wideband speech ITU-T Recommendation G.722 [13] has been selected as the mandatory codec. The signalling is based on GAP. The feature "Calling Line Identification Presentation" (CLIP) is mandatory. More information can be found in clause 6.

The internet access is provided by a transparent transport of IP packets over the DECT air-interface. This service is based on the Ethernet/LAN interworking of DPRS. More information can be found in clause 8.

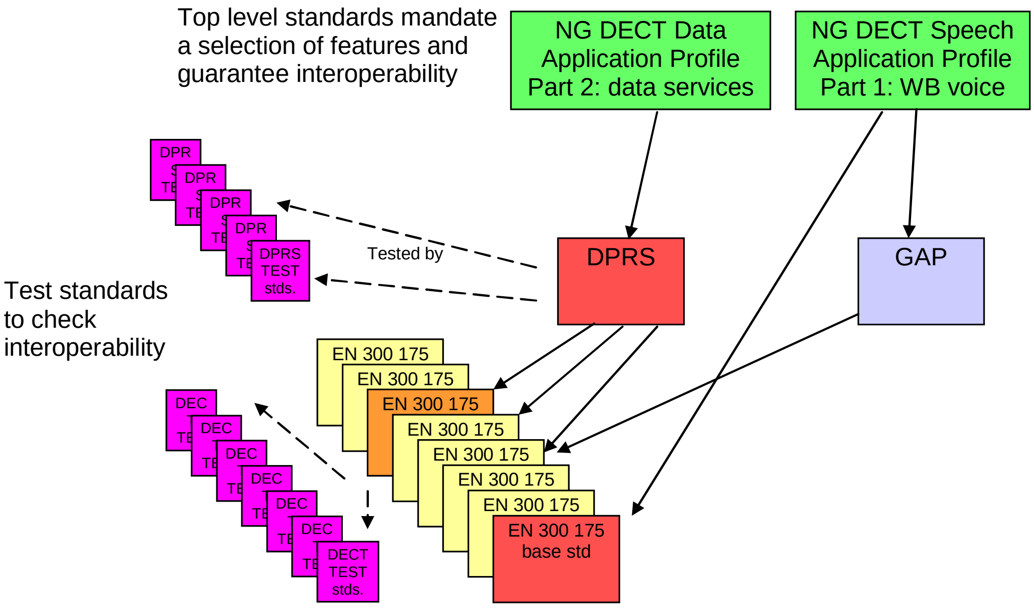

Figure 1: Structure of specifications

The New Generation DECT Technical Specifications select the mandatory features, that are defined in detail in GAP, DPRS and the base standard, which are all updated to include all new functions. Test standards are used to check that products conform to the specifications.

6 Voice and audio

In traditional telephony applications the supported bandwidth is 3,1 kHz. Increasing the bandwidth from narrow band (300 Hz to 3 400 Hz) to at least to 150 Hz to 7 000 Hz range ("wide band") will allow to increase decisively the speech quality, i.e. 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.

Possible scenarios include:

- Internal wideband calls inside a New Generation DECT system.
- Calls between two New Generation DECT systems interconnected by IP packet based network (like VoIP over the Internet).

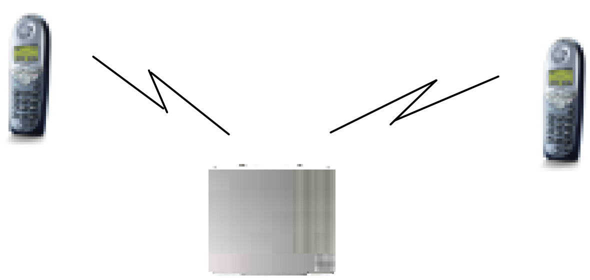

Figure 2: Internal broad-band call

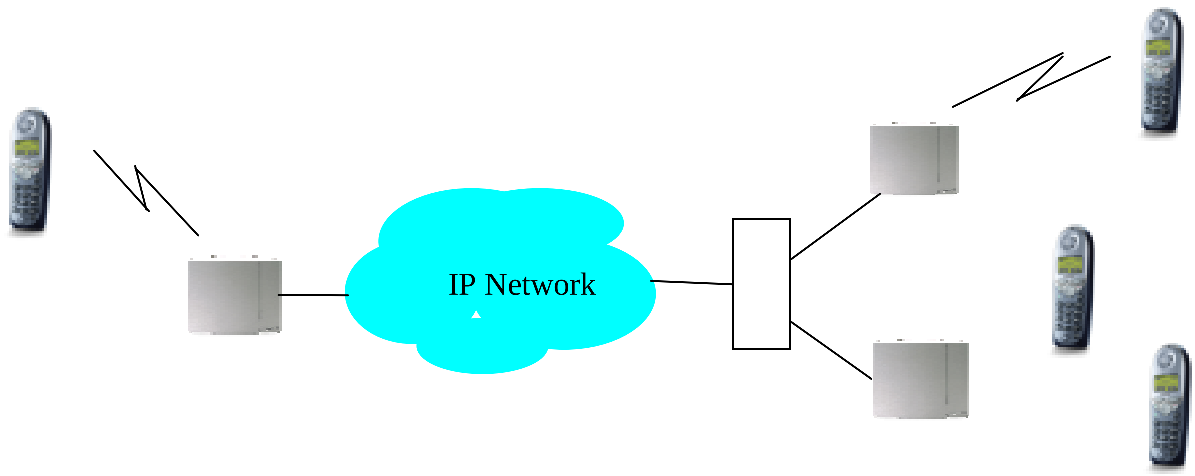

Figure 3: Wideband call via Internet

The New Generation DECT base station has to support both, narrow-band and wide-band speech terminals. A call may be modified from narrow-band to wide-band or from wide-band to narrow-band.

6.1 Speech and audio coding

Today the DECT standard codec for speech is "Adaptive Differential Pulse Code Modulation" (ADPCM) with 32 kbit/s as defined in ITU-T Recommendation G.726 [11]. It is of low complexity, offers a bandwidth of 3,1 kHz and introduces a very low delay of 0,125 ms and a quality slightly below the PSTN quality (ITU-T Recommendation G.711 [12] encoding) at 64 kbit/s

New Generation DECT improves audio quality by implementing wideband enhanced quality audio codecs. All New Generation DECT devices should implement wideband (150 Hz to 7 kHz) audio (16 kHz frequency sampling). DECT devices supporting wideband audio are to support speech coding format according to ITU-T Recommendation G.722 [13]. In addition to that, they may implement other wideband and superwideband audio codecs providing even better audio quality.

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

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

Table 1 reviews some speech and audio codecs.

Table 1: Overview of codecs

	Type	Bandwidth (kHz)	Sampling rate (kHz)	Bit rate (kbit/s)	Frame (ms)
ITU-T Recommendation G.711 [12]	LOG PCM	0,3 to 3,4	8	64	0,125
ITU-T Recommendation G.726 [11]	ADPCM	0,3 to 3,4	8	16, 24, 32, 40	0,125
ITU-T Recommendation G.722 [13]	Sub-Band ADPCM	0,05 to 7	16	64, 56, 48	0,125
ITU-T Recommendation G.729.1 [14]	EV-CELP TDBWE MDCT	0,05 to 7	16	8 to 32	20
ISO/IEC 14496-3 [15]	MPEG-4 ER AAC-LD Advanced Audio Coding Low Delay	up to 20	up to 48	range of bit rates	10 to 20 (depends on sampling rate)

6.1.1 Narrow band speech coding

ITU-T Recommendation G.726 [11] narrow band codec is mandatory for New Generation DECT in order to ensure interoperability with existing DECT systems.

ITU-T Recommendation G.711 [12] narrow band 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 [12] provides slightly higher intrinsic voice quality and no transcoding for PSTN calls. Both, A-Law and μ-Law are supported.

Table 2: ITU-T Narrow band Speech codec for New Generation DECT

Standard	ITU-T Recommendation G.726 [11]	ITU-T Recommendation G.711 [12]
	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	N	N
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 (KB)	1	≈ 0

6.1.2 Wideband speech coding

ITU-T Recommendation G.722 [13] codec 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 [13] provides a high wideband quality at bit rates of 64 kbit/s with low complexity and very low delay.

In addition, the ITU-T Recommendation G.729.1 [14] codec is recommended as an optional codec for wideband speech to provide even higher wideband quality and better robustness to packets/frames losses than
ITU-T Recommendation G.722 [13] at half the bit rate of ITU-T Recommendation G.722 [13]. 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 [19] based VoIP networks and terminals.
ITU-T Recommendation G.729.1 [14] 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 narrowband or in wideband from 14 kbit/s.
ITU-T Recommendation G.729.1 [14] already incorporates a high efficiency packet loss concealment mechanism.

Table 3: ITU-T Wideband Speech codecs for New Generation DECT

Standard	ITU-T Recommendation G.722 [13]	ITU-T Recommendation G.729.1 [14]
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 ITU-T Recommendation G.729 [19])
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 (KB)	1	17,4

To better cope with transmission errors, a Packet Loss Concealment algorithm (PLC) may be optionally implemented for ITU-T Recommendation G.722 [13]. Appendices III and IV of the ITU-T Recommendation G.722 [13] Recommendation describe packet loss concealment solutions extending the ITU-T Recommendation G.722 [13] decoder. PLC algorithms may be optionally implemented to improve voice quality in degraded transmission conditions where packets/frames may be lost (in IP networks 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.
- Appendix IV proposes an optimized complexity/quality trade off with almost no additional complexity compared with ITU-T Recommendation G.722 [13] normal decoding (0,07 WMOPS).

Since ITU-T Recommendation G.722 [13] does not incorporate any mechanism to cope with lost frames/packets, 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 [14] already incorporates a packet loss concealment mechanism.

To handle several codecs (at least ITU-T Recommendation G.726 [11] and ITU-T Recommendation G.722 [13]), 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.

6.1.3 Super-wideband speech and audio coding

MPEG-4 ER AAC-LD is optional for New Generation DECT in order to provide a higher quality than ITU-T Recommendation G.722 [13] by further extending the bandwidth to superwideband (50 Hz to 14 000 Hz, and even further, up to full audio bandwidth 20 Hz to 20 000 Hz). MPEG-4 ER AAC-LD is designed for high quality communication application including all kind of audio signals, e.g. speech and music, and provides 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 bit rate of 64 kbit/s. MPEG 4 ER AAC-LD is standardized in ISO/IEC 14496-3 [15]. The frame size is 10 ms and the algorithmic delay 20 ms.

MPEG-4 ER AAC-LD may also be used in the 32 kbit/s mode. At this bit rate, it provides a bandwidth of 11,5 kHz or more. The frame size is 20 ms and the algorithmic delay 40 ms.

Table 4: MPEG Audio codec for New Generation 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 (KB)	~28 KB (encoder) ~13 KB (decoder) IO Buffer not included	~28 KB (encoder) ~13 KB (decoder) IO Buffer not included

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

6.2 Message flow examples

6.2.1 Outgoing wideband call, default wideband speech setup parameters, ITU-T Recommendation G.722 chosen

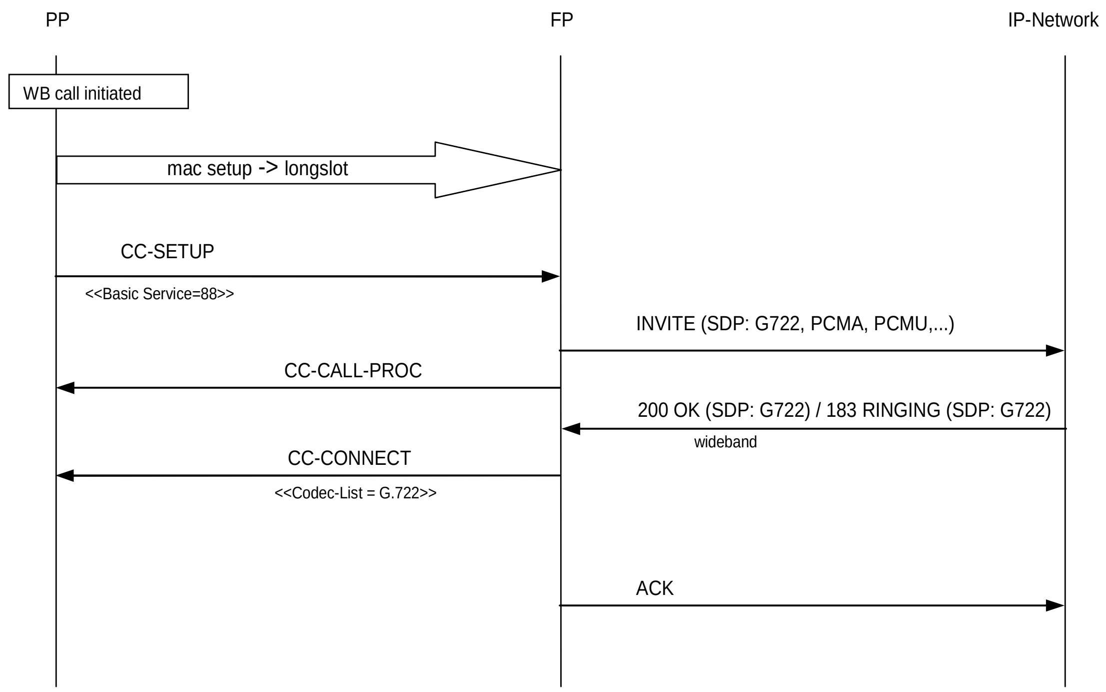

Figure 4: Outgoing call example

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. This codec list offers at least ITU-T Recommendation G.726 [11] and ITU-T Recommendation G.722 [13] mandatory codecs.

Since in this example no other Codec List is necessary on a call by call basis, the IE <<Codec-List>> can be omitted in CC-SETUP.

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

If after Service negotiation in DECT, the negotiation with the IP-network results in the need for a different speech codec for DECT, the service change procedure can be used.

6.2.2 Incoming call wideband, negotiation results in wideband

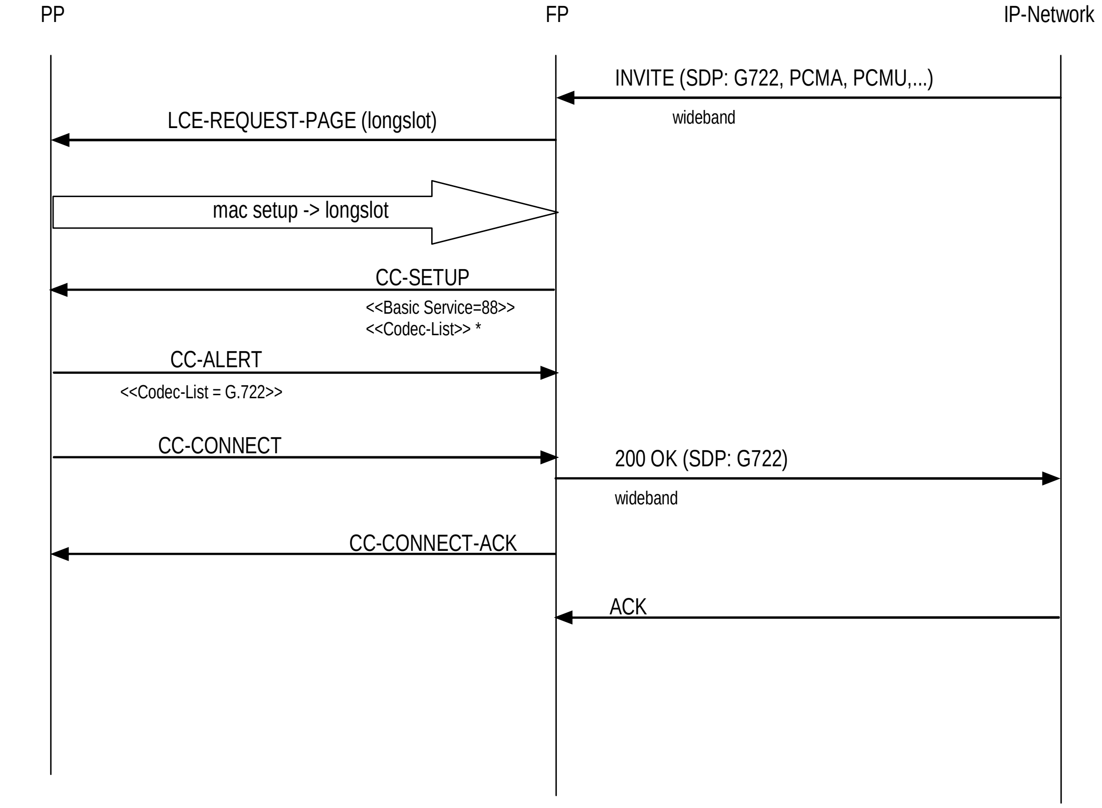

Figure 5: Incoming call example

6.2.3 Incoming Call Wideband, negotiation results in Narrowband

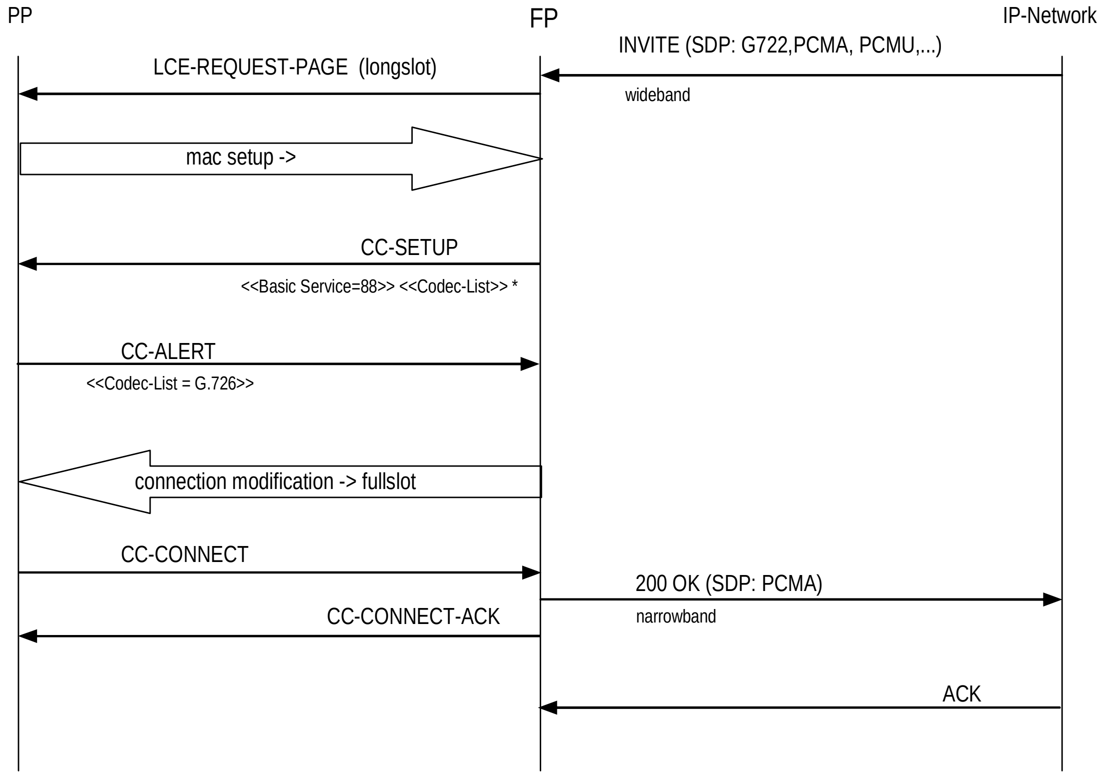

Figure 6: Incoming call example with negotiation

6.2.4 Incoming call wideband, negotiation results in narrowband, No SDP offer in invite

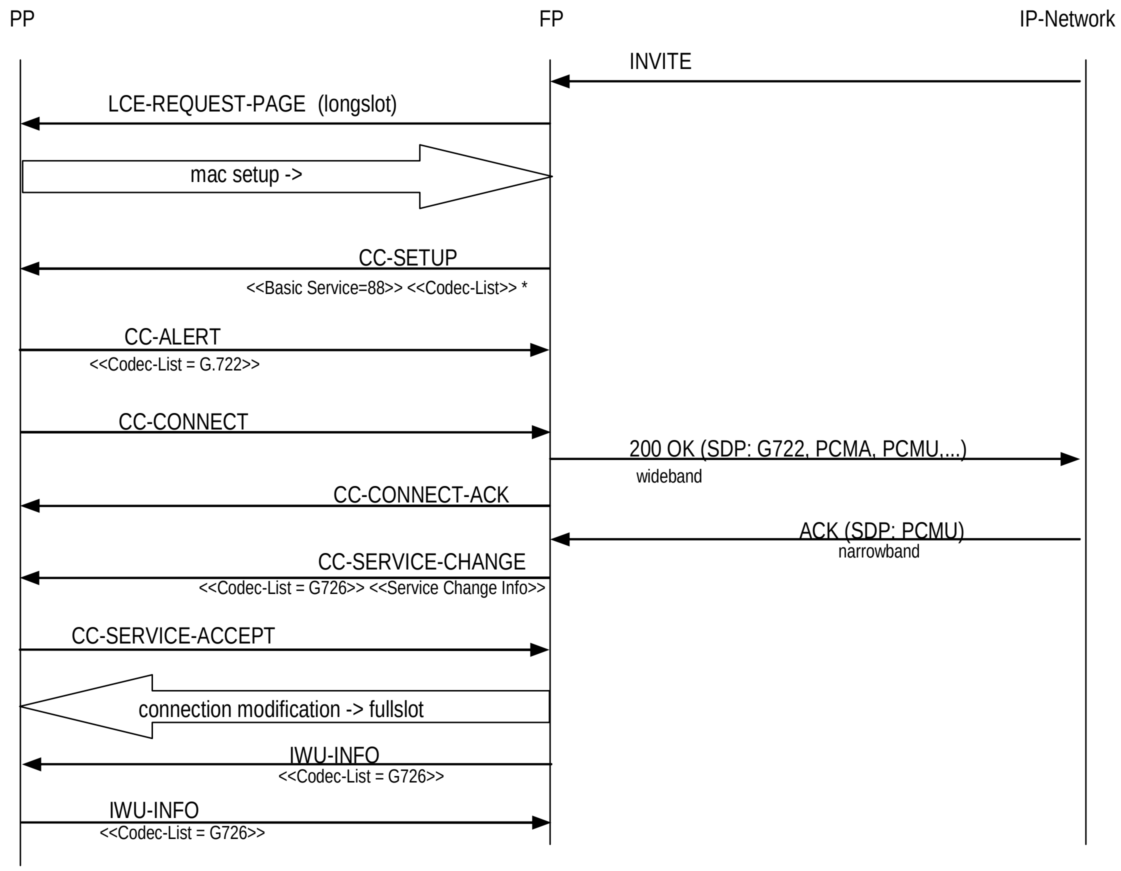

Figure 7: Incoming call example with service change

7 IP-access

The Internet Protocol version 4, IPv4, is described in RFC 791 [17]. Internet Protocol version 6, IPv6, is described in RFC 2460 [18]. The goal is to integrate the wireless DECT system into an IP-system in an optimum way.

This means:

- maintain the advantages of DECT (efficiency and quality);
- avoid interworking and aim for transparency;
- requires IP protocol handling in the terminal.

A transparent transport of IP-packets through the base station is required in New Generation DECT.

The DECT Forum has asked for a symmetric bitrate of 384 kbit/s plus 64 kbit/s for the return-channel. In order to support data-applications, which have different performance and complexity requirements, it was decided to define 3 product categories:

- Category 1: low complexity products with symmetric data of ~50 kbit/s.
- Category 2: products supporting symmetric and asymmetric data up to ~500 kbit/s.
- Category 3: products supporting symmetric and asymmetric data up to ~1 Mbit/s, the usage of optional higher layer modulation and optional multi-carrier operation results in a bit rate up to ~20 Mbit/s.

A higher category has to support all lower categories in order to ensure interoperability.

7.1 Architecture

The IP-packets are transported to/from the terminal. IPv4 packets (datagrams) vary in size, from 20 bytes (the size of the IPv4 header alone) to a maximum of 65 535 bytes. Subnetworks need not to support maximum-sized (64 KB) IP packets, as IP provides a scheme that breaks packets that are too large for a given subnetwork into fragments that travel as independent IP packets and are reassembled at the destination. The maximum packet size supported by a subnetwork is known as its Maximum Transmission Unit (MTU). Figure 8 illustrates the protocol hierarchy.

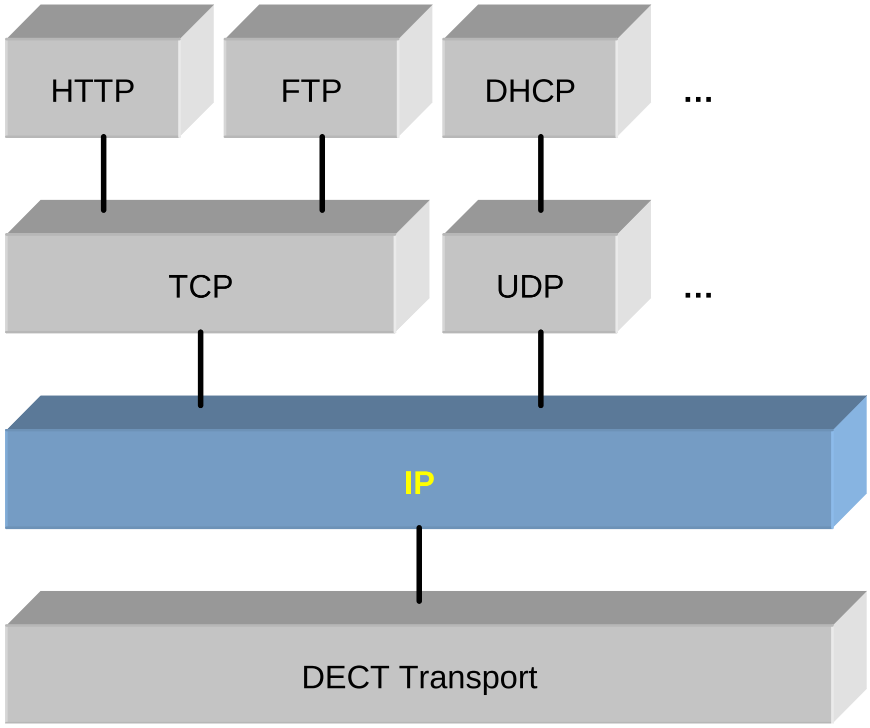

Figure 8: Protocol stack

7.2 Transport of IP-packets in DECT

The IP-packets together with relevant Ethernet-information (addresses, type) are transported in the U-plane of DECT. Not needed information such as the Ethernet-preamble and checksum are removed before encapsulation.

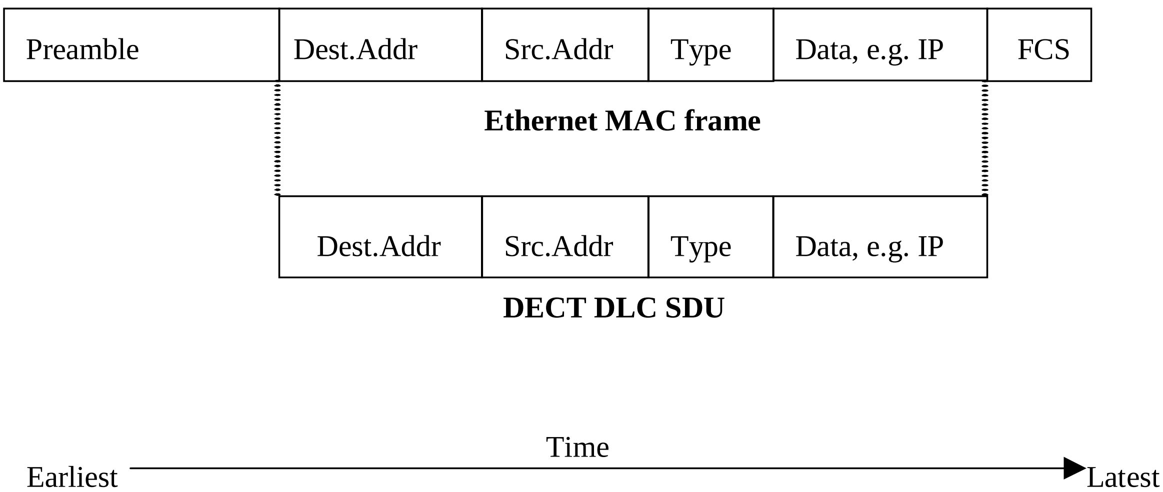

Figure 9: Transport of Ethernet/IP-packets

The Fixed Part transparently forwards the IP-data. The Fixed Part may monitor some IP content (without modifying it). The termination of the IP-protocol is done in the Portable Part.

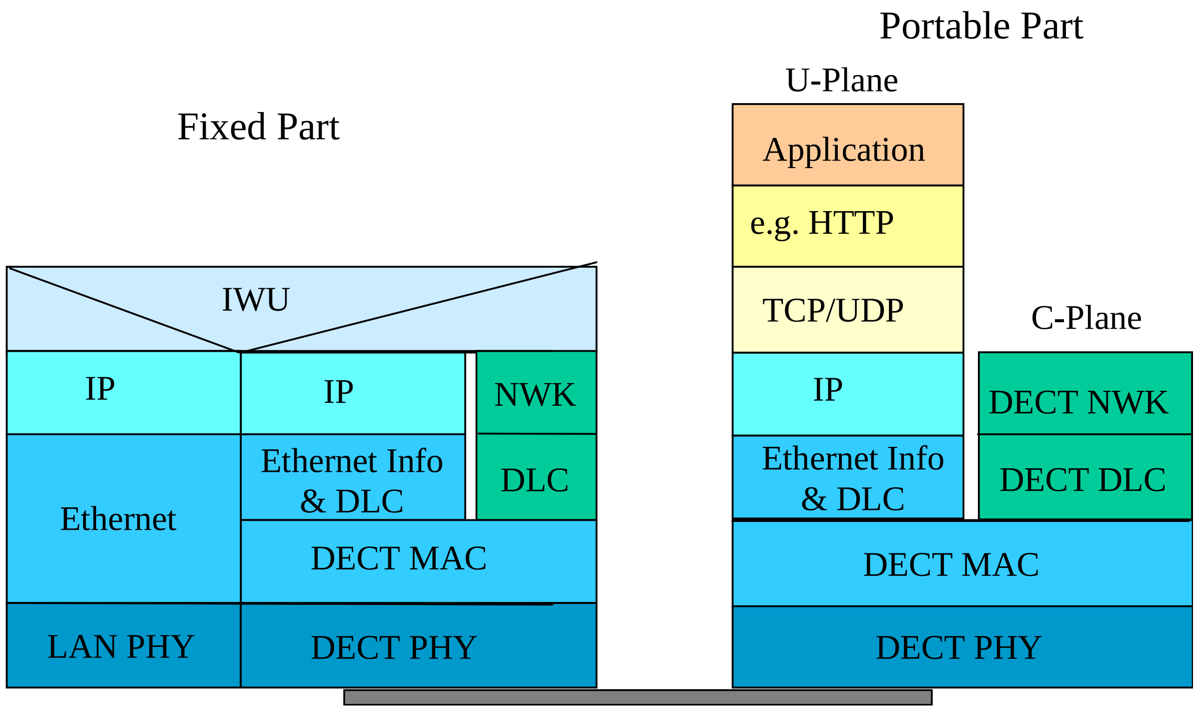

Figure 10: Protocol architecture for transport of IP-packets

On the contrary, for some applications like voice-calls over VoIP networks, it is advantageous NOT to transport VoIP packets over the DECT air-interface, but to use instead the Fixed Part as a gateway that extracts the speech information from incoming IP-packets and sends it as "DECT-speech" over the air-interface and vice-versa for outgoing "DECT-Speech". This provides better quality by a lower delay and improves the efficiency.

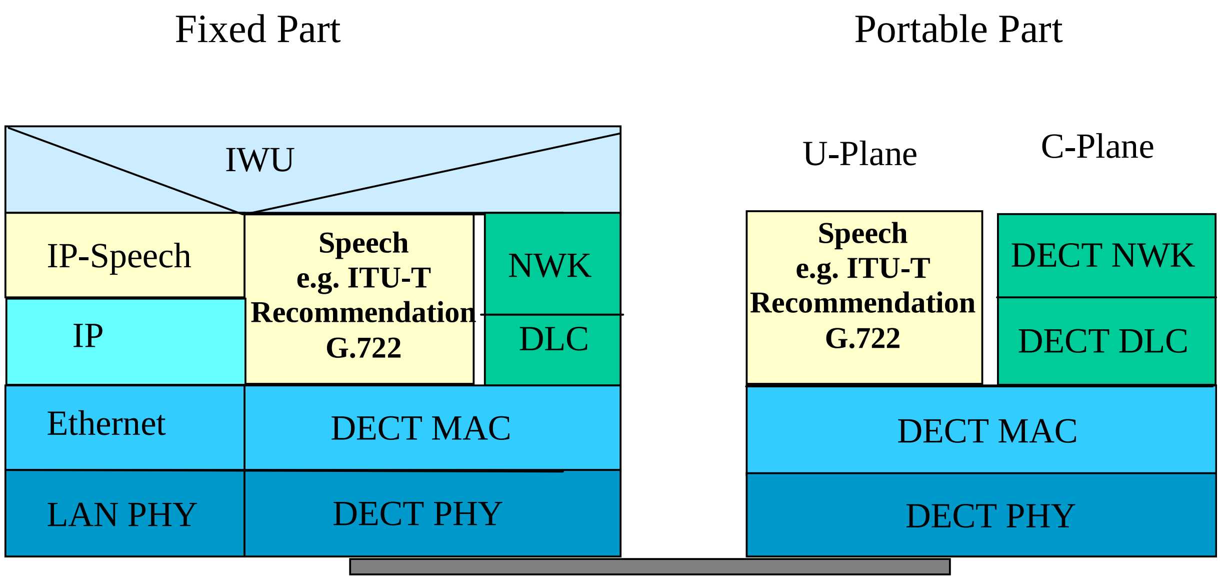

Figure 11: Protocol architecture for transport of speech

Annex A: List of use cases / features

This prioritized list has been provided by the DECT Forum.

	Use Case / Feature
1	**Voice:** Superior voice quality (Speaking and Listening) better than any existing
2	**Ease of Use:** Plug and Play functionality of all components
3	**Ease of Use:** Automatic device detection and configuration
4	**Ease of Use**: Re-Use existing handsets
5	**Reliability:** Cost, Range, Quality of Service, Battery Lifetime as good as DECT
	Personalizing Handset: DECT Handset as remote control Remote Maintain over air
	Small web services (screen: by pushing 1 button)**:** news ticker, weather, soccer results; stock, weather, soccer results, stock, traffic.
6	**Audio:** Streaming FM Quality Audio Content (Internet Radio and other content)
	Adressbook Presence: Call from Networked Address Book, using Presence Management
	Easy writing text: SMS, email
	Instant messaging: chat with multiple persons
7	**Adressbook Presence:** Dial from Phone book in the network
	Calendar with synchronization capabilities (Outlook)
8	**Voice:** Video telephony capability; Wireless Video Intercom interworking with cordless phone
	Home control capabilities: Home Monitoring, Door phone, Baby monitor, Mailbox Security for the home, Emergency Services
9	**Voice:** Multi session capability (Voice + Conference, Data, Videos, Images,…)
10	**Reliability:** Improved methods to reduce further potential health risk of RF technology in order to deal with expected new legal requirements
	Audio : Audio Content on Demand
	Reliability: Ensuring Security in all aspects: No Viruses and Worms on the Phone Set Security in communication
11	**Reliability:** Profile management and Remote Device Management helps the customer
	Audio: Streaming CD Quality Audio Content
	Adressbook Presence: Easy call with multiple persons from Networked Address Book
	Microphone Arrays: Beaming, 3D Sound 3D Displays Projecting information to the wall (portable and compact)
	Minimize radio-power consumption
	Zero Power stand-by
	Long Range
	Broadcast
	Localization
	Improved (easy-to-use) call forwarding
	Gaming
	WAP/WEB surfing

History

Document history		
V1.1.1	March 2007	Publication