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Technical Specification

Access and Terminals (AT); Digital Broadband Cable Access to the Public Telecommunications Network; IP Multimedia Time Critical Services; Part 3: Audio Codec Requirements for the Provision of Bi-Directional Audio Service over Cable Television Networks using Cable Modems

Reference

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Foreword

This Technical Specification (TS) has been produced by ETSI Technical Committee Access and Terminals (AT).

The present document is part 3 of a multi-part deliverable supporting real-time multi media services, as identified below:

Part 1: "General";

Part 2: "Architectural framework for the delivery of time critical services over cable Television networks using cable modems";

Part 3: "Audio Codec Requirements for the Provision of Bi-Directional Audio Service over Cable Television Networks using Cable Modems";

Part 4: "Network Call signalling Protocol";

Part 5: "Dynamic Quality of Service for the Provision of Real Time Services over Cable Television Networks using Cable Modems";

Part 6: "Media Terminal Adapter (MTA) device provisioning";

Part 7: "Management Information Base (MIB) Framework";

Part 8: "Media Terminal Adapter (MTA) Management Information Base (MIB)";

Part 9: "Network Call Signalling (NCS) MIB Requirements";

Part 10: "Event Message Requirements for the Provision of Real Time Services over Cable Television Networks using Cable Modems";

Part 11: "Security";

Part 12: "Internet Signalling Transport Protocol";

Part 13: "Trunking Gateway Control Protocol";

Part 14: "Operation System Support".

NOTE 1: The above list is complete for the first version of this Technical Specification (TS) (V1.1.1 2001-06). Additional parts are being proposed and these will be added to the list in future versions.

The present part is part 3 of the above mentioned series of ETSI deliverables and specifies the audio (voice) codecs that are to be used in the provisioning of bi-directional audio services over cable television distribution networks using IP technology (i.e. IPCablecom service). The present document also addresses codec options and packetization issues.

NOTE 2: The choice of a multi-part format for this deliverable is to facilitate maintenance and future enhancements.

NOTE 3: The term **MUST** or **MUST NOT** is used as a convention in the present document part to denote an absolutely mandatory aspect of the specification.

Introduction

The cable industry in Europe and across other Global regions have already deployed broadband cable television Hybrid Fibre Coax (HFC) data networks running the Cable Modem Protocol. The Cable Industry is in the rapid stages of deploying IP Voice and other time critical multimedia services over these broadband cable television networks.

The cable industry has recognized the urgent need to develop ETSI Technical Specifications aimed at developing interoperable interface specifications and mechanisms for the delivery of end to end advanced real time IP multimedia time critical services over bi-directional broadband cable networks.

IPCablecom is a set of protocols and associated element functional requirements developed to deliver Quality-of-Service (QoS) enhanced sure IP multimedia time critical communications services using packetized data transmission technology to a consumer's home over the broadband cable television Hybrid Fibre/Coaxial (HFC) data network running the Cable Modem protocol. IPCablecom utilizes a network superstructure that overlays the two-way data-ready cable television network. While the initial service offerings in the IPCablecom product line are anticipated to be Packet Voice, the long-term project vision encompasses packet video and a large family of other packet-based services.

The cable industry is a global market and therefore the ETSI standards are developed to align with standards either already developed or under development in other regions. The ETSI Specifications are consistent with the CableLabs/PacketCable set of specifications as published by the SCTE. An agreement has been established between ETSI and SCTE in the US to ensure, where appropriate, that the release of PacketCable and IPCablecom set of specifications are aligned and to avoid unnecessary duplication. The set of IPCablecom ETSI specifications also refers to ITU-SG9 draft and published recommendations relating to IP Cable Communication.

The whole set of multi-part ETSI deliverables to which the present document belongs specify a Cable Communication Service for the delivery of IP Multimedia Time Critical Services over a HFC Broadband Cable Network to the consumers home cable telecom terminal. 'IPCablecom' also refers to the ETSI working group program that shall define and develop these ETSI deliverables.

Many cable television operators are upgrading their facilities to provide two-way capability, and are using this capability to provide high-speed IP data services per ITU-T Recommendations J.83 and J.112. These operators now want to expand the capability of this delivery platform to include bi-directional voice communication and other time critical services. The present document is one of a series of documents required to achieve this goal. It provides guidance on audio (voice) codec selection that will provide for a high quality, interoperable service.

1 Scope

The present set of documents specify IPCablecom, a set of protocols and associated element functional requirements. These have been developed to deliver Quality-of-Service (QoS), enhanced sure IP multimedia time critical communication services, using packetized data transmission technology to a consumer's home over a cable television Hybrid Fibre/Coaxial (HFC) data network.

NOTE 1: IPCablecom set of documents utilize a network superstructure that overlays the two-way data-ready cable television network, e.g. as specified within ES 201 488 and ES 200 800.

While the initial service offerings in the IPCablecom product line are anticipated to be Packet Voice and Packet Video, the long-term project vision encompasses a large family of packet-based services. This may require in the future, not only careful maintenance control, but also an extension of the present set of documents.

NOTE 2: The present set of documents aims for global acceptance and applicability. It is therefore developed in alignment with standards either already existing or under development in other regions and in International Telecommunications Union (ITU).

2 References

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

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

ITU-T Recommendation G.165 (1993): "Echo cancellers".

ITU-T Recommendation G.168 (1997): "Digital network echo cancellers".

ITU-T Recommendation G.711 (1988): "Pulse code modulation (PCM) of voice frequencies".

ITU-T Recommendation G.728 (1992): "Coding of speech at 16 kbit/s using low-delay code excited linear prediction".

ITU-T Recommendation G.729 (1998): "Coding of speech at 8 kbit/s using conjugate-structure algebraic-code-excited linear-prediction (CS-ACELP); Annex E: 11.8 kbit/s CS-ACELP speech coding algorithm".

ITU-T Recommendation J.83 (1997): "Digital multi-programme systems for television, sound and data services for cable distribution".

ITU-T Recommendation J.112: "Transmission systems for interactive cable television services".

ITU-T Recommendation V.18 (1998): "Operational and interworking requirements for DCEs operating in the text telephone mode".

RFC 1890 (1996): "RTP Profile for Audio and Video Conferences with Minimal Control".

RFC 2327 (1998): "SDP: Session Description Protocol".

ETSI ES 201 488: "Data-Over-Cable Service Interface Specifications; Radio Frequency Interface Specification".

ETSI ES 200 800: "Digital Video Broadcasting (DVB); DVB interaction channel for Cable TV distribution systems (CATV)".

ITU-T Recommendation G.726: "40, 32, 24, 16 kbit/s adaptive differential pulse code modulation (ADPCM)".

3 Definitions and abbreviations

3.1 Definitions

For the purposes of the present document, the following terms and definitions apply:

Access Node: layer two termination device that terminates the network end of the ITU-T Recommendation J.112 connection

NOTE: It is technology specific. In ITU-T Recommendation J.112 annex A it is called the INA while in annex B it is the CMTS.

Cable Modem: layer two termination device that terminates the customer end of the J.112 connection

IPCablecom: ETSI working group project that includes an architecture and a series of Specifications that enable the delivery of real time services (such as telephony) over the cable television networks using cable modems

3.2 Abbreviations

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

AN	Access Node
CPE	Customer Premise Equipment
DTMF	Dual Tone Multi Frequency
HFC	Hybrid Fibre Coax
IP	Internet Protocol
MTA	Media Terminal Adaptor
PSTN	Public Switched Telephone Network
QoS	Quality-of-Service
VAD	Voice Activity Detection

4 Audio Codec Requirements

Offering a competitive and/or superior product requires support for more than high-quality delivery of audio. In addition to features and signalling capabilities, which are beyond the scope of this document, the audio codec application must provide transparent support for certain audio features. These include general detection mechanisms, DTMF, fax, analog modem, echo cancellation, and hearing-impaired support.

4.1 DTMF support

Dual-tone multi-frequency (DTMF) support allows employment of dual-tone multiple frequency signals by either an autodialing system or through manual entry of tones. In order for DTMF tones to be captured correctly by the receiving device, tonal integrity (frequency accuracy and signal duration) must be maintained even through compression and transcoding.

MTA devices must successfully pass DTMF tone transmissions. The specified codecs must be capable of transparently passing these tones in-band.

4.2 Fax and Modem Support

IPCablecom needs to support analog fax and modem interfaces for two reasons. First, fax and modem equipment are common, and customers will continue to use these familiar devices for some years to come. Sond, even with cable modem access, many users will continue to access their dial-up networks using a traditional modem.

In order to provide customers with access for analog fax and modems, Media Terminal Adapter (MTA) devices must be able to detect fax/modem signals and signal these detections using the appropriate protocol. The codec at each end is then switched to G.711 for the remainder of the session. Additionally, echo cancellation is disabled for the duration of the fax/modem session. After the fax/modem session has completed, echo cancellation is re-enabled.

A more robust solution for supporting fax is to employ fax relay. Fax relay involves demodulating the T.30 transmission and sending control and image data over the IP network. At the receiving end, the received data is remodulated and sent to the fax terminal using another T.30 session. This is described in the ITU-T standard T.38. Client devices may employ fax relay.

4.3 Echo Cancellation Support

When end-to-end delay in an audio communication is more than 20 ms, an artifact called line echo can occur. This echo, if not removed, will be heard by the remote talker (thus it is also called talker echo) whenever he or she speaks.

Line echo cancellation must be provided in IPCablecom MTA and Gateway devices to mitigate the effects of line echo. This echo canceller must allow both parties to speak simultaneously (double-talk), so that one talker does not seize the line and block out the other user from being heard.

During periods when only the remote talker is speaking, the local echo canceller should either inject comfort noise or allow some noise to pass through to the remote talker, so that a "dead-line" is not perceived. However, if local voice activity detection (VAD) is enabled, either the noise injection should be disabled, or the echo canceller should communicate its state with the VAD, in order for the VAD to not estimate the injected noise mistakenly as the true background noise.

In an application where the MTA is located in a home, the length of the echo canceller is typically short (8 ms or less). For PSTN gateway applications, the echo canceller length is typically much longer (32 ms or longer). Vendors may choose to differentiate their products by providing longer echo canceller lengths suitable for their application, or other programmable parameters.

In applications where a non-standard bi-directional audio interface is used (e.g., microphone and headset), echo cancellation may not be necessary. However, where a microphone and speakers are used, acoustic echo cancellation may be necessary, and vendors implementing these products should employ acoustic echo cancellation.

The performance of the line echo canceller should comply with either ITU-T Recommendation G.165 or ITU-T Recommendation G.168.

4.4 Asymmetrical Services Support

MTA devices should be capable of supporting employment of different codecs for upstream and downstream audio channels. This allows potential optimization of device resources, network bandwidth, and user service quality.

4.5 Hearing-impaired Services Support

For hearing-impaired people, TDD equipment can be the primary communication link to the outside world. This type of equipment has evolved lacking the type of standardization allowing broad interoperability among international manufacturers. The ITU recently adopted the V.18 standard to begin alleviating this problem. ITU-T Recommendation V.18 outlines a procedure, which includes protocol negotiation for connecting these devices.

Since CPE for the hearing impaired consists of text input/output devices coupled with voice-band modems, any system designed to support them would need to be able to pass voice-band modem tones coherently. Of the list of proposed voice codecs, only G.711 would be able to achieve this, given that the other standards are not designed for passage of complex tones associated with modem communications. Typically, these devices will interface to the PSTN via an acoustical coupler to a phone or with a regular telephone jack.

MTA devices must support detection of ITU-T Recommendation V.18 hearing-impaired tones, including V.18 annexes A, B and F. Support for ITU-T Recommendation V.18 annex G is optional. Upon detection of a V.18 signal, the codec at each end is then switched to G.711 for the remainder of the session. Additionally, echo cancellation is disabled for the duration of the V.18 call. It is optional to disable echo cancellation for annex B because it is DTMF-based. After the session has completed, echo cancellation is re-enabled.

5 Mandatory Codecs

G.711 must be supported in all MTAs. This codec provides high-quality service and is ubiquitous. It provides the "fallback" position for services such as fax, modem, and hearing-impaired services support, as well as common gateway transcoding support. In addition, G.711 is used as the fallback mode if there are not enough resources to establish a new connection using the requested codec (e.g. two channels of G.728 or G.729 annex E are already in existence, and there are not enough resources for a third connection to use a compressed codec).

5.1 µ-law and A-law Support

Both G.711 encoding modes (µ-law and A-law) must be supported. If the analog-to-digital interface at the encoder uses A-law encoding, then A-law encoding must be selected for G.711 encoding. Otherwise, if the analog-to-digital interface at the encoder uses µ-law encoding, then µ-law encoding must be selected for G.711 encoding.

However, if one end of a voice connection uses an A-law interface, while the other end uses µ-law, then the A-law decoder MUST transcode the incoming µ-law packets to A-law, while the µ-law decoder MUST transcode the incoming A-law packets to µ-law.

5.2 Packet Loss Concealment

All Media Gateways and Multimedia Terminal Adaptors MUST detect audio packet loss and implement some method to conceal losses from end-users. Specifications for low bit rate codecs (e.g. G.728 and G.729) include methods for concealment. For G.711 implementations, the method defined in G.711 Appendix I is recommended.

6 Additional Codecs

In addition to G.711, MTAs also should support at least one of the following codecs.

6.1 G.728

G.728 should be supported in all MTA's. IPCablecom has a need to provide best or high voice quality. G.728 is a mid-bitrate (16 kb/s), high-quality solution. Supporting a codec in this range provides high-quality, low-bandwidth performance for on-net calls and ensures the highest possible performance for applications, such as IVR systems. In addition, it provides superior background noise handling.

6.2 G.729 annex E

G.729 annex E should be supported in all MTAs. IPCablecom has a need to provide best or high voice quality. G.729 annex E is a mid-bitrate (11,8 kb/s), high-quality solution. Supporting a codec in this range provides high-quality, low-bandwidth performance for on-net calls and ensures the highest possible performance for applications, such as IVR systems. In addition, it provides superior background noise handling.

7 Optional Features

7.1 Wideband Codecs

Given that the majority of early customers will be "black phone" users, support for a wideband (i.e. greater than circuit voice bandwidth) codecs is not necessary. However, some vendors optionally may choose to differentiate their product by selecting components that will support higher fidelity in the event a wideband codec is provisioned at some time in the future.

7.2 Optional Codecs

A vendor may supply any codecs not described herein.

7.3 Voice Activity Detection (VAD)

A vendor may employ VAD to reduce bandwidth consumption. If employed, this capability must be optional, allowing disabling.

8 Packetization

Packet size influences both delay and the impact of packet loss. The larger the packet size, the greater the delay and the greater the impact of a lost packet. This suggests that the optimal packet size for voice applications is fairly small. Individual packets should not contain more than 20 ms of voice frames and must not contain more than 30 ms of voice frames. In addition, individual packets must contain an integral number of frames of sampling data, and any one frame of sampling data must be totally contained within one packet.

Annex A (normative): Session Description of CODECs

Session descriptor protocol (SDP) messages are used to describe multimedia sessions for the purposes of session announcement, session invitation, and other forms of multimedia session initiation. SDP descriptions are used in both Network Call Signalling (NCS) [J.ncs] and Distributed Call Signalling (DCS) [future]. This clause describes the required specification of the codec in SDP, and the required mapping of the SDP description into RSVP flowspecs.

A typical SDP description contains many fields that contain information regarding the session description (protocol version, session name, session attribute lines, etc.), the time description (time the session is active, etc.), and media description (media name and transport, media title, connection information, media attribute lines, etc.). The two critical components for specifying a codec in an SDP description are the media name and transport address (m) and the media attribute lines (a).

The media name and transport address (m) are of the form:

- m=<media> <port> <transport> <fmt list>.

The media attribute line(s) (a) are of the form:

- a=<token>:<value>.

A typical IP-delivered voice communication would be of the form:

- m=audio 3456 RTP/AVP 0;
- a=ptime:10.

On the transport address line (m), the first term defines the media type, which in the case of an IP voice communications session is audio. The sond term defines the UDP port to which the media is sent (port 3456). The third term indicates that this stream is an RTP Audio/Video profile. Finally, the last term is the media payload type as defined in the RTP Audio/Video Profile (reference RFC 1890). In this case, the 0 represents a static payload type of u-law PCM coded single channel audio sampled at 8 kHz (A value of 8 would be used to represent A law). On the media attribute line (a), the first term defines the packet formation time (10 ms).

Payload types other than those defined in RFC 1890 are dynamically bound by using a dynamic payload type from the range 96 - 127, as defined in RFC 2327, and a media attribute line. For example, a typical SDP message for G.726 would be composed as follows:

- m=audio 3456 RTP/AVP 96;
- a= rtpmap:96 G726-32/8000.

The payload type 96 indicates that the payload type is locally defined for the duration of this session, and the following line indicates that payload type 96 is bound to the encoding "G726-32" with a clock rate of 8 000 samples/s.

CODECs defined in this specification MUST be encoded with the following string names in the rtpmap parameter:

Table A.1: Codec RTPMap Parameters

Codec	Rtpmap Parameter
G.726 at 16 kb/s	G726-16/8000
G.726 at 24 kb/s	G726-24/8000
G.726 at 32 kb/s	G726-32/8000
G.726 at 40 kb/s	G726-40/8000
G.728	G728/8000
G.729A	G729A/8000
G.729E	G729E/8000

In addition, G.711 MAY be encoded with a dynamic payload type code, in an rtpmap parameter using the name PCMU/8000.

For every defined CODEC (whether it is represented in SDP as a static or dynamic payload type), table A.2 describes the mapping that MUST be used from either the payload type or ASCII string representation to the bandwidth requirements for that CODEC.

The Mapping of RTP/AVP code to RSVP Flowspec (as used by Dynamic Quality of Service [J.dqos])) must be according to table A.2. The implementation of the 15 ms time period for G.711 CODEC is mandatory. All other implementations are optional.

Table A.2: Mapping of Session Description Parameters to RSVP Flowspec

Parameters from Session Description			Flowspec parameters		Comments
RTP/AVP code	**Rtpmap**	**Ptime (ms)**	**Values b, m, M (note 1)**	**Values r, p (note 2)**	
0	<none>	9	112 bytes	12 444 bytes/s	G.711 using the Payload Type defined by IETF
0	<none>	10	120 bytes	12,000 bytes/s	
0	<none>	15	160 bytes	10 666 bytes/s	
0	<none>	20	200 bytes	10,000 bytes/s	
0	<none>	30	280 bytes	9,333 bytes/s	
96-127	PCMU/8000	9	112 bytes	12 444 bytes/s	G.711 PCM, 64 kb/s, default CODEC
96-127	PCMU/8000	10	120 bytes	12,000 bytes/s	
96-127	PCMU/8000	15	160 bytes	10 666 bytes/s	
96-127	PCMU/8000	20	200 bytes	10,000 bytes/s	
96-127	PCMU/8000	30	280 bytes	9,333 bytes/s	
96-127	G726-16/8000	10	60 bytes	6,000 bytes/s	
96-127	G726-16/8000	20	80 bytes	4,000 bytes/s	
96-127	G726-16/8000	30	100 bytes	3,333 bytes/s	
96-127	G726-24/8000	10	70 bytes	7,000 bytes/s	
96-127	G726-24/8000	20	100 bytes	5,000 bytes/s	
96-127	G726-24/8000	30	130 bytes	4,333 bytes/s	
2	<none>	10	80 bytes	8,000 bytes/s	G.726-32, identical to G.721, which is assigned Payload Type 2 by IETF
2	<none>	20	120 bytes	6,000 bytes/s	
2	<none>	30	160 bytes	5,333 bytes/s	
96-127	G726-32/8000	10	80 bytes	8,000 bytes/s	
96-127	G726-32/8000	20	120 bytes	6,000 bytes/s	
96-127	G726-32/8000	30	160 bytes	5,333 bytes/s	
96-127	G726-40/8000	10	90 bytes	9,000 bytes/s	
96-127	G726-40/8000	20	140 bytes	7,000 bytes/s	
96-127	G726-40/8000	30	190 bytes	6,333 bytes/s	
15	<none>	10	60 bytes	6,000 bytes/s	G.728, assigned Payload Type 15 by IETF
15	<none>	15	70 bytes	4 666 bytes/s	
15	<none>	20	80 bytes	4,000 bytes/s	
15	<none>	30	100 bytes	3,333 bytes/s	
96-127	G728/8000	10	60 bytes	6,000 bytes/s	G.728, LD-CELP, 16 kb/s
96-127	G728/8000	15	70 bytes	4 666 bytes/s	
96-127	G728/8000	20	80 bytes	4,000 bytes/s	
96-127	G728/8000	30	100 bytes	3,333 bytes/s	
18	<none>	10	50 bytes	5,000 bytes/s	G.729A, identical to G.729, assigned Payload Type 18 by IETF
18	<none>	20	60 bytes	3,000 bytes/s	
18	<none>	30	70 bytes	2,333 bytes/s	
96-127	G729A/8000	10	50 bytes	5,000 bytes/s	G.729A, CS-ACELP, 8 kb/s, 10 ms frame size with 5 ms lookahead
96-127	G729A/8000	20	60 bytes	3,000 bytes/s	
96-127	G729A/8000	30	70 bytes	2,333 bytes/s	

Parameters from Session Description			Flowspec parameters		Comments
RTP/AVP code	Rtpmap	Ptime (ms)	Values b, m, M (note 1)	Values r, p (note 2)	
96-127	G729E/8000	10	55 bytes	5,500 bytes/s	G.729E, CS-ACELP, 11,8 kb/s, 10 ms frame size with 5 ms lookahead
96-127	G729E/8000	20	70 bytes	3,500 bytes/s	
96-127	G729E/8000	30	85 bytes	2,833 bytes/s	
NOTE 1: b: is bucket depth (bytes). m: is minimum policed unit (bytes). M: is maximum datagram size (bytes). NOTE 2: r: is bucket rate (bytes/s). p: is peak rate (bytes/s).					

Annex B (informative): Bibliography

ITU-T Recommendation G.101 (1996): "The transmission plan".

ITU-T Recommendation G.107 (1998): "The E-Model, a computational model for use in transmission planning".

ITU-T Recommendation G.108 (1999): "Application of the E-Modal: A planning guide".

ITU-T Recommendation G.109 (1999): "Definition of categories of speech transmission quality".

ITU-T Recommendation G.177 (1999): "Transmission planning for voiceband services over hybrid Internet/PSTN connections".

ITU-T Recommendation T.30 (1996): "Procedures for document facsimile transmission in the general switched telephone network".

ITU-T Recommendation T.38 (1998): "Procedures for real-time Group 3 facsimile communication over IP networks".

List of ITU-T Recommendations referring to IPCablecom:

ITU-T Recommendation J.160: "Architectural framework for the delivery of time critical services over cable television networks using cable modems".

ITU-T Recommendation J.161: "Audio codec requirements for the provision of bi-directional audio service over cable television networks using cable modems".

ITU-T Recommendation J.162: "Network call signalling protocol for the delivery of time critical services over cable television networks using cable modems".

ITU-T Recommendation J.163: "Dynamic quality of service for the provision of real time services over cable television networks using cable modems".

ITU-T Recommendation J.164: "IPCablecom event messages".

ITU-T Recommendation J.165: "IPCablecom Internet Signalling Transport Protocol".

ITU-T Recommendation J.166: "IPCablecom management information base (MIB) framework".

ITU-T Recommendation J.167: "Media terminal adapter (MTA) device provisioning requirements for the delivery of real time services over cable television networks using cable modems".

ITU-T Recommendation J.168: "IPCablecom Media Terminal Adapter (MTA) MIB requirements".

ITU-T Recommendation J.169: "IPCablecom network call signalling (NCS) MIB requirements".

ITU-T Recommendation J.170: "IPCablecom Surity specification".

ITU-T Recommendation J.171: "IPCablecom Trunking Gateway Control Protocol (TGCP)".

History

Document history		
V1.1.1	June 2001	Publication