

International Telecommunication Union

ITU-T

TELECOMMUNICATION
STANDARDIZATION SECTOR
OF ITU

J.161

(06/2007)

SERIES J: CABLE NETWORKS AND TRANSMISSION
OF TELEVISION, SOUND PROGRAMME AND OTHER
MULTIMEDIA SIGNALS

IPCablecom

**Audio and video codec requirements and usage
for the provision of bidirectional audio services
over cable television networks using cable
modems**

ITU-T Recommendation J.161



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ITU-T Recommendation J.161

Audio and video codec requirements and usage for the provision of bidirectional audio services over cable television networks using cable modems

Summary

ITU-T Recommendation J.161 specifies the media aspects of the interfaces between IP-Cablecom client devices for audio and video communication. Specifically, it identifies the audio and video codecs necessary to provide the highest quality and the most resource-efficient service delivery to the customer. This Recommendation also specifies the performance required in client devices to support future IP-Cablecom codecs. Additionally, this Recommendation describes a suggested methodology for optimal network support for codecs.

Source

ITU-T Recommendation J.161 was approved on 20 June 2007 by ITU-T Study Group 9 (2005-2008) under the ITU-T Recommendation A.8 procedure.

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Audio and video codec requirements and usage for the provision of bidirectional audio services over cable television networks using cable modems

1 Scope

This Recommendation specifies the media aspects of the interfaces between IPCablecom client devices for audio and video communication. Specifically, it identifies the audio and video codecs necessary to provide the highest quality and the most resource-efficient service delivery to the customer. This Recommendation also specifies the performance required in client devices to support future IPCablecom codecs. Additionally, this Recommendation describes a suggested methodology for optimal network support for codecs.

1.1 Introduction and overview

The quality of audio and video delivered over the IPCablecom architecture will depend on multiple factors: the end device performance, the network's inherent quality, and the intelligence of the system resource allocation policy. This Recommendation specifies codecs and capabilities supporting audio and video applications, with a particular emphasis on the stringent requirements of IP-based voice communications.

Acceptable voice communications functionality imposes strict delay and packet-loss criteria on IP implementations and will thus stress system resources, particularly if bandwidth becomes congested or saturated. Video applications – while more forgiving to dropped packets and delay – require bandwidth of at least an order of magnitude more than audio applications. The IPCablecom architecture is designed to support both voice and video applications simultaneously.

Speech and video compression are evolving technologies. New algorithms are being enabled as more sophisticated and higher performing processors become available at lower cost. Additionally, the system infrastructure and mechanisms for allocating resources will evolve. Due to this dynamism, the priority in designing the architecture is to define a robust system to accommodate evolving technology without creating a legacy burden.

Therefore, the IPCablecom philosophy is to establish cost-effective envelopes for network and device performance to enable the most appropriate current technology, while allowing upgrades as technology and market needs evolve. To address near-term market needs, this Recommendation also specifies codec and performance mandates to deliver the quality-of-service necessary for launching competitive services.

1.2 Purpose of this Recommendation

The purpose of this Recommendation is to extend the existing IPCablecom v1Codec Recommendation by introducing two new low-bit codecs, [ITU-T T.38] fax relay for reliable fax transmission, [IETF RFC 2833] DTMF Relay for reliable DTMF transmission and metrics to measure voice quality. It is issued to facilitate design and field-testing leading to the manufacturability and interoperability of conforming hardware and software by multiple vendors.

1.3 Phasing of requirements

The codec requirements contained in this Recommendation cover both audio and video multimedia terminals (MTAs) and trunking gateways (media gateway). The term MTA-2 is used to define a terminal supporting video.

In the initial phase of IP_Cablecom, MTAs are not required to support the MTA-2 requirements as defined in clause 8. MTAs **MUST** support the requirements for audio terminals as defined in clauses 5, 6, 7, and 9.

Support for video terminals will be **REQUIRED** in later phases of IP_Cablecom. All MTA-2s **MUST** support the requirements defined in clause 8.

2 References

2.1 Normative references

The following ITU-T Recommendations and other references contain provisions which, through reference in this text, constitute provisions of this Recommendation. At the time of publication, the editions indicated were valid. All Recommendations and other references are subject to revision; users of this Recommendation are therefore encouraged to investigate the possibility of applying the most recent edition of the Recommendations and other references listed below. A list of the currently valid ITU-T Recommendations is regularly published. The reference to a document within this Recommendation does not give it, as a stand-alone document, the status of a Recommendation.

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|-----------------|--|
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| [ITU-T G.165] | ITU-T Recommendation G.165 (1993), <i>Echo cancellers</i> . |
| [ITU-T G.168] | ITU-T Recommendation G.168 (2007), <i>Digital network echo cancellers</i> . |
| [ITU-T G.711] | ITU-T Recommendation G.711 (1988), <i>Pulse code modulation (PCM) of voice frequencies</i> . |
| [ITU-T G.728] | ITU-T Recommendation G.728 (1992), <i>Coding of speech at 16 kbit/s using low-delay code excited linear prediction</i> . |
| [ITU-T G.729] | ITU-T Recommendation G.729 (2007), <i>Coding of speech at 8 kbit/s using conjugate-structure algebraic-code-excited linear prediction (CS-ACELP)</i> . |
| [ITU-T J.170] | ITU-T Recommendation J.170 (2005), <i>IP_Cablecom security specification</i> . |
| [ITU-T J.171.x] | ITU-T Recommendation J.171.x (2005), <i>IP_Cablecom trunking gateway control protocol (TGCP)</i> . |
| [ITU-T P.56] | ITU-T Recommendation P.56 (1993), <i>Objective measurement of active speech level</i> . |
| [ITU-T P.561] | ITU-T Recommendation P.561 (2002), <i>In-service non-intrusive measurement device – Voice service measurements</i> . |
| [ITU-T T.38] | ITU-T Recommendation T.38 (2007), <i>Procedures for real-time Group 3 facsimile communication over IP networks</i> . |
| [ITU-T V.18] | ITU-T Recommendation V.18 (2000), <i>Operational and interworking requirements for DCEs operating in the text telephone mode</i> . |
| [IETF RFC 2833] | IETF RFC 2833 (2000), <i>RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals</i> . |
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- [IETF RFC 3551] IETF RFC 3551 (2003), *RTP Profile for Audio and Video Conferences with Minimal Control*.
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- [ANSI/SCTE 24-21] ANSI/SCTE 24-21 (2006), *BV16 Speech Codec Specification for Voice over IP Applications in Cable Telephony*.

2.2 Informative references

- [ITU-T F.703] ITU-T Recommendation F.703 (2000), *Multimedia conversational services*.
- [ITU-T H.243] ITU-T Recommendation H.243 (2005), *Procedures for establishing communication between three or more audiovisual terminals using digital channels up to 1920 kbit/s*.
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- [ITU-T J.163] ITU-T Recommendation J.163 (2005), *Dynamic quality of service for the provision of real-time services over cable television networks using cable modems*.
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- [ITU-T T.140] ITU-T Recommendation T.140 (1998), *Protocol for multimedia application text conversation*.
- [IETF RFC 2327] IETF RFC 2327 (1998), *SDP: Session Description Protocol*.
- [IETF RFC 4103] IETF RFC 4103 (2005), *RTP Payload for Text Conversation*.
- [COX] COX (R.V.): Current Methods of Speech Coding, *International Journal of High Speed Electronics & Systems*, Vol. 8, No. 1, pp. 13-68, 1997.

3 Terms and definitions

This Recommendation uses the following terms:

3.1 audio server: An audio server plays informational announcements in IP-Cablecom network. Media announcements are needed for communications that do not complete and to provide enhanced information services to the user. The component parts of audio server services are media players and media player controllers.

3.2 call management server (CMS): Controls the audio connections. Also called a call agent in MGCP/SGCP terminology. This is one example of an application server.

3.3 cable modem termination system (CMTS): The device at a cable head-end which implements the DOCSIS RFI MAC protocol and connects to CMs over an HFC network.

3.4 delay: The absolute time required for a signal to transit from source to receiver.

3.5 dynamic quality of service (DQoS): Assigned on the fly for each communication depending on the QoS requested.

3.6 hybrid fibre/coaxial cable (HFC): An HFC system is a broadband bidirectional shared media transmission system using fibre trunks between the head-end and the fibre nodes, and coaxial distribution from the fibre nodes to the customer locations.

3.7 Internet control message protocol (ICMP): An extension to the Internet Protocol, ICMP supports packets containing error, control and information messages.

3.8 jitter: Variability in the delay of a stream of incoming packets making up a flow such as a voice communication.

3.9 latency: The time, expressed in quantity of symbols, taken for a signal element to pass through a device.

3.10 media gateway (MG): Provides the bearer circuit interfaces to the PSTN and transcodes the media stream.

3.11 media gateway controller (MGC): The overall controller function of the PSTN gateway. Receives, controls and mediates call-signalling information between the IP-Cablecom and PSTN.

3.12 multimedia terminal adapter (MTA): Contains the interface to a physical voice device, a network interface, codecs, and all signalling and encapsulation functions required for VoIP transport, class features signalling and QoS signalling.

3.13 off-net call: A communication connecting an IP-Cablecom subscriber out to a user on the PSTN.

3.14 on-net call: A communication placed by one customer to another customer entirely on the IP-Cablecom network.

3.15 pulse code modulation (PCM): A commonly employed algorithm to digitize an analog signal (such as a human voice) into a digital bit stream using simple analog to digital conversion techniques.

3.16 quality of service (QoS): Guarantees network bandwidth and availability for applications.

3.17 registered Jack-11 (RJ-11): A standard 4-pin modular connector commonly used for connecting a phone unit into a wall jack.

3.18 real-time transport protocol (RTP): A protocol for encapsulating encoded voice and video streams.

3.19 transit delays: The time difference between the instant at which the first bit of a PDU crosses one designated boundary, and the instant at which the last bit of the same PDU crosses a second designated boundary.

3.20 trunk: An analog or digital connection from a circuit switch that carries user media content and may carry voice signalling (M_F , R_2 , etc.).

3.21 user datagram protocol (UDP): A connectionless protocol built upon Internet protocol (IP).

NOTE – Delay and latency are similar concepts and frequently used interchangeably. However, delay focuses on the time to transit from transmitter (such as a speaker's mouth) to a receiver (such as a listener's ear), while latency focuses on the time to transit from a receiver to a transmitter, as would be the case for a signal going through a piece of equipment.

4 Abbreviations, acronyms and conventions

4.1 Abbreviations and acronyms

This Recommendation uses the following abbreviations and acronyms:

CIF	Common Intermediate Format
CM	DOCSIS Cable Modem
CMS	Call Management Server
CMTS	Cable Modem Termination System
Codec	Coder-DECoder
DOCSIS®	Data-Over-Cable Service Interface Specification
DQoS	Dynamic Quality of Service
DTMF	Dual-tone Multi Frequency (tones)
FEC	Forward Error Correction
HFC	Hybrid Fibre/Coaxial cable
ICMP	Internet Control Message Protocol
IP	Internet Protocol
ISDN	Integrated Services Digital Network
ISP	Internet Service Provider
IVR	Interactive Voice Response System
LSSGR	LATA Switching System Generic Requirements
MG	Media Gateway
MGC	Media Gateway Controller
MTA	Multimedia Terminal Adapter
NCS	Network Call Signalling
NTSC	National Television Standards Committee. (Defines the analog colour television, broadcast standard used today in North America.)

PAL	Phase Alternate Line. (The European colour television format that evolved from the American NTSC standard.)
PCM	Pulse Code Modulation
PSTN	Public Switched Telephone Network
QCIF	Quarter Common Intermediate Format
QoS	Quality of Service
RJ-11	Registered Jack-11
RSVP	Resource Reservation Protocol
RTCP	Real-time Transport Control Protocol
RTP	Real-time Transport Protocol
SDP	Session Description Protocol
TCP	Transmission Control Protocol
UDP	User Datagram Protocol
VAD	Voice Activity Detection
VBR	Variable Bit Rate
VoIP	Voice over IP

4.2 Conventions

Throughout this Recommendation, words that are used to define the significance of particular requirements are capitalized. These words are:

"MUST"	This word or the adjective "REQUIRED" means that the item is an absolute requirement of this Recommendation.
"MUST NOT"	This phrase means that the item is an absolute prohibition of this Recommendation.
"SHOULD"	This word means that there may exist valid reasons in particular circumstances to ignore this item, but the full implications should be understood and the case carefully weighed before choosing a different course.
"SHOULD NOT"	This phrase means that there may exist valid reasons in particular circumstances when the listed behaviour is acceptable or even useful, but the full implications should be understood and the case carefully weighed before implementing any behaviour described with this label.
"MAY"	This word or the adjective "OPTIONAL" means that this item is truly optional. One vendor may choose to include the item because a particular marketplace requires it or because it enhances the product, for example; another vendor may omit the same item.

5 Background

This clause outlines the IP-Cablecom architecture support elements and the DOCSIS network infrastructure necessary to deliver quality audio and video service. It is intended to clarify external interfaces and functional requirements necessary to implement the targeted audio and video quality using speech and video codecs.

The key requirement for voice communications using IP transmission is the ability to attain "toll" or better audio quality. Given the variable nature of shared packet mediums and the stringent human-factor requirements of this quality standard, it is necessary to optimize multiple system parameters to attain this goal. Additionally, IPCablecom has been tasked with offering superior quality, exceeding current PSTN standards where feasible. Key requirements from the IPCablecom product definition requiring architectural optimization for codecs follow.

5.1 IPCablecom voice communications quality requirements

As defined in the IPCablecom architecture Recommendation [ITU-T J.160], requirements for toll-quality voice communications service in IPCablecom include numerous metrics to ensure competitive or superior quality and service to the PSTN. In order to support these requirements, network plant and equipment may have to be groomed. In order to provide guidelines for that grooming, several network implications affecting codec performance are discussed below.

5.2 Network preparation for codec support

The critical areas of network performance, which must be optimized in tandem with codecs, are packet loss, delay and jitter. Elaboration of network/codec implications for each of these areas follows.

5.2.1 Packet loss control

There is a direct correlation between packet integrity and audio quality. Anecdotal codec research suggests initial 3% packet loss rate results, on average, in a reduction in mean opinion scores (MOS) of 0.5 point, on a scale of 5. Due to less-than-pristine conditions and human-detectable compromises with most codecs, the resulting audio quality for a 3% packet loss rate will be well below PSTN "toll" quality. Above 3%, codec performance falls off rapidly, and resulting voice quality is unacceptable.

Applications and/or codecs may provide error correction or concealment mechanisms, which may increase latency through buffering. Once delay thresholds have been exceeded, the tradeoff between delay and fidelity becomes an untenable situation.

5.2.2 Delay control

Control of overall delay requires a hand-in-hand effort by the system resources and the application – in this case, a speech or video application dominated by the codec component.

There are multiple device elements and network components inducing delay during traversal of an audio signal from capture of the speaker's voice until reception at the receiver's ear. The primary contributors to delay for an on-net voice and off-net communication along this path are:

- audio sampling and analog-to-digital conversion;
- buffering of samples (audio framing, plus look-ahead);
- compression processing;
- packetization of compressed data;
- local network (DOCSIS) traversal;
- routing to the backbone network;
- backbone traversal;
- far-end reception of packets and traversal of local access;
- buffering of out-of-order and delayed packets;
- decoding, decompression, and reconstruction of the audio stream.

The major contributors to codec-related delay in the network are described below.

5.2.2.1 Delay control: Buffering

While network jitter and corresponding buffering increase call delay, another source of buffering can be induced by the application as a corrective response to severe packet loss. Although the ultimate solution to additional buffering delay is a pristine network, realistically some packet loss will occur.

Accounting for lost packets suggests the need for support concealment or reconstruction of lost data, and in many instances these techniques employ some mechanism of redundant information encoding, temporally shifting and embedding audio frames in the data stream. This not only increases the effective bandwidth requirement, but also creates, in effect, an additional buffer to allow for reassembly, increasing delay.

In order to apply certain reconstruction methodologies in an optimal fashion, the application needs accurate data regarding the statistical characteristics of the media stream. Some information is available through real-time transport control protocol (RTCP) mechanisms, such as a gross measure of packet loss. Additional information, such as burst frequency and predictive time-of-day effects, would improve the potential of the application to make optimal adjustments. Planning for the collection and analysis of this type of network information will allow developers more options in the future, potentially creating applications that will increase network utilization efficiency or quality.

5.2.2.2 Delay control: Optimal framing/packetization

As outlined in clause 5.2.1, the loss of audio data frames can have a severe impact on audio quality. The packing of multiple audio frames into a single packet will exacerbate the problem, effectively expanding the loss of one packet into the loss of multiple adjacent audio frames of data. This also increases latency by buffering larger portions of audio samples prior to sending.

One way to minimize these effects is to send small packets containing the minimum number of frames. This will increase bandwidth use by increasing the header-to-data ratio for packets, but will minimize latency and potentially increase reconstruction quality. This suggests that the optimal packet size for voice applications is fairly small, containing compressed information for one, two, or, at most, three frames of sampling data (typically corresponding to 10, 20, or 30 milliseconds of voice frames).

5.2.2.3 Delay control: Packet timing optimization

To avoid additional buffering delay, packets **MUST** be sent at a rate equal to integral multiples of the audio sample frame rate of the codec. This synchronization results in lockstep between the codec framing and packet transmission.

The frame sizes of the mandatory codec is shown in Table 1; further frame sizes for the recommended codecs are listed in Table 2. Default packetization periods are specified in [IETF RFC 3551].

Table 1 – Frame sizes of the mandatory codec

Codec	Frame size (ms)
G.711	0.125

Table 2 – Frame sizes of the recommended codecs

Codec	Frame size (ms)
G.729 Annex E	10
G.728	0.625
iLBC	20
iLBC	30
BV16	5

5.2.3 Codec transcoding minimization

Transcoding occurs whenever a packetized voice signal encounters an edge device without compatible codec support. Transcoding introduces additional latency during the decode/recode stage. Additionally, if transcoding resources at the edge gateway are shared, additional delay can be introduced.

Transcoding between compressed codecs also results in degradation of the original sample, as current codec compression techniques are not lossless. In the event that a combination of transcoding and packet loss causes a signal to be reduced below minimum quality, it is likely that a higher bandwidth codec will be employed. Thus, transcoding artifacts can result in the unintended side effect of higher system bandwidth utilization.

In the case of on-net and off-net IP connections, transcoding can be eliminated if all necessary codecs are supported on the client. This is, in fact, impractical but can be optimized statistically if a device supports multiple codecs and can be updated periodically.

5.2.4 Bandwidth minimization

There are two primary mechanisms that client devices may employ to minimize the amount of bandwidth used for their audio/video applications:

- A compressed, low bit rate codec may be applied, thus reducing the bandwidth required.
- A codec may employ some form of variable bit rate transmission.

The selection of codecs occurs at the device's discretion or via network selection, depending on the protocol employed. Regardless, this takes place after the initial capabilities exchange to determine a compatible codec between endpoints, and assumes that the requested bandwidth is granted by the bandwidth broker element.

Variable rate transmission may occur when a codec employs methods resulting in a non-constant bitstream representation of voice data. Voice activity detection (VAD) – silence suppression – is a basic form of variable rate transmission, sending little or no data during speaker silence periods. More advanced variable bit rate (VBR) encoding occurs when a codec dynamically optimizes the compression bitstream.

6 Device requirements for audio codec support

As markets evolve, endpoint codecs will change too, and neither a provider nor a customer can be expected to replace their cable modem/MTA frequently to accommodate these market changes. Given the rapid growth of the digital wireless market in particular, it is likely that, at some point, a statistically significant portion of voice communications will require a new codec in the standard suite in order to maintain voice quality.

Since interconnection between diverse codecs requires transcoding – which introduces unwelcome delay and artifacts – one goal of the IPcablecom network is to minimize transcoding. Thus, a forward-looking approach to codec evolution is necessary – one which supports the most important