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DIGITAL SYSTEMS AND NETWORKS

Quality of service and performance – Generic and user-
related aspects

Performance parameter definitions for quality of speech and other voiceband applications utilizing IP networks

ITU-T Recommendation G.1020

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ITU-T Recommendation G.1020

Performance parameter definitions for quality of speech and other voiceband applications utilizing IP networks

Summary

The transmission of speech and other voiceband applications over packet networks brings with it new, sometimes unique, forms of quality degradation. There are many existing definitions for packet network performance parameters, yet the desire to control the quality of non-elastic isochronous applications requires additional, complementary information. The purpose of this Recommendation is to define packet network and terminal performance parameters that better reflect the perceived quality of the target applications. It is largely focused on quality impairments resulting from delay variation and packet loss which are peculiar to IP and other packet-based technologies, and which do not appear in traditional TDM networks. It discusses the interactions and trade-offs among these packet impairments, and describes mechanisms such as de-jitter buffers and packet loss concealment for reducing their effects on the quality of speech and other applications. However, this Recommendation avoids overlap by making reference to existing definitions wherever possible.

The parameters defined by this Recommendation extend beyond the IP layer in many cases. End-to-end packet system (combination of end terminals and network) parameters are also necessary to determine the speech/voiceband quality. Clauses 5, 6 and 7 collect the parameter definitions for source terminals, packet networks, and destination terminals, respectively. Overall parameters are defined in clause 8.

In this version, the Recommendation adds formal definitions for many of the voice metrics identified in RFC 3611, RTCP Extended Reports (RTCP-XR).

Annex A defines the gateway-specific reference points and parameters. Annex B provides information on packet loss distributions and packet loss models, and defines burst packet loss parameters. Annex C specifies an adaptive de-jitter buffer emulator and defines de-jitter buffer parameters. Appendix I tabulates the RTCP-XR parameters, along with the examination existing Recommendations and the final action, which typically was the addition of a parameter in this Recommendation.

Source

ITU-T Recommendation G.1020 was approved on 14 July 2006 by ITU-T Study Group 12 (2005-2008) under the ITU-T Recommendation A.8 procedure.

FOREWORD

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Introduction

The transmission of speech and other voiceband applications over packet networks brings with it new, sometimes unique, forms of quality degradation. There are many existing definitions for packet network performance parameters, yet the desire to control the quality of non-elastic isochronous applications requires additional, complementary information. The purpose of this Recommendation is to define packet network and terminal performance parameters that better reflect the perceived quality of the target applications, extending beyond the IP layer in many cases. End-to-end packet system (combination of end terminals and network) parameters are also necessary to determine the speech/voiceband quality, and this Recommendation defines them as well.

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ITU-T Recommendation G.1020

Performance parameter definitions for quality of speech and other voiceband applications utilizing IP networks

1 Scope

This Recommendation defines a set of performance parameters for packet networks and end terminals that can assist in quantifying the end-to-end quality of speech and other voiceband applications. It is largely focused on quality impairments resulting from delay variation and packet loss which are peculiar to IP and other packet-based technologies, and which do not appear in traditional TDM networks. It discusses the interactions and trade-offs among these packet impairments, and describes mechanisms such as de-jitter buffers and packet loss concealment for reducing their effects on the quality of speech and other applications.

This Recommendation recognizes existing performance parameter definitions, and avoids duplication. Many factors that determine the quality of speech and voiceband applications are common to both TDM and IP-based networks, and are addressed in existing Recommendations. In the parlance of ITU-T Rec. I.350, the scope of this Recommendation is limited to the information transfer function of the 3×3 matrix, and only to the bearer channel. Call processing aspects of connection access and disengagement (e.g., dialtone delay and post-dialling delay) are not considered in this Recommendation. Furthermore, this Recommendation does not specify numerical objectives for packet networks or end terminals, although this will be the subject of follow-on work.

Figure 1 illustrates this scope, along with some other specifications with their areas of coverage. This Recommendation only defines parameters that describe packet terminal and packet transmission impairments that are unique to speech and voiceband application quality assessment.

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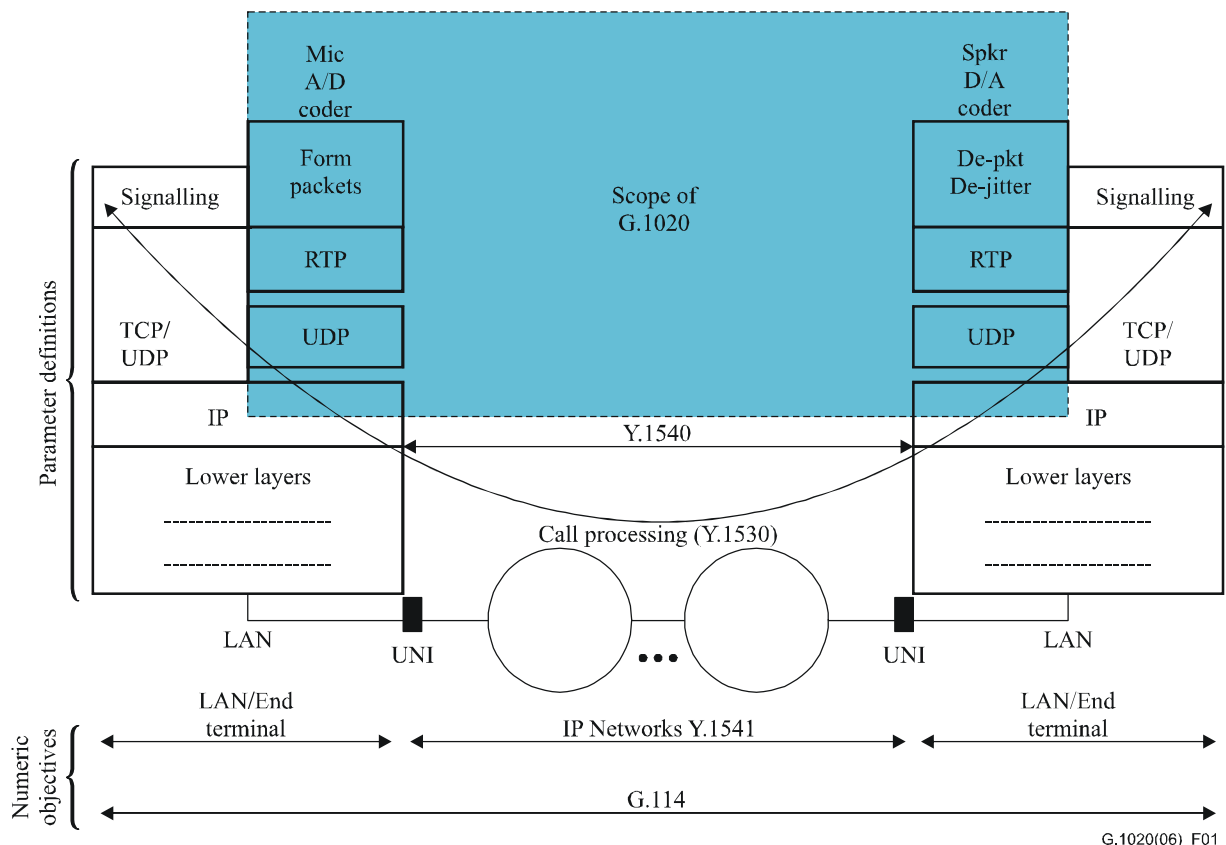


Figure 1/G.1020 – Scope of ITU-T Rec. G.1020 as it relates to other performance specifications

Note that the number of IP networks between terminals is not limited in these definitions.

Other ITU-T Recommendations supplement the parameters provided in this Recommendation. For example, transmission planning over hybrid Internet/PSTN connections is covered in ITU-T Rec. G.177. Still, other Recommendations specify such parameters in the context of assessing the performance of IP packet transfer on international data communication services (e.g., ITU-T Rec. Y.1540). Network performance objectives for different QoS classes of IP-based services are described in ITU-T Rec. Y.1541, and end-to-end one-way delay objectives are specified in ITU-T Rec. G.114.

New Recommendations that complement ITU-T Rec. G.1020 are anticipated. Call processing aspects of the connection are in development. The ITU-T is also currently working on new Recommendations dealing with situations where the network is mainly an IP network with islands of PSTN, and where the network is mainly a PSTN network with islands of IP. ITU-T work to specify the performance objectives for VoIP terminals and gateways was recently completed in ITU-T Rec. P.1010. Finally, there is work on methods to assess the performance by measuring metrics related to the end-to-end quality of VoIP.

The definitions of packet transmission parameters that are unique to ATM networks are explicitly out of the scope of this Recommendation.

This Recommendation should be particularly useful to those new to the area of Voice over IP (VoIP) who want to gain a better understanding of the factors affecting the quality of these telecommunication systems. Developers of telecommunication equipment can use the parameters defined in this Recommendation to specify relevant aspects of their contribution to end-to-end performance. Service providers can use these parameters to effectively summarize performance of IP network solutions.

2 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.

- ITU-T Recommendation G.107 (2005), *The E-model, a computational model for use in transmission planning*.
- ITU-T Recommendation G.113 (2001), *Transmission impairments due to speech processing*.
- ITU-T Recommendation G.114 (2003), *One-way transmission time*.
- ITU-T Recommendation G.177 (1999), *Transmission planning for voiceband services over hybrid Internet/PSTN connections*.
- ITU-T Recommendation I.356 (2000), *B-ISDN ATM layer cell transfer performance*.
- ITU-T Recommendation P.51 (1996), *Artificial mouth*.
- ITU-T Recommendation P.57 (2005), *Artificial ears*.
- ITU-T Recommendation P.1010 (2004), *Fundamental voice transmission objectives for VoIP terminals and gateways*.
- ITU-T Recommendation Y.1540 (2002), *Internet protocol data communication service – IP packet transfer and availability performance parameters*.
- ITU-T Recommendation Y.1541 (2006), *Network performance objectives for IP-based services*.
- IETF RFC 3611 (2003), *RTP Control Protocol Extended Reports (RTCP XR)*.

3 Definitions

This Recommendation defines the following terms:

3.1 ear reference point: A virtual point for geometric reference located at the entrance to the listener's ear, traditionally used for calculating telephonometric loudness ratings [P.57].

3.2 terminal input measurement point: A measurement point in the physical medium connecting a terminal to an IP network that is crossed as IP packets leave the IP network and enter the terminal. This measurement point is as close to the terminal as possible.

3.3 IP terminal: An endpoint device intended for connecting to an IP network to support speech communications. These devices can be dedicated (e.g., a telephone set) or general purpose (e.g., a computer running an application that performs the terminal function).

3.4 terminal output reference point: A measurement point in the physical medium connecting a terminal to an IP network that is crossed as IP packets leave the terminal and enter the IP network. This measurement point is as close to the terminal as possible.

3.5 de-jitter buffer: A buffer designed to remove the delay variation (i.e., jitter) in packet arrival times. Data is put into the de-jitter buffer at a variable rate (i.e., whenever they are received from the network), and taken out at a constant rate.

3.6 mouth reference point: The point on the reference axis, 25 mm in front of the lip plane [P.51].

3.7 real-time signal: A signal accurately representing acoustic or electrical signals in the time domain.

3.8 receive electrical reference point: An electrical point of reference that is equivalent to the ear reference point from a terminal delay measurement perspective.

3.9 send electrical reference point: An electrical point of reference that is equivalent to the mouth reference point from a terminal delay measurement perspective.

4 Abbreviations

This Recommendation uses the following abbreviations:

ADC	Analogue-to-Digital Converter
ATM	Asynchronous Transfer Mode
DAC	Digital-to-Analogue Converter
DSCP	Differentiated Services Code Point
Dst	Destination
HDLC	High-level Data Link Control
IETF	Internet Engineering Task Force
IP	Internet Protocol
IPER	IP Packet Error Ratio
IPErr	Errored IP Packet Count
IPLR	IP Packet Loss Ratio
IPPM	IP Performance Metrics working group
IPRE	IP packet transfer Reference Event
IPSLB	IP Packet Severe Loss Block
IPSLBR	IP Packet Severe Loss Block Ratio
IPTD	IP Packet Transfer Delay
IPv4	Internet Protocol version 4
IPv6	Internet Protocol version 6
MAPDV2	Mean Absolute Packet Delay Variation 2
NA	Not Available
PSTN	Public Switched Telephone Network
QoS	Quality of Service
RFC	Request For Comments
RSVP-TE	Resource Reservation Protocol – Traffic Engineering
RTCP	Real-time Transport Control Protocol
RTP	Real-time Transport Protocol
RTPErr	Errored RTP Packet Count
SPR	Spurious Packet Ratio
Src	Source