INTERNATIONAL TELECOMMUNICATION UNION

ITU-T

TELECOMMUNICATION
STANDARDIZATION SECTOR
OF ITU

H.460.9

Amendment 1

(03/2004)

SERIES H: AUDIOVISUAL AND MULTIMEDIA SYSTEMS

Infrastructure of audiovisual services – Supplementary services for multimedia

Support for online QoS-monitoring reporting within H.323 systems

Amendment 1: New Annex B: Extended performance metrics

ITU-T Recommendation H.460.9 (2002) – Amendment 1

ITU-T H-SERIES RECOMMENDATIONS

AUDIOVISUAL AND MULTIMEDIA SYSTEMS

CHARACTERISTICS OF VISUAL TELEPHONE SYSTEMS	H.100–H.199
INFRASTRUCTURE OF AUDIOVISUAL SERVICES	
General	H.200–H.219
Transmission multiplexing and synchronization	H.220–H.229
Systems aspects	H.230–H.239
Communication procedures	H.240–H.259
Coding of moving video	H.260–H.279
Related systems aspects	H.280–H.299
Systems and terminal equipment for audiovisual services	H.300–H.349
Directory services architecture for audiovisual and multimedia services	H.350–H.359
Quality of service architecture for audiovisual and multimedia services	H.360–H.369
Supplementary services for multimedia	**H.450–H.499**
MOBILITY AND COLLABORATION PROCEDURES	
Overview of Mobility and Collaboration, definitions, protocols and procedures	H.500–H.509
Mobility for H-Series multimedia systems and services	H.510–H.519
Mobile multimedia collaboration applications and services	H.520–H.529
Security for mobile multimedia systems and services	H.530–H.539
Security for mobile multimedia collaboration applications and services	H.540–H.549
Mobility interworking procedures	H.550–H.559
Mobile multimedia collaboration inter-working procedures	H.560–H.569
BROADBAND AND TRIPLE-PLAY MULTIMEDIA SERVICES	
Broadband multimedia services over VDSL	H.610–H.619

For further details, please refer to the list of ITU-T Recommendations.

ITU-T Recommendation H.460.9

Support for online QoS-monitoring reporting within H.323 systems

Amendment 1

New Annex B: Extended performance metrics

Summary

This annex contains an improved set of Voice over IP call performance metrics that are consistent with those defined in the RTP Control Protocol Extended Reports (RTCP XR) [RFC 3611]. The use of a common set of metrics for QoS reporting via call control protocols and media path protocols lowers the complexity of end-systems and makes it easier to compare data when resolving service quality problems.

Source

Amendment 1 to ITU-T Recommendation H.460.9 was approved on 15 March 2004 by ITU-T Study Group 16 (2001-2004) under the ITU-T Recommendation A.8 procedure.

FOREWORD

The International Telecommunication Union (ITU) is the United Nations specialized agency in the field of telecommunications. The ITU Telecommunication Standardization Sector (ITU-T) is a permanent organ of ITU. ITU-T is responsible for studying technical, operating and tariff questions and issuing Recommendations on them with a view to standardizing telecommunications on a worldwide basis.

The World Telecommunication Standardization Assembly (WTSA), which meets every four years, establishes the topics for study by the ITU-T study groups which, in turn, produce Recommendations on these topics.

The approval of ITU-T Recommendations is covered by the procedure laid down in WTSA Resolution 1.

In some areas of information technology which fall within ITU-T's purview, the necessary standards are prepared on a collaborative basis with ISO and IEC.

NOTE

In this Recommendation, the expression "Administration" is used for conciseness to indicate both a telecommunication administration and a recognized operating agency.

Compliance with this Recommendation is voluntary. However, the Recommendation may contain certain mandatory provisions (to ensure e.g. interoperability or applicability) and compliance with the Recommendation is achieved when all of these mandatory provisions are met. The words "shall" or some other obligatory language such as "must" and the negative equivalents are used to express requirements. The use of such words does not suggest that compliance with the Recommendation is required of any party.

INTELLECTUAL PROPERTY RIGHTS

ITU draws attention to the possibility that the practice or implementation of this Recommendation may involve the use of a claimed Intellectual Property Right. ITU takes no position concerning the evidence, validity or applicability of claimed Intellectual Property Rights, whether asserted by ITU members or others outside of the Recommendation development process.

As of the date of approval of this Recommendation, ITU had received notice of intellectual property, protected by patents, which may be required to implement this Recommendation. However, implementors are cautioned that this may not represent the latest information and are therefore strongly urged to consult the TSB patent database.

CONTENTS

ITU-T Recommendation H.460.9

Support for online QoS-monitoring reporting within H.323 systems

Amendment 1

New Annex B: Extended performance metrics

B.1 Scope

This annex describes a set of Extended Performance Metrics for Voice over IP QoS reporting that provide more detailed insight into call quality and causes of degradation than basic RTCP statistics. The metrics described in this annex are consistent with those described in the RTCP XR Voice over IP Metrics Payload described in IETF RFC 3611.

B.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 (2003), *The E-Model, a computational model for use in transmission planning*.

– ITU-T Recommendation G.108 (1999), *Application of the E-Model: A planning guide*.

– IETF RFC 3611 (2003), *RTP Control Protocol Extended Reports (RTCP XR)*.

B.3 Definitions

N/A.

B.4 Abbreviations

This Recommendation uses the following abbreviations:

MOSCQ Mean Opinion Score for Conversational Quality

MOSLQ Mean Opinion Score for Listening Quality

RTCP RTP Control Protocol

RTCP XR RTCP Extended Reports

RTP Real-time Transfer Protocol

B.5 Parameter descriptions

B.5.1 Network packet loss rate

The proportion of packets lost since the start of transmission expressed as an 8-bit binary fraction obtained by dividing the number of packets lost in the transmission path by the total number of packets expected and multiplying this value by 256 and taking the integer part. Thus a value of 0 would correspond to a packet loss rate of zero and a value of 64 would correspond to a packet loss rate of 0.25 (corresponding to 25 per cent).

B.5.2 Jitter buffer discard rate

The proportion of packets discarded by the receiving jitter buffer since the start of transmission expressed as an 8-bit binary fraction obtained by dividing the number of packets discarded by the total number of packets expected and multiplying this value by 256 and taking the integer part.

B.5.3 Burst loss density

The average proportion of packets both lost and discarded during burst periods expressed as an 8-bit binary fraction. This is obtained by dividing the sum of the number of packets lost in the transmission path and discarded by the jitter buffer during burst periods by the total number of packets expected during burst periods, multiplying this value by 256 and taking the integer part.

A burst is a period during which a high proportion of packets are either lost in transit or discarded due to late arrival. In general, a burst is likely to result in audible degradation to call quality.

A burst is defined as a longest sequence that:

a) starts with a lost or discarded packet;

b) does not contain any occurrences of Gmin or more consecutive received (and not discarded) packets; and

c) ends with a lost or discarded packet.

B.5.4 Gap loss density

The average proportion of packets lost and discarded during gap periods expressed as an 8-bit binary fraction. This is obtained by dividing the sum of the number of packets lost in the transmission path and discarded by the jitter buffer during gap periods by the total number of packets expected during gap periods, multiplying this value by 256 and taking the integer part.

A gap is a period between bursts; the above burst definition means that during gaps the packet loss rate is low and lost/discarded packets are isolated and hence can be effectively masked by packet loss concealment algorithms.

B.5.5 Burst duration

The average length of burst periods expressed in milliseconds.

B.5.6 Gap duration

The average length of gap periods expressed in milliseconds.

B.5.7 RTCP round-trip delay

The round-trip delay between RTP instances, expressed in milliseconds.

B.5.8 End system delay

The end system delay, comprising encode, decode and jitter buffer delay, expressed in milliseconds. This may be combined with the RTCP Round-Trip Delay to estimate the overall Voice over IP segment round-trip delay.

B.5.9 Signal level

The ratio of the signal level to a 0 dBm0 reference, expressed in dBm0.

B.5.10 Noise level

The ratio of the silent period background noise level to a 0 dBm0 reference, expressed in dBm0.

B.5.11 Residual echo return loss

The echo return loss after the effects of echo cancellation, expressed in dB.

B.5.12 Gmin

A parameter used to define bursts. This is by default set to 16, which sets the threshold packet loss rate between bursts and gaps to approximately 6%.

B.5.13 R factor

A value representing the receiving end call quality of this stream, calculated per ITU-T Rec. G.107. Table 1/G.108 provides interpretive information about the value of the R factor.

B.5.14 External R factor

A value representing the effects of an externally connected network, calculated per ITU-T Rec. G.107. Table 1/G.108 provides interpretive information about the value of the R factor.

B.5.15 Estimated MOS-LQ

An estimated receiving end Listening Quality MOS, calculated per ITU-T Rec. G.107 and multiplied by 10.

B.5.16 Estimated MOS-CQ

An estimated receiving end Conversational Quality MOS, calculated per ITU-T Rec. G.107 and multiplied by 10.

B.5.17 Packet loss concealment type

Type of packet loss concealment algorithm in use. Indicates unknown, silence insertion, "standard" (i.e., per appropriate ITU Recommendation) and "enhanced".

B.5.18 Jitter buffer type

Indication of whether the receiving jitter buffer is fixed, adaptive or unknown.

B.5.19 Jitter buffer adaptation rate

The adaptation rate for adaptive jitter buffers. This is defined as the time taken in milliseconds adjusting to a step from 30 ms to 100 ms in peak-to-peak jitter divided by twice the frame size in milliseconds. This value need only be provided if the Jitter Buffer Type is specified.

B.5.20 Jitter buffer nominal size

The current nominal jitter buffer size for fixed or adaptive jitter buffers. This is expressed in milliseconds. This value need only be provided if the Jitter Buffer Type is specified.

B.5.21 Jitter buffer max size

The maximum jitter buffer size in milliseconds for fixed or adaptive jitter buffers. This value need only be provided if the Jitter Buffer Type is specified.

B.5.22 Jitter buffer absolute max

The maximum achievable size of the buffer in milliseconds for adaptive jitter buffers. This value need only be provided if the Jitter Buffer Type is specified.

B.6 ASN.1 definition

```
QOS-MONITORING-EXTENDED-VOIP-REPORT   DEFINITIONS  AUTOMATIC TAGS ::=
BEGIN
IMPORTS
   GenericIdentifier   FROM   H323-MESSAGES;
ExtendedRTPMetrics      ::= SEQUENCE
{
     networkPacketLossRate   INTEGER (0..255)   OPTIONAL,
     jitterBufferDiscardRate INTEGER (0..255)   OPTIONAL,
     burstMetrics            BurstMetrics       OPTIONAL,
     rtcpRoundTripDelay      INTEGER (0..65535) OPTIONAL,
     endSystemDelay          INTEGER (0..65535) OPTIONAL,
     signalLevel             INTEGER (-127..10) OPTIONAL,
     noiseLevel              INTEGER (-127..0)  OPTIONAL,
     residualEchoReturnLoss  INTEGER (0..127)   OPTIONAL,
     rFactor                 INTEGER (0..100)   OPTIONAL,
     extRFactor              INTEGER (0..100)   OPTIONAL,
     estimatedMOSLQ          INTEGER (10..50)   OPTIONAL,
     estimatedMOSCQ          INTEGER (10..50)   OPTIONAL,
     plcType                 PLCtypes           OPTIONAL,
     jitterBufferParms       JitterBufferParms  OPTIONAL,
     ...
}
BurstMetrics ::= SEQUENCE
{
     gmin                    INTEGER (0..255)   OPTIONAL,
     burstLossDensity        INTEGER (0..255)   OPTIONAL,
     gapLossDensity          INTEGER (0..255)   OPTIONAL,
     burstDuration           INTEGER (0..65535) OPTIONAL,
     gapDuration             INTEGER (0..65535) OPTIONAL,
     ...
}
PLCtypes ::= CHOICE
{
     unspecified   NULL,
     disabled      NULL,
     enhanced      NULL,
     standard      NULL,
     ...
}

JitterBufferParms ::= SEQUENCE
{
     jitterBufferType        JitterBufferTypes  OPTIONAL,
     jitterBufferAdaptRate   INTEGER (0..15)    OPTIONAL,
     jitterBufferNominalSize INTEGER (0..65535) OPTIONAL,
     jitterBufferMaxSize     INTEGER (0..65535) OPTIONAL,
     jitterBufferAbsoluteMax INTEGER (0..65535) OPTIONAL,
     ...
}
JitterBufferTypes ::= CHOICE
{
     unknown       NULL,
     reserved      NULL,
     nonadaptive   NULL,
     adaptive      NULL,
     ...
}
qosMonitoringExtendedRTPMetrics    GenericIdentifier   ::=   standard:2
END        -- of QOS-MONITORING-EXTENDED-VOIP-REPORT
```

B.7 Linkage to H.460.9 data

If present, an element of type **ExtendedRTPMetrics** shall be included together with identifier *qosMonitoringExtendedRTPMetrics* in element *extensions* of the **RTCPMeasures** structure, as defined in Annex A/H.460.9.

SERIES OF ITU-T RECOMMENDATIONS

Series A	Organization of the work of ITU-T
Series B	Means of expression: definitions, symbols, classification
Series C	General telecommunication statistics
Series D	General tariff principles
Series E	Overall network operation, telephone service, service operation and human factors
Series F	Non-telephone telecommunication services
Series G	Transmission systems and media, digital systems and networks
Series H	**Audiovisual and multimedia systems**
Series I	Integrated services digital network
Series J	Cable networks and transmission of television, sound programme and other multimedia signals
Series K	Protection against interference
Series L	Construction, installation and protection of cables and other elements of outside plant
Series M	TMN and network maintenance: international transmission systems, telephone circuits, telegraphy, facsimile and leased circuits
Series N	Maintenance: international sound programme and television transmission circuits
Series O	Specifications of measuring equipment
Series P	Telephone transmission quality, telephone installations, local line networks
Series Q	Switching and signalling
Series R	Telegraph transmission
Series S	Telegraph services terminal equipment
Series T	Terminals for telematic services
Series U	Telegraph switching
Series V	Data communication over the telephone network
Series X	Data networks and open system communications
Series Y	Global information infrastructure, Internet protocol aspects and Next Generation Networks
Series Z	Languages and general software aspects for telecommunication systems

Printed in Switzerland
Geneva, 2004