INTERNATIONAL TELECOMMUNICATION UNION

ITU-T H.460.9

TELECOMMUNICATION
STANDARDIZATION SECTOR
OF ITU

(11/2002)

SERIES H: AUDIOVISUAL AND MULTIMEDIA SYSTEMS

Supplementary services for multimedia

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

ITU-T Recommendation H.460.9

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.399
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

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

Summary

This Recommendation describes the procedures and the signalling protocol for reporting online Quality of Service (QoS) monitoring on a call-by-call basis within the H.323 system. This monitoring is performed by the end-points based on real time traffic statistics and reported to the gatekeeper at periodic intervals, or at call conclusion. These procedures use the H.323 Generic Extensibility Framework (GEF) and therefore do not require any changes to the base standards.

Source

ITU-T Recommendation H.460.9 was prepared by ITU-T Study Group 16 (2001-2004) and approved under the WTSA Resolution 1 procedure on 29 November 2002.

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.

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

1 Scope

This Recommendation describes the procedures and the signalling protocol based on H.225.0/RAS for reporting of per call, online Quality of Service (QoS) monitoring.

QoS for real-time traffic like Voice over IP (VoIP) relies on the QoS of the underlying IP-network. The VoIP user may experience bad voice quality if the network does not meet the requirements for real-time traffic in respect to end-to-end delay, network jitter, and packet loss. Currently, dedicated network probes have to be installed in the network. This Recommendation proposes that the H.323 endpoints capture QoS-monitoring measures, as provided by means of Real-time Transport Control Protocol (RTCP – RFC 1889) per call for a period of time, and forward this information periodically and/or at the end of the call to the gatekeeper.

NOTE – RTCP conveys feedback about RTP streams directly between endpoints; the gatekeeper does not get this information. Therefore, the gatekeeper has no way of judging the quality of media channels directly.

At the gatekeeper this information can be processed and registered per call, e.g. in the Call Detail Record (CDR), or be forwarded to a dedicated network element for post processing. The registration and post processing of these reports are beyond the scope of this Recommendation. Based on this information the service provider can take appropriate actions if these reports show insufficient QoS for real-time traffic at certain communication relations.

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.

2.1 Normative references

[1] ITU-T Recommendation H.323 Version 4 (2000), *Packet-based multimedia communications systems*.

[2] ITU-T Recommendation H.225.0 Version 4 (2000), *Call signalling protocols and media stream packetization for packet-based multimedia communication systems*.

[3] ITU-T Recommendation X.680 (2002), *Information technology – Abstract Syntax Notation One (ASN.1): Specification of basic notation*.

[4] ITU-T Recommendation X.691 (2002), *Information technology – ASN.1 encoding rules: Specification of Packed Encoding Rules (PER)*.

[5] ITU-T Recommendation H.460.1 (2002), *Guidelines for the use of the Generic Extensible Framework (GEF)*.

[6] ITU-T Recommendation H.225.0 Annex G (1999), *Communication between administrative domains*.

[7] ITU-T Recommendation H.501 (2002), *Protocol for mobility management and intra/inter-domain communication in multimedia systems*.

[8] IETF RFC 1889 (1996), *RTP: A Transport Protocol for Real-Time Applications*.

2.2 Informative reference

– IETF RFC 1305 (1992), *Network Time Protocol (Version 3) Specification, Implementation and analysis*.

3 Abbreviations and Acronyms

This Recommendation uses the following abbreviations:

ACF (RAS) Admission Confirm
ARQ (RAS) Admission Request
ASN.1 Abstract Syntax Notation One
CDR Call Detail Record
DCF (RAS) Disengage Confirm
DRQ (RAS) Disengage Request
GEF Generic Extensible Framework
IETF Internet Engineering Task Force
IRR (RAS) Information Request Response
NTP Network Time Protocol (RFC 1305)
PER (ASN.1) Packed Encoding Rules
QoS Quality of Service
RAS Registration, Admission and Status
RCF (RAS) Registration Confirm
RFC Request for Comments
RR (RTCP) Receiver Report
RRQ (RAS) Registration Request
RTCP RTP Control Protocol
RTP Real-time Transport Protocol
RTT Round Trip Time
SR (RTCP) Sender Report
UDP User Datagram Protocol

4 Feature description

4.1 Negotiation of the feature

Usage of the QoS-monitoring reporting feature is negotiated between endpoint and gatekeeper at call set-up time as part of the admission request procedure. For this purpose, an endpoint that supports this feature shall include the feature descriptor defined in 4.4 (see Table 1) in **featureSet.supportedFeatures** of the ARQ message.

If the gatekeeper also supports this feature, it may include the same feature descriptor in the **featureSet** of the ACF message, either as **neededFeature** if the feature is required for the call, as **desiredFeature** if the feature is only desired, or as **supportedFeature** if the gatekeeper does not

care. Omission of the feature indicator from the ACF message means that this feature is not supported or its use is not wanted for this call.

An endpoint may also indicate support of this feature when registering with a gatekeeper, by including the feature descriptor in the RRQ message in the same way as described above for ARQ. Likewise, a gatekeeper may indicate its support by including the feature descriptor in the RCF message.

4.2 The QoS-monitoring reporting endpoint procedure

4.2.1 Initiation of the procedure

The procedure is initiated during the H.225.0 Admission procedure. The Admission Confirm (ACF) message contains the following fields that will be set as part of the procedure:

featureSet – Depending on the content of this field the endpoint:

- shall include QoS-monitoring report parameters in the **genericData** field of IRR message(s) and/or the DRQ/DCF message if the feature is indicated in **neededFeatures**;
- should include QoS-monitoring report parameters in the **genericData** field of IRR message(s) and/or the DRQ/DCF message if the feature is indicated in **desiredFeatures**;
- may include QoS-monitoring report parameters in the **genericData** field of IRR message(s) and/or the DRQ/DCF message if the feature is indicated in **supportedFeatures**;
- shall not include any QoS-monitoring report parameters in IRR or DRQ/DCF message(s) if the feature is not indicated at all in **featureSet**.

irrFrequency – The frequency, in seconds, that the endpoint shall send Information Request Response (IRR) messages to the gatekeeper during a call. The frequency, in seconds, is an implementation – dependent time interval for gathering QoS-monitoring statistics. Since RTCP packets are generated at a rate of one packet per five seconds for unicast point-to-point connections at a random variation of [0.5 … 1.5] times, the time interval should not be less than eight seconds in order to guarantee that the statistics will be meaningful.

4.2.2 Continuation of the procedure

For each media connection a unidirectional RTP data flow is generated by the endpoint. At a separate transport address (default: UDP/RTP-port number plus one) RTCP messages are exchanged containing feedback information from which network transport service QoS can be derived. See RFC 1889 for details regarding RTP and RTCP. For the time interval specified in **irrFrequency** of the Admission Confirm (ACF) message and conveyed during the initiation procedure, this information shall be gathered and statistically processed in terms of short-term and long-term measures. The resulting values shall be populated and included in the **genericData** field of Information Request Response (IRR) messages that are issued to the gatekeeper at a frequency of **irrFrequency**.

If the optional parameter **qosMonitoringFinalOnly** was included when the feature was initiated, then no QoS monitoring reporting data shall be included in any IRR messages.

NOTE – If no **irrFrequency** value was returned in the ACF message, no unsolicited IRR messages and thus no periodic QoS-monitoring reports will be sent, but a final report will be included in DRQ or DCF if the QoS-monitoring reporting feature was initiated.

4.2.3 Termination of the procedure

When the call is released, either a Disengage Request (DRQ) message is issued by the endpoint that carries in the **genericData** field the information for the final time interval, or the endpoint includes this information in the DCF message if it received a DRQ message from the gatekeeper.

4.2.4 Single QoS-monitoring report only at the end of the call

In order to allow an endpoint to send only a summary QoS-monitoring report at the end of the call, regardless of the value of **irrFrequency**, parameter **qosMonitoringFinalOnly** may be included when negotiating the feature. The **genericData** for QoS-monitoring reporting will then be included only in the DRQ or DCF message.

4.3 The QoS-monitoring reporting gatekeeper (Annex G/H.225.0) procedure

Since one gatekeeper obtains information only from H.323 endpoints registered to it, it is necessary to exchange complementary QoS-monitoring report information for bidirectional media connections between the two gatekeepers involved in a particular call. This can be done by means of the usage information exchange procedures specified in Annex G/H.225.0, by including the **genericData** for QoS-monitoring reporting in **UsageIndication** messages. The data sent by a gatekeeper in a QoS-monitoring report are those gathered by this gatekeeper locally for a specific call from an endpoint's QoS-monitoring reports.

Exchange of **genericData** for QoS-monitoring reporting is negotiated between gatekeepers on a per-call basis as described in 4.1 above, except that **UsageRequest/UsageConfirmation** messages are used rather than ARQ/ACF. If reports are required in both directions, the usage information exchange procedures are initiated independently for each direction. The gatekeeper requesting QoS-monitoring reports indicates in the **UsageRequest** message when **UsageIndication** messages shall be sent; periodically or only at the end of the call. The interval for sending periodic reports may be different for each direction, and also different from the interval for IRR messages.

Refer to ITU-T Rec. H.501 for details regarding the usage report messages.

4.4 Generic data usage

The Generic Extensibility Framework (GEF) shall be used to convey QoS-monitoring reports with H.225.0/RAS between H.323 endpoint and gatekeeper, and/or between gatekeepers as described in tabular form in the following.

The reports may include:

- End-to-end delay as sender (estimated from round trip time for RTCP packets);
- Packet Loss and Network Jitter as receiver;
- Optionally additional data, e.g. for a specific profile (see 4.5 for more details).

Each report consists of a single parameter in raw format, containing a PER-encoded ASN.1 structure as specified in Annex A.

Table 1/H.460.9 – Tabular specification of the QoS-monitoring reporting feature

Feature name:	QoS-monitoring Reporting
Feature Description:	QoS-monitoring reporting
Feature identifier type:	Standard
Feature identifier value:	9: qosMonitoringReportID

qosMonitoringReportID: Identifies the QoS-monitoring report feature using the standard field of **GenericIdentifier** with a unique integer value.

Table 2/H.460.9 – Tabular specification of the qosMonitoringFinalOnly parameter

Parameter name:	qosMonitoringFinalOnly
Parameter description:	Requests QoS-monitoring report at end of call only
Parameter identifier type:	Standard
Parameter identifier value:	0
Parameter type:	no content
Parameter cardinality:	Zero or one

qosMonitoringFinalOnly: If included, QoS-monitoring reports will only be sent in the DRQ/DCF message when the call finishes.

Table 3/H.460.9 – Tabular specification of the QoS-monitoring reporting parameter

Parameter name:	qosMonitoringReportData
Parameter description:	QoS-monitoring reporting data
Parameter identifier type:	Standard
Parameter identifier value:	1
Parameter type:	Raw: ASN.1 PER
Parameter cardinality:	Zero or one

qosMonitoringReportData: This data is sent in H225.0/RAS messages; IRR, DRQ/DCF and optionally also in H.501 **UsageIndication** messages to notify the QoS-monitoring information for all established media connections. The content of this parameter is a raw field consisting of the ASN.1 PER encoded **qosMonitoringReportData** as specified in the ASN.1 notation below (see Annex A).

4.5 Extensibility

The ASN.1 data structure in Annex A provides for extensibility in three ways:

- The ASN.1 types are defined as extensible in the usual way, for additions in later versions of this Recommendation.
- Non-standard data may be included in various places, analogous to H.225.0. This allows implementation-specific data to be transparently conveyed.
- An opaque container for extensions may contain additional data, e.g. for specific applications or payload profiles.

The data that may go into the extensions container can be specified by various means – an annex to this Recommendation, another Recommendation, a non-standard specification etc. Each extension is uniquely identified by a **GenericIdentifier** and contains its data in form of an octet string. Extensions defined in the context of this Recommendation may use **GenericIdentifiers** of type standard, while other extensions will use type oid (i.e. object identifiers).

5 Description of ASN.1 types and fields

Only fields introduced by this Recommendation are explained here. Other fields are adopted from H.225.0. For the detailed determination of the values, please refer to Annex A/H.225.0.

Some values can or will only be reported by a media sender or receiver. For bidirectional media connections, the complete picture will be obtained only if the information from each side is combined.

5.1 End-to-End delay measures

These measurements can only be performed by a media sender. Therefore, this information can only be provided by the media sender.

EstimatedEnd2EndDelay – Value derived from the RTCP RTT (see Figure A.2/H.225.0 for the determination of RTT) divided by 2. Only the middle 32 bits of the 64-bit NTP timestamp are used, as described in A.4/H.225.0.

meanEstimatedEnd2EndDelay – Mean value of **EstimatedEnd2EndDelay** for the currently reported time interval.

worstEstimatedEnd2EndDelay – Worst case value of **EstimatedEnd2EndDelay** for the currently reported time interval.

5.2 Packet loss measures

This information shall only be provided by the media receiver because it reflects the QoS as perceived by the user receiving the media.

cumulativeNumberOfPacketsLost – Cumulative number of packets lost of last sent RTCP Receiver Report (RR).

packetLostRate – Difference between Cumulative number of packets lost of last sent RR of last time interval and last sent RR of currently reported time interval per second.

fractionLostRate – Sum of Fraction lost of all sent RRs for the currently reported interval per second.

5.3 Throughput measures

These measurements can only be performed by a media receiver. Therefore, this information can only be provided by a media receiver.

estimatedThroughput – Difference between Sender's packet count of last received RTCP Sender Report (SR) of last time interval and last received SR of currently reported time interval reduced by **packetLostRate** (see 5.2) multiplied with the average packet size of the packet (excluding the link layer protocol) per second.

NOTE – The packet size may differ during media transmission. Only the maximum number of media samples is fixed.

5.4 Network jitter measures

This information shall only be provided by the media receiver because it reflects the QoS as perceived by the user receiving the media.

CalculatedJitter – Corresponds to Interarrival Jitter of sent RTCP Receiver Report (RR) in timestamp units presented as 32-bit integer value of 64-bit NTP timestamp (see A.4/H.225.0).

meanJitter – Average value of **CalculatedJitter** calculated from all sent RRs for the currently reported time interval.

worstJitter – Worst case value of **CalculatedJitter** from all sent RRs for the currently reported time interval.

Annex A

ASN.1 definitions

```
QOS-MONITORING-REPORT DEFINITIONS AUTOMATIC TAGS ::=
BEGIN
IMPORTS
   NonStandardParameter,
   TransportChannelInfo,
   CallIdentifier,
   CallReferenceValue,
   ConferenceIdentifier,
   BandWidth,
   GenericIdentifier
FROM H323-MESSAGES;

EstimatedEnd2EndDelay  ::= INTEGER (0..4294967295)
                           -- middle 32 bits of 64 bit NTP timestamp

CalculatedJitter       ::= INTEGER (0..4294967295)
                           -- measured in timestamp units

Extension              ::= SEQUENCE
{
    extensionId            GenericIdentifier,
    extensionContent       OCTET STRING  OPTIONAL,
    ...
}

RTCPMeasures           ::= SEQUENCE
{
    rtpAddress             TransportChannelInfo,
    rtcpAddress            TransportChannelInfo,
    sessionId              INTEGER (1..255),
    nonStandardData        NonStandardParameter OPTIONAL,
    mediaSenderMeasures    SEQUENCE
    {
         worstEstimatedEnd2EndDelay    EstimatedEnd2EndDelay  OPTIONAL,
         meanEstimatedEnd2EndDelay     EstimatedEnd2EndDelay  OPTIONAL,
         ...
    }  OPTIONAL,
    mediaReceiverMeasures  SEQUENCE
    {
         cumulativeNumberOfPacketsLost INTEGER (0..4294967295)  OPTIONAL,
         packetLostRate                INTEGER (0..65535)  OPTIONAL,
         worstJitter                   CalculatedJitter  OPTIONAL,
         estimatedThroughput           BandWidth  OPTIONAL, -- in 100s of bits
         fractionLostRate              INTEGER (0..65535)  OPTIONAL,
         meanJitter                    CalculatedJitter  OPTIONAL,
         ...
    }  OPTIONAL,
    extensions             SEQUENCE OF Extension  OPTIONAL,
    ...
}
```

```
PerCallQoSReport   ::= SEQUENCE
{
    nonStandardData        NonStandardParameter  OPTIONAL,
    callReferenceValue     CallReferenceValue,
    conferenceID           ConferenceIdentifier,
    callIdentifier         CallIdentifier,
    mediaChannelsQoS       SEQUENCE OF RTCPMeasures  OPTIONAL,
                               -- one element per media channel
    extensions             SEQUENCE OF Extension  OPTIONAL,
    ...
}

QosMonitoringReportData ::= CHOICE
{
    periodic     PeriodicQoSMonReport,   -- included in IRR messages
    final        FinalQosMonReport,      -- included in DRQ/DCF message
    interGK      InterGKQosMonReport,    -- included in usageIndication message
    ...
}

PeriodicQoSMonReport   ::= SEQUENCE
{
    perCallInfo        SEQUENCE OF PerCallQoSReport,   -- one element per call
    extensions         SEQUENCE OF Extension  OPTIONAL,
    ...
}

FinalQosMonReport      ::= SEQUENCE
{
    mediaInfo          SEQUENCE OF RTCPMeasures,     -- one element per channel
    nonStandardData    NonStandardParameter  OPTIONAL,
    extensions         SEQUENCE OF Extension  OPTIONAL,
    ...
}

InterGKQosMonReport    ::= SEQUENCE
{
    mediaInfo          SEQUENCE OF RTCPMeasures,     -- one element per channel
    nonStandardData    NonStandardParameter  OPTIONAL,
    extensions         SEQUENCE OF Extension  OPTIONAL,
    ...
}

--
-- H.460.9 Identifiers: --
--
qosMonitoringReportID      GenericIdentifier  ::= standard:9
qosMonitoringFinalOnly     GenericIdentifier  ::= standard:0
qosMonitoringReportData    GenericIdentifier  ::= standard:1

END    -- of ASN.1
```

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 and Internet protocol aspects
Series Z	Languages and general software aspects for telecommunication systems

Printed in Switzerland
Geneva, 2003