Network Management
White Paper

Iris Performance Intelligence
Media KPI Algorithms

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REVISION HISTORY

Revision	Date	Author	Description
0.1	19 April 2010	Alessio Biasutto	Final for 7.10.2 release
0.2	21 April 2010	Alessio Biasutto	Reviewed
0.3	10 May 2010	Alessio Biasutto	Added H.248 Packets Out of Sequence KPIs
0.4	25 June 2010	Alessio Biasutto	Added SIP Publish KPIs
1.0	25 November 2010	Pierantonio Bottaro	- Updated SIP EoCQ KPIs - Removed Total Packet Out of Sequence from H248, RTCP, SIP; only on RTP - One Way calls on RTP only - Updated H248 XNQ: Removed: NE/FE Sum IPDV NE/FE Average IPDV NE/FE Sum IPDV Cycles NE/FE Average IPDV Cycles NE/FE Sum Jitter Buffer Adaptation Events Added : NE/FE Reporting Sessions NE/FE Reporting Sessions Duration (s) NE/FE Normalized IPDV Sum NE/FE Average Global Maximum IPDV (ms) FE Normalized Jitter Buffer Adaptation Events
1.1	12 April 2011	Pierantonio Bottaro	Updated SIP EoCQ section with call failed formula
2.0	11 May 2011	Pierantonio Bottaro	Updated with 7.11.1 GA Added MGCP EoCQ.
3.0	1 September 2011	Pierantonio Bottaro	Updated with 7.11.2 GA SIP EoCQ draft13 support.
4.0	23 December	Pierantonio	Updated with 7.11.3 GA

	2011	Bottaro	New KPIs for RTPC_XR. Introduced One Way Call into RTP
4.1	16 May 2012	Pierantonio Bottaro	Updated removing Total Packet Out of Sequence from RTP
4.2	28 May 2012	Pierantonio Bottaro	Updated with 7.12.1 contents: - Added RTSP support
5.0	5 November 2012	Pierantonio Bottaro	Updated with 7.12.2 contents: - Added Packet Out of Sequence KPI in RTP
5.1	21 November 2012	Pierantonio Bottaro	Updated after QA check about Packet Out of Sequence KPI in RTP
6.0	22 May 2013	Eric Williamson	Updated with 7.13.1 content: • Timeout KPIs for SIP EoCQ, H.248, MGCP, RTP, RTCP, and RTCP-XR protocols

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INTRODUCTION

The intent of this document is to describe the Tektronix' Media Key Performance Indicators (KPIs) delivered in IPI on SIP, H.248, RTP, MGCP and RTCP protocols.

This document will be modified periodically as new KPIs/KQIs are added to the product.

GENERAL

The summations (Σ) used in the KPIs are aggregations over the selected time interval. This time interval is configurable, for example it can be set to 5 minutes.

IPI SIP KEY PERFORMANCE INDICATORS

We will refer to the following call flow diagrams through out the KPI discussions. Please note that these call flows have been simplified for the KPI discussion. For example, often the Reply message is omitted if its inclusion is not necessary in the relevant discussion. KPIs are calculated on Source and Destination nodes considering the direction of the call establishment

Figure 1. Call Flow – Off Hook to Ringing

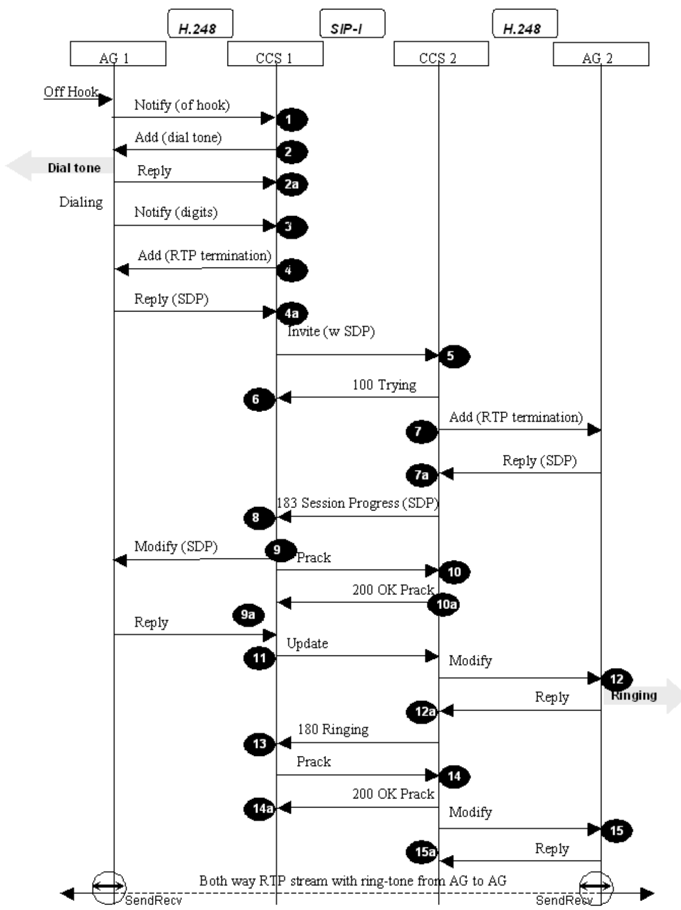


Figure 2. Call Flow – Call Answer

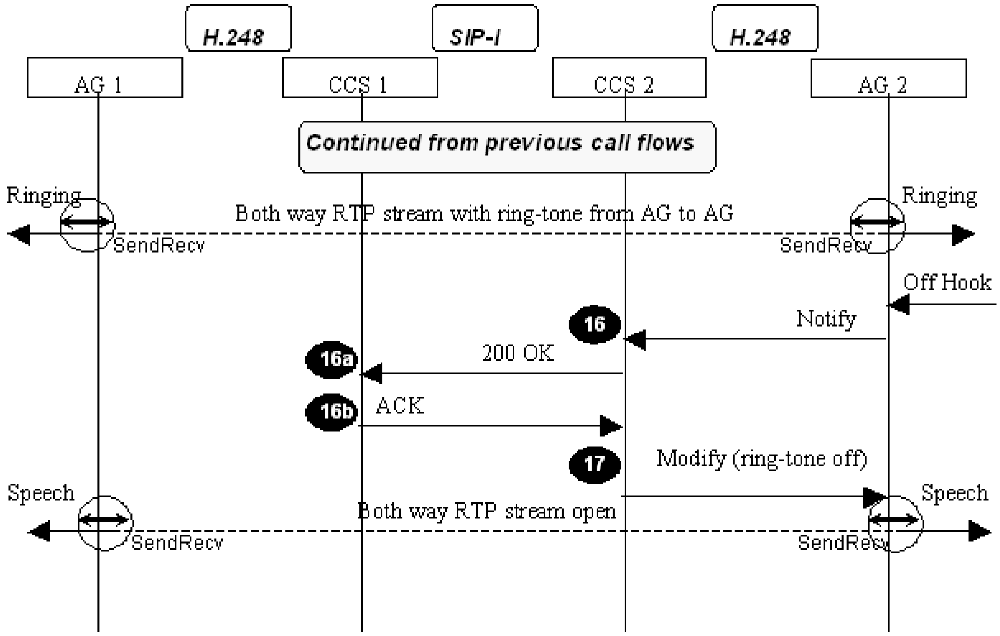
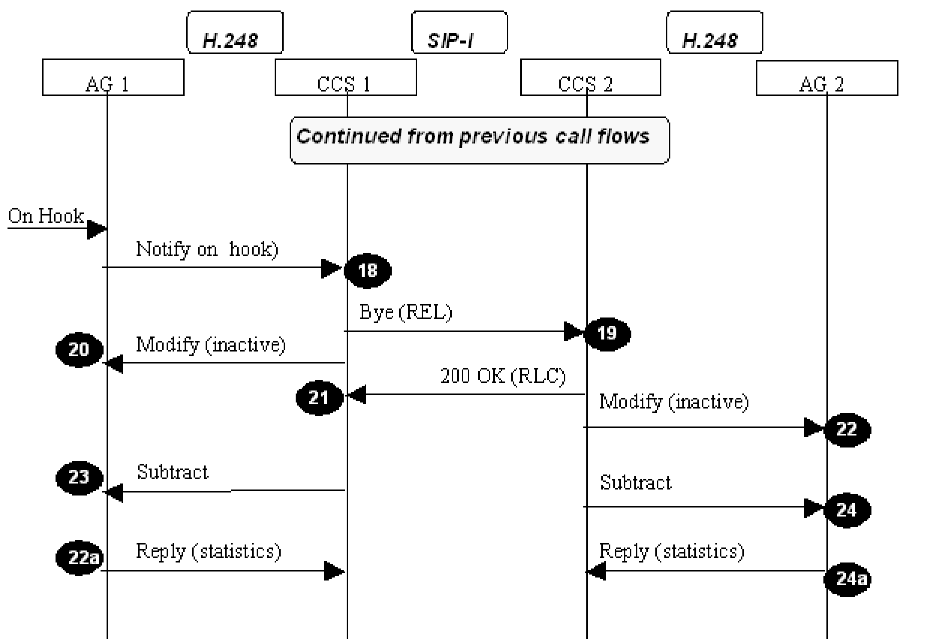


Figure 3. Call Flow – Call Release



SIP EoCQ

Before releasing a call, EOCQ KPIs can be carried within the SIP Publish message. These KPIs are defined in the standard IETF draft-ietf-sipping-rtcp-summary-13.

IPI provides the Worst and Average value of the metrics defined in the IETF document for Local and Remote as detailed in the following section of this document for each metric.

For a specific "X" metric, the Average value is provided with the following formula:

= Σ (value of X in the selected period) / (Number of calls providing the X metric in the selected period).

The Worst value for the following Local or Remote metrics is calculated as the minimum value in the selected period:

- Signal Level,
- MOS LQ,
- MOS CQ,
- R Factor Listening Quality,
- R Factor Conversational Quality,
- External R Factor In,
- External R Factor Out

For all the other Remore or Local metrics, the Worst valus is calculated as the maximum in the selected period.

Session Attempts

Description:

Counts the number of Media Streams reported in SIP EOCQ during the measured period

Σ Session Attempts

Session Completions

Description:

Counts the number of Media Streams that completed normally during the measured period.

Σ Media Session Completed

A Media Session Completion is determined in the signaling by looking at the reported G10 Timeouts, and the SIP Response Code on the SIP Leg. The Application can be configured so that all SIP Response Codes and Timeouts can be mapped to Success, Failure, or Timeout.

NOTE: When a Session is successful based on response code, then it is classified based on MOS_LQ , and it is successful when:

- if Remote and Local MOS LQ are available, both are Fair, Good or Excellent
- if only Remote MOS LQ is available, it is Fair, Good or Excellent
- if only Local MOS LQ is available, it is Fair, Good or Excellent

Session Failures

Description:

Counts the number of Media Streams that failed normally during the measured period.

Σ Media Session Failed

A Media Session Failure is determined in the signaling by looking at the reported G10 Timeouts, and the SIP Response Code on the SIP Leg. The Application can be configured so that all SIP Response Codes and Timeouts can be mapped to Success, Failure, or Timeout.

NOTE: When a Session is successful based on response code, then it is classified based on MOS_LQ , and it is failed when:

- Remote or Local MOS LQ is Poor or Unacceptable.
- both Remote and Local MOS LQ IE are NOT available. In this case, the session will have response cause "No MOS Available". By default, such value is mapped into failures, but it is possible to configure IPI to put such response cause as a success.

Session Failure Percentage (Session Failure %)

Description:

Computes the percentage of Media Streams that failed normally during the measured period.

(Σ Failures) / (Σ Session Attempts) * 100

Session Timeouts

Description:

Counts the number of Media Streams that timed out normally during the measured period.

Σ Media Session Timeouts

A Media Session Timeout is determined in the signaling by looking at the reported G10 Timeouts, and the SIP Response Code on the SIP Leg. The Application can be configured so that all SIP Response Codes and Timeouts can be mapped to Success, Failure, or Timeout.

Session Timeout Percentage (Session Timeout %)

Description:

Computes the percentage of Media Streams that timed out normally during the measured period.

(Σ Timeouts) / (Σ Session Attempts) * 100

Jitter Buffer Nominal

This value corresponds to "JB nominal" in RFC3611 in the VoIP Metrics Report Block. This is the current nominal jitter buffer delay in milliseconds, which corresponds to the nominal jitter buffer delay for packets that arrive exactly on time.

For this metric, IPI provides Average and Worst value for both Local and Remote.

Jitter Buffer Max

This value corresponds to "JB maximum" in RFC3611 in the VoIP Metrics Report Block. This is the current maximum jitter buffer delay in milliseconds which corresponds to the earliest arriving packet that would not be discarded.

For this metric, IPI provides Average and Worst value for both Local and Remote.

Jitter Buffer Absolute Max

This value corresponds to "JB abs max" in RFC3611 in the VoIP Metrics Report Block. This is the absolute maximum delay in milliseconds that the adaptive jitter buffer can reach under worst case conditions.

For this metric, IPI provides Average and Worst value for both Local and Remote.

Network Packet Loss Rate

Network Packet Loss corresponds to "loss rate" in RFC3611 in the VoIP Metrics Report Block. It is the fraction of RTP data packets from the source lost since the beginning of reception.

For this metric, IPI provides Average and Worst value for both Local and Remote.

Jitter Buffer Discard Rate

This value corresponds to "discard rate" in RFC3611 in the VoIP Metrics Report Block. It is the fraction of RTP data packets from the source that have been discarded since the beginning of reception, due to late or early arrival, under-run or overflow at the receiving jitter buffer.

For this metric, IPI provides Average and Worst value for both Local and Remote.

Burst Density

This value corresponds to "burst density" in RFC3611 in the VoIP Metrics Report Block. Is is the fraction of RTP data packets within burst periods since the beginning of reception that were either lost or discarded.

For this metric, IPI provides Average and Worst value for both Local and Remote.

Burst Duration

This value corresponds to "burst duration" in RFC3611 in the VoIP Metrics Report Block. It is the mean duration, expressed in milliseconds, of the burst periods that have occurred since the beginning of reception.

For this metric, IPI provides Average and Worst value for both Local and Remote.

Gap Loss Density

This value corresponds to "gap density" in RFC3611 in the VoIP metrics Report Block. It is the fraction of RTP data packets within inter-burst gaps since the beginning of reception that were either lost or discarded.

For this metric, IPI provides Average and Worst value for both Local and Remote.

Gap Duration

This value corresponds to "gap duration" in RFC3611 in the VoIP Metrics Report Block. It is the mean duration, expressed in milliseconds, of the gap periods that have occurred since the beginning of reception.

For this metric, IPI provides Average and Worst value for both Local and Remote.

Round Trip Delay

This value corresponds to "round trip delay" in RFC3611 in the VoIP Metrics Report. The parameter is expressed in milliseconds. It is the most recently calculated round trip time between RTP interfaces, expressed in milliseconds.

For this metric, IPI provides Average and Worst value for both Local and Remote.

End System Delay

This value corresponds to "end system delay" in RFC3611 in the VoIP Metrics Report Block. The parameter is expressed in milliseconds. It is the most recently estimated end system delay, expressed in milliseconds.

For this metric, IPI provides Average and Worst value for both Local and Remote.

One Way Delay

This value SHOULD be measured using the methods defined in IETF RFC 2679. The parameter is expressed in milliseconds.

For this metric, IPI provides Average and Worst value for both Local and Remote.

Symmetric One Way Delay

This value is computed by adding Round Trip Delay to the local and remote End System Delay and dividing by two.

For this metric, IPI provides Average and Worst value for both Local and Remote.

Interarrival Jitter

This value SHOULD be measured using the methods defined in IETF RFC 2679. The parameter is expressed in milliseconds.

For this metric, IPI provides Average and Worst value for both Local and Remote.

Mean Absolute Jitter

It is recommended that MAJ be measured as defined in ITU-T G.1020. This parameter is often referred to as MAPDV. The parameter is expressed in milliseconds.

For this metric, IPI provides Average and Worst value for both Local and Remote.

Signal Level

It is recommended that MAJ be measured as defined in ITU-T G.1020. This parameter is often referred to as MAPDV. The parameter is expressed in milliseconds. The noise level is defined as the ratio of the silent period background noise level to a 0 dBm0 reference, expressed in decibels as a signed integer in two's complement form.

For this metric, IPI provides Average and Worst value for both Local and Remote.

Noise Level

This field corresponds to "noise level" in RFC3611 in the VoIP Metrics Report Block. This field provides the ratio of the silent period background noise level to a 0 dBm0 reference, expressed in decibels.

For this metric, IPI provides Average and Worst value for both Local and Remote.

Residual Echo Return Loss

This field corresponds to "RERL" in RFC3611 in the VoIP Metrics Report Block. This field provides the ratio between the original signal and the echo level in decibels, as measured after echo cancellation or suppression has been applied.

For this metric, IPI provides Average and Worst value for both Local and Remote.

R Factor Listening Quality

It refers to ListeningQualityR. This field reports the listening quality expressed as an R factor (per G.107). This does not include the effects of echo or delay. The range of R is 0-95 for narrowband calls and 0-120 for wideband calls. Algorithms for computing this value SHOULD be compliant with ITU-T recommendations P.564 [10] and G.107 [11].

For this metric, IPI provides Average and Worst value for both Local and Remote.

R Factor Conversational Quality

It refers to ConversationalQualityR . This field corresponds to "R factor" in RFC3611 in the VoIP Metrics Report Block. This parameter provides a cumulative measurement of voice quality from the start of the session to the reporting time.

The range of R is 0-95 for narrowband calls and 0-120 for wideband calls. Algorithms for computing this value SHOULD be compliant with ITU-T Recommendation P.564 and G.107.

For this metric, IPI provides Average and Worst value for both Local and Remote.

External R Factor In

This field corresponds to "ext. R factor" in RFC3611 in the VoIP Metrics Report Block. This parameter reflects voice quality as measured by the local endpoint for incoming connection on "other" side (refer to RFC3611 for a more detailed explanation). The range of R is 0-95 for narrowband calls and 0-120 for wideband calls. Algorithms for computing this value SHOULD be compliant with ITU-T Recommendation P.564 and G.107.

For this metric, IPI provides Average and Worst value for both Local and Remote.

External R Factor Out

It refers to ExternalR-Out.This field corresponds to "ext. R factor" in RFC3611 in the VoIP Metrics Report Block. Here, the value is copied from RTCP XR message received from the remote endpoint on "other" side of this endpoint refer to RFC3611 for a more detailed explanation). The range of R is 0-95 for narrowband calls and 0-120 for wideband calls.

For this metric, IPI provides Average and Worst value for both Local and Remote.

MOS-LQ

This field corresponds to "MOSLQ" in RFC3611 in the VoIP Metrics Report Block. This parameter is the estimated mean opinion score for listening voice quality on a scale from 1 to 5, in which 5 represents "Excellent" and 1 represents "Unacceptable". Algorithms for computing this value SHOULD be compliant with ITU-T Recommendation P.564 [10]. This field provides a text name for the algorithm used to estimate MOS-LQ.

For this metric, IPI provides Average and Worst value for both Local and Remote.

MOS-CQ

This field corresponds to "MOSCQ" in RFC3611 in the VoIP Metrics Report Block. This parameter is the estimated mean opinion score for conversation voice quality on a scale from 1 to 5, in which 5 represents excellent and 1 represents unacceptable. Algorithms for computing this value SHOULD be compliant with ITU-T Recommendation P.564 with regard to the listening quality element of the computed MOS score.

For this metric, IPI provides Average and Worst value for both Local and Remote.

Jitter Buffer Increase

This is a custom specific metric not specified into IETF draft-ietf-sipping-rtcp-summary-13.

Jitter Buffer Increase is the counter of instances of de-jitter buffer growth event.

For this metric, IPI provides Local Average and Worst value.

Jitter Buffer Decrease

This is a custom specific metric not specified into IETF draft-ietf-sipping-rtcp-summary-13.

Jitter Buffer Decrease is the counter of instances of de-jitter buffer shrinkage event.

For this metric, IPI provides Local Average and Worst value.

Loss Max

This is a custom specific metric not specified into IETF draft-ietf-sipping-rtcp-summary-13.

LossMax is the maximum of percentage of loss in any given RTCP interval.

For this metric, IPI provides Average and Worst value for both Local and Remote.

Discard Max

This is a custom specific metric not specified into IETF draft-ietf-sipping-rtcp-summary-13.

DiscardMax is the maximum of percentage of discard in any given RTCP interval

For this metric, IPI provides Local Average and Worst value.

Max Absolute Jitter

This is a custom specific metric not specified into IETF draft-ietf-sipping-rtcp-summary-13.

Max_Jitter is defined in RTCP-XR RFC 3661, Sec. 4.6.

For this metric, IPI provides Local Average and Worst value.

Max Round Trip Delay

This is a custom specific metric not specified into IETF draft-ietf-sipping-rtcp-summary-13.

Max Round Trip Delay is the max value of all RTCP turnaround delay instances which is calculated .

For this metric, IPI provides Local Average and Worst value.

Max One Way Delay

This is a custom specific metric not specified into IETF draft-ietf-sipping-rtcp-summary-13.

MaxOneWayDelay is the max value of all RTCP one-way delay instances.

For this metric, IPI provides Local Average and Worst value.

Cumulative Loss

This is a custom specific metric not specified into IETF draft-ietf-sipping-rtcp-summary-13.

Cumulative Loss is the “cumulative number of packets lost” field of the RTCP packet last received.

For this metric, IPI provides Remote Average and Worst value.

NoRTCPreceived

This is a custom specific metric not specified into IETF draft-ietf-sipping-rtcp-summary-13.

No RTCP Received is a flag that indicates that No RTCP was received on the related session.

For this metric, IPI provides the Remote Average Number of sessions with No RTCP received.

Bin Counts

Description:

The Bin Count KPIs count the number of Media Sessions where the MOS LQ was between a certain range during the measured period.

Σ Session Attempts where MOS < X and MOS > Y

The Bin Counts KPI for a given measurement period records the number of Media Sessions where the MOS was between a certain range. The available Bin KPIs are the following:

- Excellent (Total number of sessions and percentage calculated as ExcellentBin/SessionAttempts)
- Good (Total number of sessions and percentage calculated as GoodBin/SessionAttempts)
- Fair (Total number of sessions and percentage calculated as FairBin/SessionAttempts)
- Poor (Total number of sessions and percentage calculated as PoorBin/SessionAttempts)
- Unacceptable (Total number of sessions and percentage calculated as UnacceptableBin/SessionAttempts)

IPI H.248 KEY PERFORMANCE INDICATORS

We will refer to the following call flow diagrams throughout the KPI discussions. Please note that these call flows have been simplified for the KPI discussion. For example, often the Reply message is omitted if its inclusion is not necessary in the relevant discussion. KPIs are calculated on Source and Destination nodes considering the direction of the call or of the stream.

Figure 4. Call Flow - Successful Call

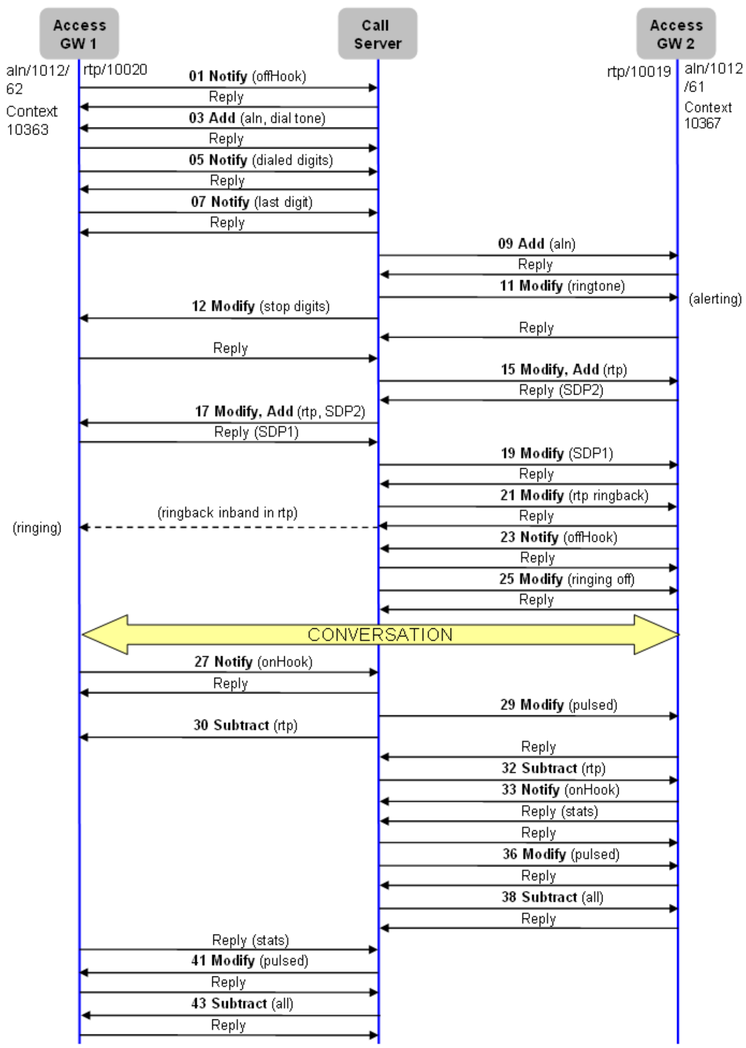
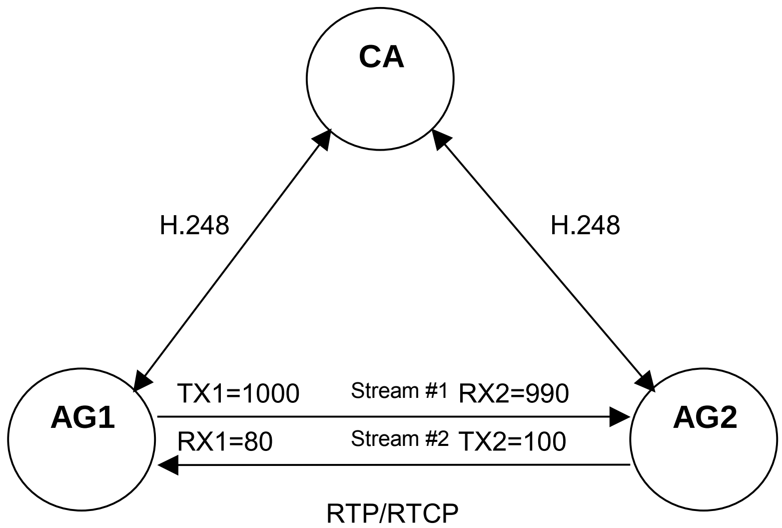


Figure 5. Call Flow – Stream Illustration



Local Node (Reporting Node)	Remote Node	Protocol	Sent	Receive d	Lost	From Call Leg
AG1	AG2	H.248 EOCQ	1000	80	20	AG1 H.248 Call Leg
AG2	AG1	H.248 EOCQ	100	990	10	AG2 H.248 Call Leg

Call QoS KPIs

Session Attempts

Description:

Counts the number of Media Streams reported in H.248 EOCQ during the measured period. This counts the number of Media H.248 EOCQ reports from a common media endpoint address.

Σ Session Attempts

A Session Attempt is determined in the signaling by the H.248 Reply to the Subtract, which has RTP statistics, this Subtract message is sent from AG to the CCA in the Call Flow. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

AG1 would be Source of 1 session attempt while AG2 would be Destination of 1 session attempt from monitoring Stream #1 in the call flow in Call Flow –Stream Illustration in Figure 5.

AG2 would be Source of 1 session attempt while AG1 would be Destination of 1 session attempt from monitoring Stream #2 in the call flow in Call Flow –Stream Illustration in Figure 5.

Session Completions

Description:

Counts the number of Media Streams that completed normally during the measured period. This counts the number of Media H.248 EOCQ reports from a common media endpoint address.

Σ Media Session Completed

A Media Session Completion is determined in the signaling by looking at the reported GeoProbe Timeouts, and the H.248 Response Code on the H.248 Leg. The Application can be configured so that all H.248 Response Codes and Timeouts can be mapped to Success, Failure, or Timeout. If the Response Code or Timeout on the H.248 Leg is 200 Normal Completion, then the Call is determined to be a Completion.

In the successful call case, in Figure 4 since there was no Response Code reported in the H.248 session, the 200 Normal Completion response code is reported.

NOTE: If the average MOS for the stream falls within the range of “Unacceptable” or “Poor” then the Session will be mapped to 8000 or 8001 respectively and counted as a Failure.

Session Failures

Description:

Counts the number of Media Streams that failed normally during the measured period. This uses the number of Media H.248 EOCQ reports from a common media endpoint address.

Σ Media Session Failed

A Media Session Failure is determined in the signaling by looking at the reported GeoProbe Timeouts, and the H.248 Response Code on the H.248 Leg. The Application can be configured so that all H.248 Response Codes and Timeouts can be mapped to Success, Failure, or Timeout. If the Response Code or Timeout on the H.248 Leg is mapped to Failure, then the Call is determined to be a Failure.

In the successful call case, in Figure 4 since there was no Response Code reported in the H.248 session, the 200 Normal Completion response code is reported.

NOTE: If the average MOS for the stream falls within the range of “Unacceptable” or “Poor” then the Session will be mapped to 8000 or 8001 respectively and counted as a Failure.

Session Failure Percentage (Session Failure %)

Description:

Computes the percentage of Media Streams that failed normally during the measured period. This uses the number of Media H.248 EOCQ reports from a common media endpoint address that completed normally.

(Σ Failures) / (Σ Session Attempts) * 100

Session Timeouts

Description:

Counts the number of Media Streams that timed out during the measured period. This uses the number of Media H.248 EOCQ reports from a common media endpoint address.

Σ Media Session Timeouts

A Media Session Timeout is determined in the signaling by looking at the reported GeoProbe Timeouts, and the H.248 Response Code on the H.248 Leg. The Application can be configured so that all H.248 Response Codes and Timeouts can be mapped to Success, Failure, or Timeout

Session Timeout Percentage (Session Timeout %)

Description:

Computes the percentage of Media Streams that timed out normally during the measured period. This uses the number of Media H.248 EOCQ reports from a common media endpoint address that completed normally.

(Σ Timeouts) / (Σ Session Attempts) * 100

Total Packets (Sent)

Description:

Counts the number of RTP Packets during the measured period. This uses the packets sent statistic from the Media H.248 EOCQ reports from a common media endpoint address.

Σ Packets Sent

A Total Packets is determined in the signaling by reporting the Packets Sent in the stream in the outgoing direction (rtp/ps) from the endpoint H.248 Reply to the Subtract, this Subtract message is sent from AG to the CCA in the Call Flow. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

Total Packets Lost

Description:

Counts the number of RTP Packets that were lost during the measured period. This uses the packets lost statistic from the Media H.248 EOCQ reports from a common media endpoint address.

Σ Packets Lost

A Total Packets is determined in the signaling by reporting the Packets missing in the stream in the incoming direction (rtp/pl) from the endpoint perspective. This information is available in the H.248 Reply to the Subtract message sent from AG to the CCA in the Call Flow. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

Total Packets Lost Percentage (Packets Lost %)

Description:

Computes the Ratio of the number of Total Packets Lost during the measured period. This uses the packets lost statistic from the Media H.248 EOCQ reports from a common media endpoint address.

Σ Total Packets Lost / Σ Total Packets * 100

Total Packets Lost Percentage is determined by counting the number of RTP Packets missing in the stream in the incoming direction divided by the number of packets Sent from the other end of the stream to the AG. Media stream messages are illustrated in the call flow in Call Flow –Stream Illustration in Figure 5.

Total Octets (Sent)

Description:

Counts the number of RTP Octets that were sent during the measured period. This uses the octets sent statistic from the Media H.248 EOCQ reports from a common media endpoint address.

Σ Octets Sent

A Total Octets Sent is determined in the signaling by reporting the Octets Sent in the stream in the outgoing direction (nt/os) from the endpoint perspective. This information is available in H.248 Reply to the Subtract message sent from AG to the CCA in the Call Flow. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

Average RTP Delay

Description:

The Average RTP Delay KPI measures RTP Packet Latency during the Media Sessions during the measured period. This uses the packet delay statistic from the Media H.248 EOCQ reports from a common media endpoint address.

Σ RTP Delay / Σ Session Attempts

Average RTP Delay is determined in the signaling by reporting the Packet Delay (rtp/delay) from the endpoint perspective. This information is available in H.248 Reply to the Subtract message sent from AG to the CCA in the Call Flow. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

Average RTP Jitter

Description:

The Average RTP Jitter KPI measures the variation in arrival rates between individual packets during the measured period. This uses the packets jitter statistic from the Media H.248 EOCQ reports from a common media endpoint address.

Σ RTP Jitter / Σ Session Attempts

Average RTP Jitter is determined in the signaling by reporting the Packet Jitter sent in the stream in the incoming direction (rtp/jit) from the endpoint perspective. This information is available in H.248 Reply to the Subtract message sent from AG to the CCA in the Call Flow. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

Average Kilobits Sent

Description:

Counts the number of RTP Kilobits that were sent during the measured period. This uses the octets sent statistic from the Media H.248 EOCQ reports from a common media endpoint address.

(Σ Octets Sent / 1024) / Σ Session Attempts

Average Kilobits is determined in the signaling by reporting the Octets sent in the stream in the outgoing direction (nt/os) and dividing by 1024 from the endpoint perspective. This information is available in H.248 Reply to the Subtract message sent from AG to the CCA in the Call Flow. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

Average MOS

Description:

The Average MOS (Mean Opinion Score) KPI measures the quality of the stream during the measured period using the MOS-CQ E-Model algorithm. This uses the packet delay, jitter and loss statistics from the Media H.248 EOCQ reports from a common media endpoint address.

Σ MOS / Σ Session Attempts

The Average MOS KPI is determined in the signaling by applying the MOS-CQ E-Model algorithm to the measured Delay, Jitter and Packet Loss KPIs measured from the endpoint perspective. This information is available in H.248 Reply to the Subtract H.248 Reply to the Subtract message is sent from AG to the CCA in the Call Flow. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

Minimum MOS

Description:

The Minimum MOS (Mean Opinion Score) KPI measures the Minimum Average MOS of the stream during the measured period using the MOS-CQ E-Model algorithm. This uses the packet delay, jitter and loss statistics from the Media H.248 EOCQ reports from a common media endpoint address.

MINVALUE (MOS)

The Minimum MOS for a given measured period records the Minimum Average MOS for the worst 5-minute period Originated at the AG during the measured period.

For example, if during a 24-hour period there was a 5-minute interval where the Average MOS was 1, and this was the worst score for the 24-hour period, then the Minimum MOS for that 24-hour period was 1.

Average R-Factor

Description:

The Average R-Factor KPI measures the quality of the stream during the measured period using the MOS-CQ/R-Factor E-Model algorithm. This uses the packet delay, jitter and loss statistics from the Media H.248 EOCQ reports from a common media endpoint address.

Σ R-Factor / Σ Session Attempts

The Average R-Factor KPI is determined in the signaling by applying the R-Factor E-Model algorithm to the measured Delay, Jitter and Packet Loss KPIs measured from the endpoint perspective. This information is available in H.248 Reply to the Subtract message is sent from AG to the CCA in the Call Flow. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

Minimum R-Factor

Description:

The Minimum R-Factor KPI measures the Minimum Average R-Factor of the stream during the measured period using the MOS-CQ/R-Factor E-Model algorithm. This uses the packet delay, jitter and loss statistics from the Media H.248 EOCQ reports from a common media endpoint address.

MINVALUE (R-Factor)

The Minimum R-Factor for a given measured period records the Minimum Average R-Factor for the worst 5-minute period Originated at the AG during the measured period.

For example, if during a 24-hour period there was a 5-minte interval where the Average R-Factor was 25, and this was the worst score for the 24-hour period, then the Minimum R-Factor for that 24-hour period was 25.

Bin Counts

Description:

The Bin Count KPIs count the number of Media Sessions where the MOS was between a certain range during the measured period. This uses the packet delay, jitter and loss statistics from the Media H.248 EOCQ reports from a common media endpoint address.

Σ Session Attempts where MOS < X and MOS > Y

The Bin Counts KPI for a given measurement period records the number of Media Sessions where the MOS was between a certain range. The available Bin KPIs are the following:

- Excellent (Total and percentage calculated as ExcellentBin/SessionAttempts)
- Good (Total and percentage calculated as GoodBin/SessionAttempts)
- Fair (Total and percentage calculated as FairBin/SessionAttempts)
- Poor(Total and percentage calculated as PoorBin/SessionAttempts)
- Unacceptable (Total and percentage calculated as UnacceptableBin/SessionAttempts)

Extended Media XNQ KPIs

Session Attempts

Description:

Counts the number of Media Streams reported in H.248 XNQ during the measured period. This counts the number of Media H.248 XNQ reports from a common media endpoint address.

Σ Session Attempts

A Session Attempt is determined in the signaling by the H.248 Reply to the Subtract, which has XNQ RTP statistics. This Subtract message is sent from AG to the CCA in the Call Flow. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

Session Completions

Description:

Counts the number of Media Streams that completed normally during the measured period. This counts the number of Media H.248 XNQ reports from a common media endpoint address.

Σ Media Session Completed

A Media Session Completion is determined in the signaling by looking at the reported GeoProbe Timeouts, and the H.248 Response Code on the H.248 Leg. The Application can be configured so that all H.248 Response Codes and Timeouts can be mapped to Success, Failure, or Timeout. If the Response Code or Timeout on the H.248 Leg is 200 Normal Completion, then the Call is determined to be a Completion.

In the successful call case, in Figure 4 since there was no Response Code reported in the H.248 session, the 200 Normal Completion response code is reported.

NOTE: If the average MOS for the stream falls within the range of “Unacceptable” or “Poor” then the Session will be mapped to 8000 or 8001 respectively and counted as a Failure.

Session Failures

Description:

Counts the number of Media Streams that failed normally during the measured period. This uses the number of Media H.248 XNQ reports from a common media endpoint address.

Σ Media Session Failed

A Media Session Failure is determined in the signaling by looking at the reported GeoProbe Timeouts, and the H.248 Response Code on the H.248 Leg. The Application can be configured so that all H.248 Response Codes and Timeouts can be mapped to Success, Failure, or Timeout. If the Response Code or Timeout on the H.248 Leg is mapped to Failure, then the Call is determined to be a Failure.

In the successful call case, in Figure 4 since there was no Response Code reported in the H.248 session, the 200 Normal Completion response code is reported.

NOTE: If the average MOS for the stream falls within the range of “Unacceptable” or “Poor” then the Session will be mapped to 8000 or 8001 respectively and counted as a Failure.

Session Failure Percentage (Session Failure %)

Description:

Computes the percentage of Media Streams that failed normally during the measured period. This uses the number of Media H.248 XNQ reports from a common media endpoint address that completed normally.

(Σ Session Failures) / (Σ Session Attempts) * 100

Session Timeouts

Description:

Counts the number of Media Streams that failed normally during the measured period. This uses the number of Media H.248 XNQ reports from a common media endpoint address.

Σ Media Session Timeout

A Media Session Failure is determined in the signaling by looking at the reported GeoProbe Timeouts, and the H.248 Response Code on the H.248 Leg. The Application can be configured so that all H.248 Response Codes and Timeouts can be mapped to Success, Failure, or Timeout. If the Response Code or Timeout on the H.248 Leg is mapped to timeout, then the Call is determined to be a timeout.

Session Timeout Percentage (Session Timeout %)

Description:

Computes the percentage of Media Streams that timed out normally during the measured period. This uses the number of Media H.248 XNQ reports from a common media endpoint address that completed normally.

(Σ Session Timeouts) / (Σ Session Attempts) * 100

Session Duration

Description:

Sums the RTP endpoint durations for all the H.248 calls as reported by the media endpoint reports. This uses the endpoint duration statistic from the Media H.248 EOCQ reports from a common media endpoint address.

Σ Session Duration

The RTP endpoint duration is determined in the signaling by reporting the RTP endpoint duration statistic in the stream (rtp/dur) for the endpoint H.248 Reply to the Subtract, this Reply message is sent from AG to the CCA in the Call Flow. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

Average Session Duration

Description:

Computes the Average Session Duration. This uses the endpoint duration statistic from the Media H.248 EOCQ reports from a common media endpoint address.

(Σ Session Duration) / (Σ Session Attempts)

Near End Extended Media XNQ KPIs

NE Reporting Sessions

Description:

Counts the number of NE Media Streams reported in H.248 XNQ during the measured period. This counts the number of NE Media H.248 XNQ reports from a common media endpoint address

Σ NE Session Attempts

NE Reporting Sessions Duration

Description:

Sums the RTP NE endpoint durations for all the H.248 calls as reported by the media endpoint reports. This uses the endpoint duration statistic from the NE Media H.248 EOCQ reports from a common media endpoint address.

Σ NE Session Duration

NE Maximum IPDV

Description:

Takes the maximum of all the Near End (NE) IPDV for all the H.248 calls as reported by the media endpoint reports. This uses the NE IPDV Max Diff statistic from the Media H.248 XNQ reports from a common media endpoint address.

MAXVALUE (Near End (NE) IPDV)

The Maximum Near End (NE) IPDV is determined in the signaling by reporting the NE IPDV statistic in the stream (xnq/nvmaxdiff) for the endpoint H.248 Reply to the Subtract, this Reply message is sent from AG to the CCA in the Call Flow. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

NE Average Maximum IPDV

Description:

Computes the Average Maximum Near End (NE) IPDV. This uses the endpoint duration statistic from the Media H.248 XNQ reports from a common media endpoint address.

(Σ Maximum Near End (NE) IPDV) / (Σ NE Session Attempts)

NE Global Maximum IPDV

Description:

Takes the maximum of all the Near End (NE) Global Maximum IPDV for all the H.248 calls as reported by the media endpoint reports. This uses the NE Global Maximum IPDV statistic from the Media H.248 XNQ reports from a common media endpoint address.

MAXVALUE (Near End (NE) Global Maximum IPDV)

The Maximum Near End (NE) Global Maximum IPDV is determined in the signaling by reporting the NE IPDV statistic in the stream (xnq/nvrange) for the endpoint H.248 Reply to the Subtract, this Reply message is sent from AG to the CCA in the Call Flow. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

NE Average Global Maximum IPDV

Description:

Computes the Average Global Maximum Near End (NE) IPDV. This uses the endpoint duration statistic from the Media H.248 XNQ reports from a common media endpoint address.

(Σ Maximum Near End (NE) IPDV) / (Σ NE Session Attempts)

NE Maximum IPDV Cycles

Description:

Takes the maximum of all the Near End (NE) IPDV Cycles for all the H.248 calls as reported by the media endpoint reports. This uses the NE IPDV Cycles statistic from the Media H.248 XNQ reports from a common media endpoint address.

MAXVALUE (Near End (NE) IPDV Cycles)

The Maximum Near End (NE) IPDV Cycles is determined in the signaling by reporting the NE IPDV Cycles statistic in the stream (xnq/nvcyc) for the endpoint H.248 Reply to the Subtract, this Reply message is sent from AG to the CCA in the Call Flow. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

NE Normalized IPDV Sum

Description:

Calculates the average of the Near End (NE) IPDV Sum for all the H.248 calls as reported by the media endpoint reports. This uses the NE IPDV Sum statistic from the Media H.248 XNQ reports from a common media endpoint address.

Σ Near End (NE) IPDV Sum / Σ NE IPDV Cycles

The Near End (NE) IPDV Sum is determined in the signaling by reporting the NE Jitter Buffer Adaptation Events statistic in the stream (xnq/njbevents) for the endpoint H.248 Reply to the Subtract, this Reply message is sent from AG to the CCA in the Call Flow divided by the call duration of the media session. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

NE Average Jitter Buffer Adaptation Events

Description:

Calculates the average of the Near End (NE) Jitter Buffer Adaptation Events for all the H.248 calls as reported by the media endpoint reports. This uses the NE Jitter Buffer Adaptation Events statistic from the Media H.248 XNQ reports from a common media endpoint address.

Σ Near End (NE) Jitter Buffer Adaptation Events / Σ NE Session Attempts

The Near End (NE) Average Jitter Buffer Adaptation Events is determined in the signaling by reporting the NE Jitter Buffer Adaptation Events statistic in the stream (xnq/njbevents) for the endpoint H.248 Reply to the Subtract, this Reply message is sent from AG to the CCA in the Call Flow. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

NE Normalized Jitter Buffer Adaptation Events

Description:

Calculates the average of the Near End (NE) Jitter Buffer Adaptation Events for all the H.248 calls as reported by the media endpoint reports. This uses the NE Jitter Buffer Adaptation Events statistic from the Media H.248 XNQ reports from a common media endpoint address.

Σ Near End (NE) Jitter Buffer Adaptation Events / Σ NE Session Duration

The Near End (NE) Normalized Jitter Buffer Adaptation Events is determined in the signaling by reporting the NE Jitter Buffer Adaptation Events statistic in the stream (xnq/njbevents) for the endpoint H.248 Reply to the Subtract, this Reply message is sent from AG to the CCA in the Call Flow divided by the call duration of the media session. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

NE Sum Jitter Time Degraded

Description:

Sums the Near End (NE) Sum Jitter Time Degraded for all the H.248 calls as reported by the media endpoint reports. This uses the NE Jitter Time Degraded statistic from the Media H.248 XNQ reports from a common media endpoint address.

Σ Near End (NE) Sum Jitter Time Degraded

The Near End (NE) Sum Jitter Time Degraded is determined in the signaling by reporting the NE Sum IPDV Jitter Time Degraded statistic in the stream (xnq/ntdegjit) for the endpoint H.248 Reply to the Subtract, this Reply message is sent from AG to the CCA in the Call Flow. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

NE Average Jitter Time Degraded

Description:

Calculates the average of the Near End (NE) Jitter Time Degraded for all the H.248 calls as reported by the media endpoint reports. This uses the NE Jitter Time Degraded statistic from the Media H.248 XNQ reports from a common media endpoint address.

Σ Near End (NE) Jitter Time Degraded / Σ NE Session Attempts

The Near End (NE) Average Jitter Time Degraded is determined in the signaling by reporting the NE Jitter Time Degraded statistic in the stream (xnq/ntdegjit) for the endpoint H.248 Reply to the Subtract, this Reply message is sent from AG to the CCA in the Call Flow. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

NE Sum Network Time Degraded

Description:

Sums the Near End (NE) Sum Network Time Degraded for all the H.248 calls as reported by the media endpoint reports. This uses the NE Network Time Degraded statistic from the Media H.248 XNQ reports from a common media endpoint address.

Σ Near End (NE) Sum Network Time Degraded

The Near End (NE) Sum Network Time Degraded is determined in the signaling by reporting the NE Sum IPDV Network Time Degraded statistic in the stream (xnq/ntdegnet) for the endpoint H.248 Reply to the Subtract, this Reply message is sent from AG to the CCA in the Call Flow. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

NE Average Network Time Degraded

Description:

Calculates the average of the Near End (NE) Network Time Degraded for all the H.248 calls as reported by the media endpoint reports. This uses the NE Network Time Degraded statistic from the Media H.248 XNQ reports from a common media endpoint address.

Σ Near End (NE) Network Time Degraded / Σ NE Session Attempts

The Near End (NE) Average Network Time Degraded is determined in the signaling by reporting the NE Network Time Degraded statistic in the stream (xnq/ntdegnet) for the endpoint H.248 Reply to the Subtract, this Reply message is sent from AG to the CCA in the Call Flow. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

NE Sum Network Degraded Seconds

Description:

Sums the Near End (NE) Sum Network Degraded Seconds for all the H.248 calls as reported by the media endpoint reports. This uses the NE Network Degraded Seconds statistic from the Media H.248 XNQ reports from a common media endpoint address.

Σ Near End (NE) Sum Network Degraded Seconds

The Near End (NE) Sum Network Degraded Seconds is determined in the signaling by reporting the NE Sum IPDV Network Degraded Seconds statistic in the stream (xnq/nes) for the endpoint H.248 Reply to the Subtract, this Reply message is sent from AG to the CCA in the Call Flow. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

NE Average Network Degraded Seconds

Description:

Calculates the average of the Near End (NE) Network Degraded Seconds for all the H.248 calls as reported by the media endpoint reports. This uses the NE Network Degraded Seconds statistic from the Media H.248 XNQ reports from a common media endpoint address.

Σ Near End (NE) Network Degraded Seconds / Σ NE Session Attempts

The Near End (NE) Average Network Degraded Seconds is determined in the signaling by reporting the NE Network Degraded Seconds statistic in the stream (xnq/nes) for the endpoint H.248 Reply to the Subtract, this Reply message is sent from AG to the CCA in the Call Flow. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

NE Sum Network Severely Degraded Count

Description:

Sums the Near End (NE) Sum Network Severely Degraded Count for all the H.248 calls as reported by the media endpoint reports. This uses the NE Network Severely Degraded Count statistic from the Media H.248 XNQ reports from a common media endpoint address.

Σ Near End (NE) Sum Network Severely Degraded Count

The Near End (NE) Sum Network Severely Degraded Count is determined in the signaling by reporting the NE Sum IPDV Network Severely Degraded Count statistic in the stream (xnq/nses) for the endpoint H.248 Reply to the Subtract, this Reply message is sent from AG to the CCA in the Call Flow. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

NE Average Network Severely Degraded Count

Description:

Calculates the average of the Near End (NE) Network Severely Degraded Count for all the H.248 calls as reported by the media endpoint reports. This uses the NE Network Severely Degraded Count statistic from the Media H.248 XNQ reports from a common media endpoint address.

Σ Near End (NE) Network Severely Degraded Count / Σ NE Session Attempts

The Near End (NE) Average Network Severely Degraded Count is determined in the signaling by reporting the NE Network Degraded Seconds statistic in the stream (xnq/nses) for the endpoint H.248 Reply to the Subtract, this Reply message is sent from AG to the CCA in the Call Flow. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

NE Sum RTP Cumulative Packet Loss

Description:

Sums the Near End (NE) Sum RTP Cumulative Packet Loss for all the H.248 calls as reported by the media endpoint reports. This uses the NE RTP Cumulative Packet Loss statistic from the Media H.248 XNQ reports from a common media endpoint address.

Σ Near End (NE) Sum RTP Cumulative Packet Loss

The Near End (NE) Sum RTP Cumulative Packet Loss is determined in the signaling by reporting the NE Sum IPDV RTP Cumulative Packet Loss statistic in the stream (xnq/ncumpl) for the endpoint H.248 Reply to the Subtract, this Reply message is sent from AG to the CCA in the Call Flow. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

NE Average RTP Cumulative Packet Loss

Description:

Calculates the average of the Near End (NE) RTP Cumulative Packet Loss for all the H.248 calls as reported by the media endpoint reports. This uses the NE RTP Cumulative Packet Loss statistic from the Media H.248 XNQ reports from a common media endpoint address.

Σ Near End (NE) RTP Cumulative Packet Loss / Σ NE Session Attempts

The Near End (NE) Average RTP Cumulative Packet Loss is determined in the signaling by reporting the NE Network Degraded Seconds statistic in the stream (xnq/ncumpl) for the endpoint H.248 Reply to the Subtract, this Reply message is sent from AG to the CCA in the Call Flow. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

Far End Extended Media XNQ KPIs

FE Reporting Sessions

Description:

Counts the number of FE Media Streams reported in H.248 XNQ during the measured period. This counts the number of FE Media H.248 XNQ reports from a common media endpoint address

Σ FE Session Attempts

FE Reporting Sessions Duration

Description:

Sums the RTP FE endpoint durations for all the H.248 calls as reported by the media endpoint reports. This uses the endpoint duration statistic from the FE Media H.248 EOCQ reports from a common media endpoint address.

Σ FE Session Duration

FE Maximum IPDV

Description:

Takes the maximum of all the Far End (FE) IPDV for all the H.248 calls as reported by the media endpoint reports. This uses the FE IPDV Maximum Diff statistic from the Media H.248 XNQ reports from a common media endpoint address.

MAXVALUE (Far End (FE) IPDV)

The Maximum Far End (FE) IPDV is determined in the signaling by reporting the FE IPDV statistic in the stream (xnq/fvmaxdiff) for the endpoint H.248 Reply to the Subtract, this Reply message is sent from AG to the CCA in the Call Flow. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

FE Average Maximum IPDV

Description:

Computes the Average Maximum Far End (FE) IPDV. This uses the endpoint duration statistic from the Media H.248 XNQ reports from a common media endpoint address.

(Σ Average Maximum Far End (FE) IPDV) / (ΣFE Session Attempts)

FE Global Maximum IPDV

Description:

Takes the maximum of all the Far End (FE) Global Maximum IPDV for all the H.248 calls as reported by the media endpoint reports. This uses the FE Global Maximum IPDV statistic from the Media H.248 XNQ reports from a common media endpoint address.

MAXVALUE (Far End (FE) Global Maximum IPDV)

The Maximum Far End (FE) Global Maximum IPDV is determined in the signaling by reporting the FE IPDV statistic in the stream (xnq/fvrange) for the endpoint H.248 Reply to the Subtract, this Reply message is sent from AG to the CCA in the Call Flow. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

FE Average Global Maximum IPDV

Description:

Computes the Average Global Maximum Far End (NE) IPDV. This uses the endpoint duration statistic from the Media H.248 XNQ reports from a common media endpoint address.

(Σ Maximum Far End (NE) IPDV) / (Σ FE Session Attempts)

FE Maximum IPDV Cycles

Description:

Takes the maximum of all the Far End (FE) IPDV Cycles for all the H.248 calls as reported by the media endpoint reports. This uses the FE IPDV Cycles statistic from the Media H.248 XNQ reports from a common media endpoint address.

MAXVALUE (Far End (FE) IPDV Cycles)

The Maximum Far End (FE) IPDV Cycles is determined in the signaling by reporting the FE IPDV Cycles statistic in the stream (xnq/fvcyc) for the endpoint H.248 Reply to the Subtract, this Reply message is sent from AG to the CCA in the Call Flow. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

FE Normalized IPDV Sum

Description:

Calculates the average of the Far End (FE) IPDV Sum for all the H.248 calls as reported by the media endpoint reports. This uses the NE IPDV Sum statistic from the Media H.248 XNQ reports from a common media endpoint address.

Σ Far End (FE) IPDV Sum / Σ FE IPDV Cycles

The Far End (FE) IPDV Sum is determined in the signaling by reporting the FE Jitter Buffer Adaptation Events statistic in the stream (xnq/njbevents) for the endpoint H.248 Reply to the Subtract, this Reply message is sent from AG to the CCA in the Call Flow divided by the call duration of the media session. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

FE Average Jitter Buffer Adaptation Events

Description:

Calculates the average of the Far End (FE) Jitter Buffer Adaptation Events for all the H.248 calls as reported by the media endpoint reports. This uses the FE Jitter Buffer Adaptation Events statistic from the Media H.248 XNQ reports from a common media endpoint address.

Σ Far End (FE) Jitter Buffer Adaptation Events / Σ Session Attempts

The Far End (FE) Average Jitter Buffer Adaptation Events is determined in the signaling by reporting the FE Jitter Buffer Adaptation Events statistic in the stream (xnq/fjbevents) for the endpoint H.248 Reply to the Subtract, this Reply message is sent from AG to the CCA in the Call Flow. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

FE Normalized Jitter Buffer Adaptation Events

Description:

Calculates the average of the Near End (FE) Jitter Buffer Adaptation Events for all the H.248 calls as reported by the media endpoint reports. This uses the FE Jitter Buffer Adaptation Events statistic from the Media H.248 XNQ reports from a common media endpoint address.

Σ Far End (FE) Jitter Buffer Adaptation Events / Σ FE Session Duration

The Far End (NE) Normalized Jitter Buffer Adaptation Events is determined in the signaling by reporting the FE Jitter Buffer Adaptation Events statistic in the stream (xnq/njbevents) for the endpoint H.248 Reply to the Subtract, this Reply message is sent from AG to the CCA in the Call Flow divided by the call duration of the media session. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

FE Sum Jitter Time Degraded

Description:

Sums the Far End (FE) Sum Jitter Time Degraded for all the H.248 calls as reported by the media endpoint reports. This uses the FE Jitter Time Degraded statistic from the Media H.248 XNQ reports from a common media endpoint address.

Σ Far End (FE) Sum Jitter Time Degraded

The Far End (FE) Sum Jitter Time Degraded is determined in the signaling by reporting the FE Sum IPDV Jitter Time Degraded statistic in the stream (xnq/ftdegjit) for the endpoint H.248 Reply to the Subtract, this Reply message is sent from AG to the CCA in the Call Flow. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

FE Average Jitter Time Degraded

Description:

Calculates the average of the Far End (FE) Jitter Time Degraded for all the H.248 calls as reported by the media endpoint reports. This uses the FE Jitter Time Degraded statistic from the Media H.248 XNQ reports from a common media endpoint address.

Σ Far End (FE) Jitter Time Degraded / Σ FE Session Attempts

The Far End (FE) Average Jitter Time Degraded is determined in the signaling by reporting the FE Jitter Time Degraded statistic in the stream (xnq/ftdegjit) for the endpoint H.248 Reply to the Subtract, this Reply message is sent from AG to the CCA in the Call Flow. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

FE Sum Network Time Degraded

Description:

Sums the Far End (FE) Sum Network Time Degraded for all the H.248 calls as reported by the media endpoint reports. This uses the FE Network Time Degraded statistic from the Media H.248 XNQ reports from a common media endpoint address.

Σ Far End (FE) Sum Network Time Degraded

The Far End (FE) Sum Network Time Degraded is determined in the signaling by reporting the FE Sum IPDV Network Time Degraded statistic in the stream (xnq/ftdegnet) for the endpoint H.248 Reply to the Subtract, this Reply message is sent from AG to the CCA in the Call Flow. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

FE Average Network Time Degraded

Description:

Calculates the average of the Far End (FE) Network Time Degraded for all the H.248 calls as reported by the media endpoint reports. This uses the FE Network Time Degraded statistic from the Media H.248 XNQ reports from a common media endpoint address.

Σ Far End (FE) Network Time Degraded / Σ FE Session Attempts

The Far End (FE) Average Network Time Degraded is determined in the signaling by reporting the FE Network Time Degraded statistic in the stream (xnq/ftdegnet) for the endpoint H.248 Reply to the Subtract, this Reply message is sent from AG to the CCA in the Call Flow. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

FE Sum Network Degraded Seconds

Description:

Sums the Far End (FE) Sum Network Degraded Seconds for all the H.248 calls as reported by the media endpoint reports. This uses the FE Network Degraded Seconds statistic from the Media H.248 XNQ reports from a common media endpoint address.

Σ Far End (FE) Sum Network Degraded Seconds

The Far End (FE) Sum Network Degraded Seconds is determined in the signaling by reporting the FE Sum IPDV Network Degraded Seconds statistic in the stream (xnq/fes) for the endpoint H.248 Reply to the Subtract, this Reply message is sent from AG to the CCA in the Call Flow. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

FE Average Network Degraded Seconds

Description:

Calculates the average of the Far End (FE) Network Degraded Seconds for all the H.248 calls as reported by the media endpoint reports. This uses the FE Network Degraded Seconds statistic from the Media H.248 XNQ reports from a common media endpoint address.

Σ Far End (FE) Network Degraded Seconds / Σ FE Session Attempts

The Far End (FE) Average Network Degraded Seconds is determined in the signaling by reporting the FE Network Degraded Seconds statistic in the stream (xnq/fes) for the endpoint H.248 Reply to the Subtract, this Reply message is sent from AG to the CCA in the Call Flow. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

FE Sum Network Severely Degraded Count

Description:

Sums the Far End (FE) Sum Network Severely Degraded Count for all the H.248 calls as reported by the media endpoint reports. This uses the FE Network Severely Degraded Count statistic from the Media H.248 XNQ reports from a common media endpoint address.

Σ Far End (FE) Sum Network Severely Degraded Count

The Far End (FE) Sum Network Severely Degraded Count is determined in the signaling by reporting the FE Sum IPDV Network Severely Degraded Count statistic in the stream (xnq/fses) for the endpoint H.248 Reply to the Subtract, this Reply message is sent from AG to the CCA in the Call Flow. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

FE Average Network Severely Degraded Count

Description:

Calculates the average of the Far End (FE) Network Severely Degraded Count for all the H.248 calls as reported by the media endpoint reports. This uses the FE Network Severely Degraded Count statistic from the Media H.248 XNQ reports from a common media endpoint address.

Σ Far End (FE) Network Severely Degraded Count / Σ FE Session Attempts

The Far End (FE) Average Network Severely Degraded Count is determined in the signaling by reporting the FE Network Degraded Seconds statistic in the stream (xnq/fses) for the endpoint H.248 Reply to the Subtract, this Reply message is sent from AG to the CCA in the Call Flow. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

FE Sum RTP Cumulative Packet Loss

Description:

Sums the Far End (FE) Sum RTP Cumulative Packet Loss for all the H.248 calls as reported by the media endpoint reports. This uses the FE RTP Cumulative Packet Loss statistic from the Media H.248 XNQ reports from a common media endpoint address.

Σ Far End (FE) Sum RTP Cumulative Packet Loss

The Far End (FE) Sum RTP Cumulative Packet Loss is determined in the signaling by reporting the FE Sum IPDV RTP Cumulative Packet Loss statistic in the stream (xnq/fcumpl) for the endpoint H.248 Reply to the

Subtract, this Reply message is sent from AG to the CCA in the Call Flow. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

FE Average RTP Cumulative Packet Loss

Description:

Calculates the average of the Far End (FE) RTP Cumulative Packet Loss for all the H.248 calls as reported by the media endpoint reports. This uses the FE RTP Cumulative Packet Loss statistic from the Media H.248 XNQ reports from a common media endpoint address.

Σ Far End (FE) RTP Cumulative Packet Loss / Σ FE Session Attempts

The Far End (FE) Average RTP Cumulative Packet Loss is determined in the signaling by reporting the FE Network Degraded Seconds statistic in the stream (xnq/fcumpl) for the endpoint H.248 Reply to the Subtract, this Reply message is sent from AG to the CCA in the Call Flow. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

Round Trip Delay Extended Media XNQ KPIs

Sum Round Trip Delay

Description:

Sums Round Trip Delay for all the H.248 calls as reported by the media endpoint reports. This uses the Minimum Round Trip Delay statistic from the Media H.248 XNQ reports from a common media endpoint address.

Σ Sum Round Trip Delay

The Sum Round Trip Delay is determined in the signaling by reporting the Minimum Round Trip Delay statistic in the stream (xnq/rtdmin) for the endpoint H.248 Reply to the Subtract, this Reply message is sent from AG to the CCA in the Call Flow. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

Last Round Trip Delay

Description:

Sums the Last Round Trip Delay for all the H.248 calls as reported by the media endpoint reports. This uses the Last Round Trip Delay statistic from the Media H.248 XNQ reports from a common media endpoint address.

Σ Last Round Trip Delay

The Last Round Trip Delay is determined in the signaling by reporting the Last Round Trip Delay statistic in the stream (xnq/rtdnow) for the endpoint H.248 Reply to the Subtract, this Reply message is sent from AG to the CCA in the Call Flow. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

Minimum Round Trip Delay

Description:

Takes the Minimum Trip Delay for all the H.248 calls as reported by the media endpoint reports. This uses the Minimum Round Trip Delay statistic from the Media H.248 XNQ reports from a common media endpoint address.

MINVALUE (Minimum Round Trip Delay)

The Minimum Round Trip Delay is determined in the signaling by reporting the Minimum Round Trip Delay statistic in the stream (xnq/rtdmin) for the endpoint H.248 Reply to the Subtract, this Reply message is sent from AG to the CCA in the Call Flow. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

Maximum Round Trip Delay

Description:

Takes the Maximum Trip Delay for all the H.248 calls as reported by the media endpoint reports. This uses the Maximum Round Trip Delay statistic from the Media H.248 XNQ reports from a common media endpoint address.

MAXVALUE (Maximum Round Trip Delay)

The Maximum Round Trip Delay is determined in the signaling by reporting the Maximum Round Trip Delay statistic in the stream (xnq/rtdmax) for the endpoint H.248 Reply to the Subtract, this Reply message is sent from AG to the CCA in the Call Flow. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

Average Last Round Trip Delay

Description:

Takes the Average Trip Delay for all the H.248 calls as reported by the media endpoint reports. This uses the Average Last Round Trip Delay statistic from the Media H.248 XNQ reports from a common media endpoint address.

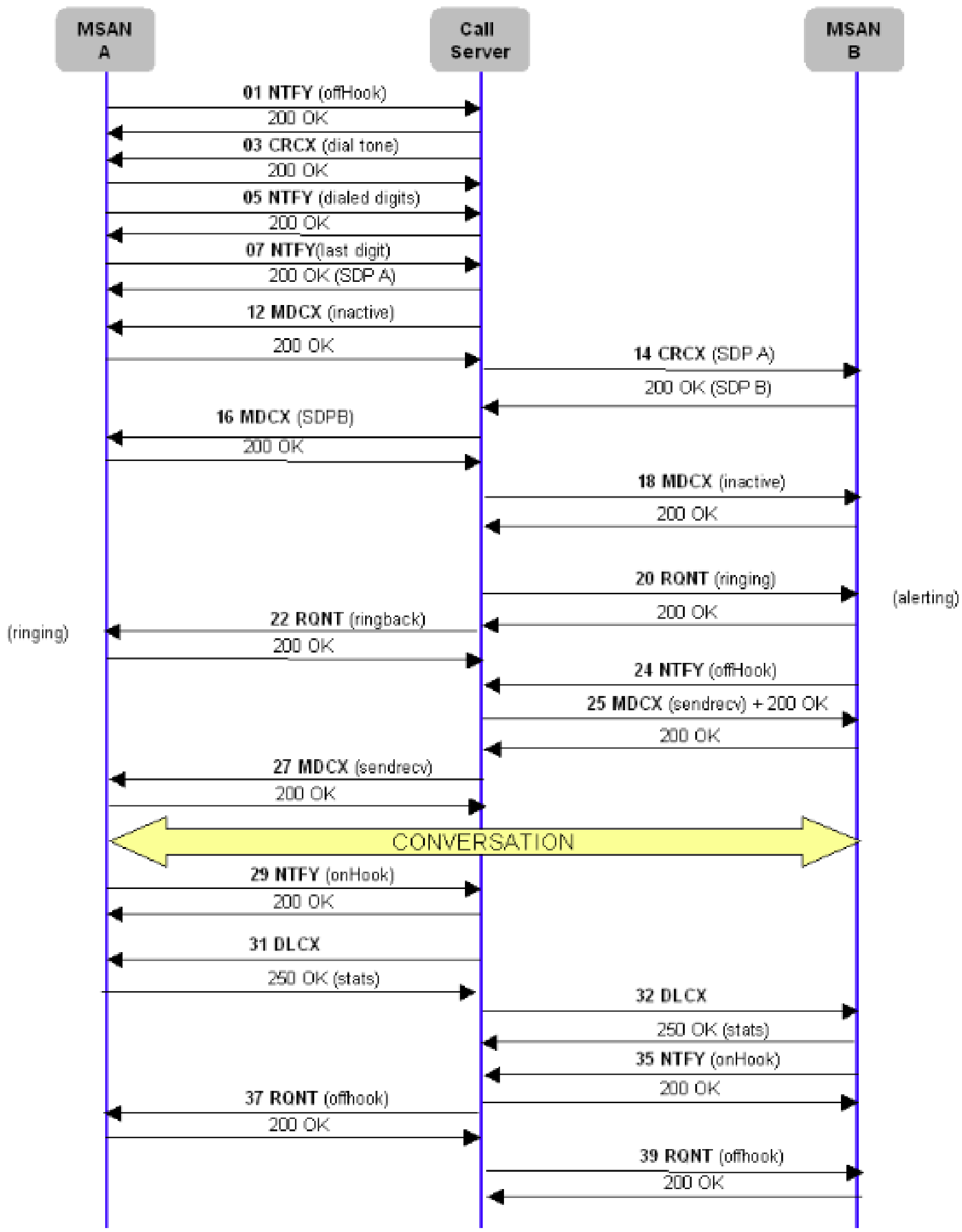
Σ Average Last Round Trip Delay / Σ FE Session Attempts

The Average Last Round Trip Delay is determined in the signaling by reporting the Average Last Round Trip Delay statistic in the stream (xnq/rtdnow) for the endpoint H.248 Reply to the Subtract, this Reply message is sent from AG to the CCA in the Call Flow. This message is illustrated in message #40 and #34 in the call flow in Figure 4.

IPI MGCP KEY PERFORMANCE INDICATORS

We will refer to the following call flow diagrams through out the KPI discussions. Please note that these call flows have been simplified for the KPI discussion. For example, often the Reply message is omitted if its inclusion is not necessary in the relevant discussion. KPIs are calculated on Source and Destination nodes considering the direction of the call or of the stream.

Figure 6. Call Flow - Successful Call



Call QoS KPIs

Session Attempts

Description:

Counts the number of Media Streams reported in MGCP EOCQ during the measured period. This counts the number of Media MGCP EOCQ reports from a common media endpoint address.

Σ Session Attempts

A Session Attempt is determined in the signaling by the MGCP DLCX which has RTP statistics. This message is illustrated in message #32 and #34 in the call flow in Figure 6.

Session Completions

Description:

Counts the number of Media Streams that completed normally during the measured period. This counts the number of Media MGCQ EOCQ reports from a common media endpoint address.

Σ Media Session Completed

A Media Session Completion is determined in the signaling by looking at the reported GeoProbe Timeouts, and the MGCP Response Code on the MGCP Leg. The Application can be configured so that all MGCP Response Codes and Timeouts can be mapped to Success, Failure, or Timeout. If the Response Code or Timeout on the MGCP Leg is 200 Normal Completion, then the Call is determined to be a Completion.

In the successful call case, in Figure 6 since there was no Response Code reported in the MGCP session, the 200 Normal Completion response code is reported.

NOTE: If the average MOS for the stream falls within the range of “Unacceptable” or “Poor” then the Session will be mapped to 8000 or 8001 respectively and counted as a Failure.

Session Failures

Description:

Counts the number of Media Streams that failed normally during the measured period. This uses the number of Media MGCP EOCQ reports from a common media endpoint address.

Σ Media Session Failed

A Media Session Failure is determined in the signaling by looking at the reported GeoProbe Timeouts, and the MGCP Response Code on the MGCP Leg. The Application can be configured so that all MGCP Response Codes and Timeouts can be mapped to Success, Failure, or Timeout. If the Response Code or Timeout on the MGCP Leg is 200 Normal Completion, then the Call is determined to be a Completion.

In the successful call case, in Figure 6 since there was no Response Code reported in the MGCP session, the 200 Normal Completion response code is reported.

NOTE: If the average MOS for the stream falls within the range of “Unacceptable” or “Poor” then the Session will be mapped to 8000 or 8001 respectively and counted as a Failure.

Session Failure Percentage (Session Failure %)

Description:

Computes the percentage of Media Streams that failed normally during the measured period. This uses the number of Media MGCP EOCQ reports from a common media endpoint address that completed normally.

(Σ Failures) / (Σ Session Attempts) * 100

Session Timeouts

Description:

Counts the number of Media Streams that timed out normally during the measured period. This uses the number of Media MGCP EOCQ reports from a common media endpoint address.

Σ Media Session Timeouts

A Media Session Timeout is determined in the signaling by looking at the reported GeoProbe Timeouts, and the MGCP Response Code on the MGCP Leg. The Application can be configured so that all MGCP Response Codes and Timeouts can be mapped to Success, Failure, or Timeout. If the Response Code or Timeout on the MGCP Leg is mapped to a timeout, then the Call is determined to be a Timeout.

Session Timeout Percentage (Session Timeout %)

Description:

Computes the percentage of Media Streams that timed out normally during the measured period. This uses the number of Media MGCP EOCQ reports from a common media endpoint address that completed normally.

(Σ Timeouts) / (Σ Session Attempts) * 100

Total Packets (Sent)

Description:

Counts the number of RTP Packets during the measured period. This uses the packets sent statistic from the Media MGCP EOCQ reports from a common media endpoint address.

Σ Packets Sent

A Packets Sent is determined in the signaling by reporting the Packets Sent in the MGCP DLCX message which has RTP statistics in the stream of outgoing direction. This message is illustrated in message #32 and #34 in the call flow in Figure 6.

Total Packets Lost

Description:

Counts the number of RTP Packets that were lost during the measured period. This uses the packets lost statistic from the Media MGCP EOCQ reports from a common media endpoint address.

Σ Packets Lost

A Total Packets is determined in the signaling by reporting the Total Packets in the MGCP DLCX message which has RTP statistics in the stream of incoming direction. This message is illustrated in message #32 and #34 in the call flow in Figure 6.

Total Packets Lost Percentage (Packets Lost %)

Description:

Computes the Ratio of the number of Total Packets Lost during the measured period. This uses the packets lost statistic from the Media MGCP EOCQ reports from a common media endpoint address.

Σ Total Packets Lost / Σ Total Packets * 100

Total Packets Lost Percentage is determined by counting the number of RTP Packets missing in the stream in the incoming direction divided by the number of packets Sent from the other end of the stream to the AG.

Total Octets (Sent)

Description:

Counts the number of RTP Octets that were sent during the measured period. This uses the octets sent statistic from the Media MGCP EOCQ reports from a common media endpoint address.

Σ Octets Sent

A Total Octets Sent is determined in the signaling by reporting the Octets Sent in the stream in the outgoing direction (nt/os) from the endpoint perspective . This information is available in the MGCP DLCX message which has RTP statistics. This message is illustrated in message #32 and #34 in the call flow in Figure 6.

Average RTP Delay

Description:

The Average RTP Delay KPI measures RTP Packet Latency during the Media Sessions during the measured period. This uses the packet delay statistic from the Media MGCP EOCQ reports from a common media endpoint address.

Σ RTP Delay / Σ Session Attempts

Average RTP Delay is determined in the signaling by reporting the Packet Delay (rtp/delay) from the endpoint perspective. This information is available in the MGCP DLCX message which has RTP statistics. This message is illustrated in message #32 and #34 in the call flow in Figure 6.

Average RTP Jitter

Description:

The Average RTP Jitter KPI measures the variation in arrival rates between individual packets during the measured period. This uses the packets jitter statistic from the Media MGCP EOCQ reports from a common media endpoint address.

Σ RTP Jitter / Σ Session Attempts

Average RTP Jitter is determined in the signaling by reporting the Packet Jitter sent in the stream in the incoming direction (rtp/jit) from the endpoint perspective. This information is available in the MGCP DLCX message which has RTP statistics. This message is illustrated in message #32 and #34 in the call flow in Figure 6.

Average Kilobits Sent

Description:

Counts the number of RTP Kilobits that were sent during the measured period. This uses the octets sent statistic from the Media MGCP EOCQ reports from a common media endpoint address.

(Σ Octets Sent / 1024) / Σ Session Attempts

Average Kilobits is determined in the signaling by reporting the Octets sent in the stream in the outgoing direction (nt/os) and dividing by 1024 from the endpoint perspective. This information is available in the MGCP DLCX message which has RTP statistics. This message is illustrated in message #32 and #34 in the call flow in Figure 6.

Average MOS

Description:

The Average MOS (Mean Opinion Score) KPI measures the quality of the stream during the measured period using the MOS-CQ E-Model algorithm. This uses the packet delay, jitter and loss statistics from the Media MGCP EOCQ reports from a common media endpoint address.

Σ MOS / Σ Session Attempts

The Average MOS KPI is determined in the signaling by applying the MOS-CQ E-Model algorithm to the measured Delay, Jitter and Packet Loss KPIs measured from the endpoint perspective. This information is available in the MGCP DLCX message which has RTP statistics. This message is illustrated in message #32 and #34 in the call flow in Figure 6.

Minimum MOS

Description:

The Minimum MOS (Mean Opinion Score) KPI measures the Minimum Average MOS of the stream during the measured period using the MOS-CQ E-Model algorithm. This uses the packet delay, jitter and loss statistics from the Media MGCP EOCQ reports from a common media endpoint address.

MINVALUE (MOS)

The Minimum MOS for a given measured period records the Minimum Average MOS for the worst 5-minute period Originated at the AG during the measured period.

For example, if during a 24-hour period there was a 5-minute interval where the Average MOS was 1, and this was the worst score for the 24-hour period, then the Minimum MOS for that 24-hour period was 1.

Average R-Factor

Description:

The Average R-Factor KPI measures the quality of the stream during the measured period using the MOS-CQ/R-Factor E-Model algorithm. This uses the packet delay, jitter and loss statistics from the Media MGCP EOCQ reports from a common media endpoint address.

Σ R-Factor / Σ Session Attempts

The Average R-Factor KPI is determined in the signaling by applying the R-Factor E-Model algorithm to the measured Delay, Jitter and Packet Loss KPIs measured from the endpoint perspective. This information is available in the MGCP DLCX message which has RTP statistics. This message is illustrated in message #32 and #34 in the call flow in Figure 6.

Minimum R-Factor

Description:

The Minimum R-Factor KPI measures the Minimum Average R-Factor of the stream during the measured period using the MOS-CQ/R-Factor E-Model algorithm. This uses the packet delay, jitter and loss statistics from the Media MGCP EOCQ reports from a common media endpoint address.

MINVALUE (R-Factor)

The Minimum R-Factor for a given measured period records the Minimum Average R-Factor for the worst 5-minute period Originated at the AG during the measured period.

For example, if during a 24-hour period there was a 5-minte interval where the Average R-Factor was 25, and this was the worst score for the 24-hour period, then the Minimum R-Factor for that 24-hour period was 25.

Bin Counts

Description:

The Bin Count KPIs count the number of Media Sessions where the MOS was between a certain range during the measured period. This uses the packet delay, jitter and loss statistics from the Media MGCP EOCQ reports from a common media endpoint address.

Σ Session Attempts where MOS < X and MOS > Y

The Bin Counts KPI for a given measurement period records the number of Media Sessions where the MOS was between a certain range. The available Bin KPIs are the following:

- Excellent (Total and percentage calculated as ExcellentBin/SessionAttempts)
- Good (Total and percentage calculated as GoodBin/SessionAttempts)
- Fair (Total and percentage calculated as FairBin/SessionAttempts)
- Poor(Total and percentage calculated as PoorBin/SessionAttempts)
- Unacceptable (Total and percentage calculated as UnacceptableBin/SessionAttempts)

IPI RTP KEY PERFORMANCE INDICATORS

We will refer to the following call flow diagrams through out the KPI discussions. Please note that these call flows have been simplified for the KPI discussion. For example, often the Reply message is omitted if its inclusion is not necessary in the relevant discussion. KPIs are calculated on Source and Destination nodes considering the direction of the stream.

Figure 7. Call Flow - Successful Call

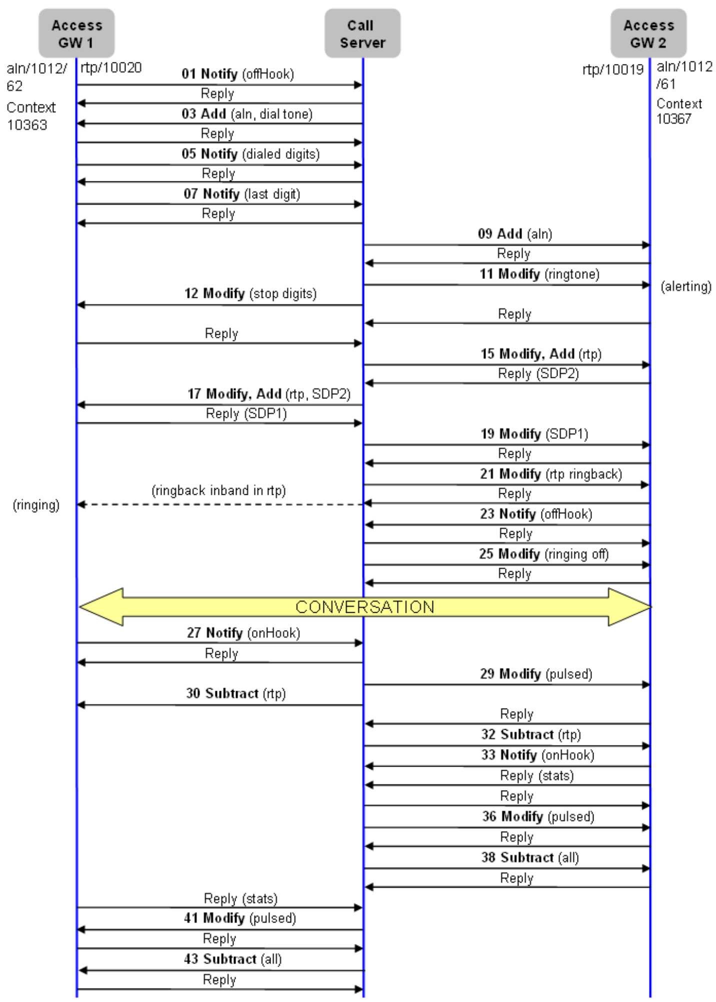
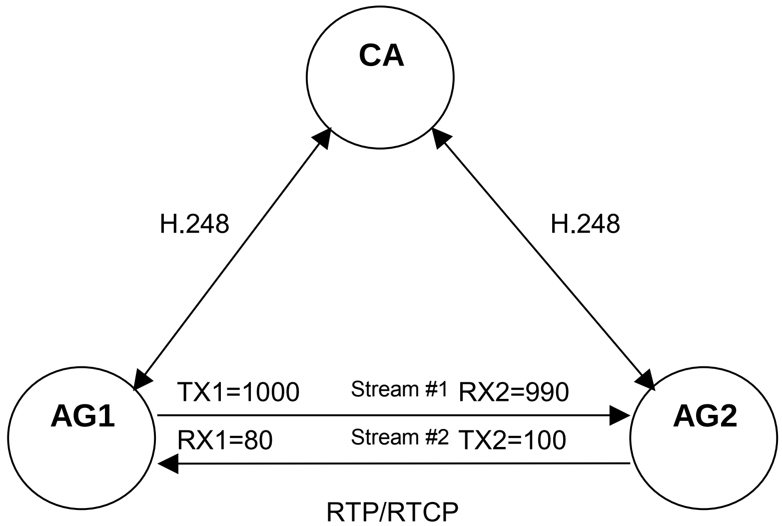
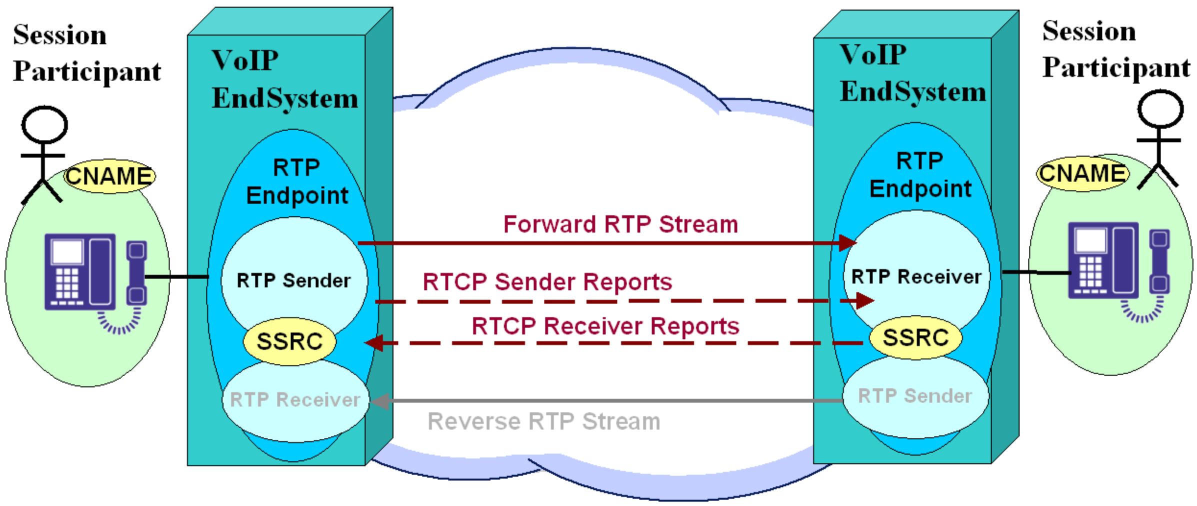


Figure 8. Call Flow – Stream Illustration



Originatin g Node	Terminatin g Node	Protocol	Sent	Receive d	Lost	From Call Leg
AG1 (Reporting)	AG2	H.248 EOCQ	1000	80	20	AG1 H.248 Call Leg
AG2 (Reporting)	AG1	H.248 EOCQ	100	990	10	AG2 H.248 Call Leg
AG1	AG2	Direct RTP	1000	990	10	RTP Stream #1
AG2	AG1	Direct RTP	100	80	20	RTP Stream #2

Figure 9. RTCP Sender and Receiver Reports used in RTP Delay



Call QoS KPIs

Session Attempts

Description:

Counts the number of Media Streams during the measured period.

Σ Session Attempts

A Session Attempt is determined in the signaling by the monitoring of a RTP stream in the Call Flow. The media stream messages are illustrated in the call flow in Call Flow –Stream Illustration in Figure 8. If there were a stream in the direction of AG1 to AG2 then this would count as a Session Attempt.

AG1 will be Source of a session attempt while AG2 will be Destination of a session attempt from monitoring Stream #1 in the call flow in Call Flow –Stream Illustration in Figure 8.

AG2 will be Source of a session attempt while AG1 will be Destination of a session attempt from monitoring Stream #2 in the call flow in Call Flow –Stream Illustration in Figure 8.

Session Completions

Description:

Counts the number of Media Streams that completed normally during the measured period.

Σ Media Session Completed

A Media Session Completion is determined in the signaling by looking at the reported GeoProbe Timeouts, and the Media Response Code on the Media Leg. The Application can be configured so that all RTP Response Codes and Timeouts can be mapped to Success, Failure, or Timeout. If the Response Code or Timeout on the Media Leg is 200 Normal Completion, then the Call is determined to be a Completion.

In the successful call case, media stream messages are illustrated in the call flow in Call Flow –Stream Illustration in Figure 8, there is no Response Code reported in the RTP session. In this case the 200 Normal Completion response code is reported.

NOTE: If the average MOS for the stream falls within the range of “Unacceptable” or “Poor” then the Session will be mapped to 8000 or 8001 respectively and counted as a Failure.

Session Failures

Description:

Counts the number of Media Streams that failed during the measured period.

Σ Media Session Failed

A Media Session Failure is determined in the signaling by looking at the reported GeoProbe Timeouts, and the Media Response Code on the Media stream going “into” the AG. The Application can be configured so that all RTP Response Codes and Timeouts can be mapped to Success, Failure, or Timeout. If the Response Code or Timeout on the Media Leg is mapped to Failure, then the Call is determined to be a Failure.

In the successful call case, media stream messages are illustrated in the call flow in Call Flow –Stream Illustration in Figure 8, there is no Response Code reported in the RTP session. In this case the 200 Normal Completion response code is reported.

NOTE: If the average MOS for the stream falls within the range of “Unacceptable” or “Poor” then the Session will be mapped to 8000 or 8001 respectively and counted as a Failure.

Session Failures Percentage (Session Failure %)

Description:

Computes the percentage of Media Streams that failed during the measured period.

(Σ Failures) / (Σ Session Attempts) * 100

Session Timeouts

Description:

Counts the number of Media Streams that failed during the measured period.

Σ Media Session Timeouts

A Media Session Timeout is determined in the signaling by looking at the reported GeoProbe Timeouts, and the Media Response Code on the Media stream going “into” the AG. The Application can be configured so that all RTP Response Codes and Timeouts can be mapped to Success, Failure, or Timeout. If the Response Code or Timeout on the Media Leg is mapped to timeout, then the Call is determined to be a timeout.

Session Timeouts Percentage (Session Timeout %)

Description:

Computes the percentage of Media Streams that timed out during the measured period.

(Σ Timeouts) / (Σ Session Attempts) * 100

One Way Sessions

Description:

Counts the number of Media Streams that have been established and that do not have a peer stream on the backwards direction.

Σ One Way Sessions

A One Way Session Failure is determined in the signaling by checking the established RTP stream. GeoProbe marks One-Way sessions in the Status Bits.

One Way Session Failures Percentage (One Way Session %)

Description:

Computes the percentage of Media Streams that have been set up in one direction only.

(Σ Failures) / (Σ Session Attempts) * 100

Total Packets (Sent)

Description:

Counts the number of RTP Packets that were Sent during the measured period.

Σ Packets Sent

Total Packets Sent is determined in the signaling by counting the number of RTP Packets sent. The media stream messages are illustrated in the call flow in Call Flow –Stream Illustration in Figure 8.

Note: In direct RTP monitoring, the number of packets counted depends on where the probes are monitoring the streams.

Total Packets Lost

Description:

Counts the number of RTP Packets that were lost during the measured period.

Σ Packets Lost

Total Packets Lost is determined in the signaling by counting the number of RTP Packets missing in the stream. The system determines the number of missing packets by looking at the sequence numbers in the RTP headers. The media stream messages are illustrated in the call flow in Call Flow –Stream Illustration in Figure 8.

Note: In direct RTP monitoring, the number of lost packets depends on where the probes are monitoring the streams.

Total Packets Lost Percentage (Packets Lost %)

Description:

Computes the Ratio of the number of Total Packets Lost during the measured period.

Σ Total Packets Lost / Σ Total Packets Sent * 100

Total Packets Lost Percentage is determined by dividing Total Packets Lost by the number of Total Packets Sent. Media stream messages are illustrated in the call flow in Call Flow –Stream Illustration in Figure 8.

Total Packets Out of Sequence

Description:

Counts the number of RTP Packets that were received out of sequence during the measured period.

Σ Total Packets Out of Sequence

Total Packets Lost is determined in the signaling by counting the number of RTP Packets where the sequence number is lower than the latest received sequence number. The media stream messages are illustrated in the call flow in Call Flow –Stream Illustration in Figure 8.

Note: In direct RTP monitoring, the number of out of sequence packets depends on where the probes are monitoring the streams.

Total Packets Out of Sequence Percentage (Packets Out of Sequence %)

Description:

Computes the Ratio of the number of Total Packets Out of Sequence during the measured period.

Σ Total Packets Out of Sequence / Σ Total Packets Sent * 100

Total Packets Out of Sequence Percentage is determined by dividing Total Packets Out of Sequence by the number of Total Packets Sent. Media stream messages are illustrated in the call flow in Call Flow –Stream Illustration in Figure 8.

Average Packet Iat

Description:

It is the Average Packet Inter-arrival Time in ms.

Σ (Average Packet Inter-arrival by Session) / (Number of Sessions with Average Packet Inter-arrival available)

Total Octets (Sent)

Description:

Counts the number of RTP Octets that were Sent during the measured period.

Σ Octets Sent

Total Octets Sent is determined in the signaling by counting the number of RTP Octets. Media stream messages are illustrated in the call flow in Call Flow –Stream Illustration in Figure 5.

Note: In direct RTP monitoring, the number of octets counted depends on where the probes are monitoring the streams.

Average RTP Delay

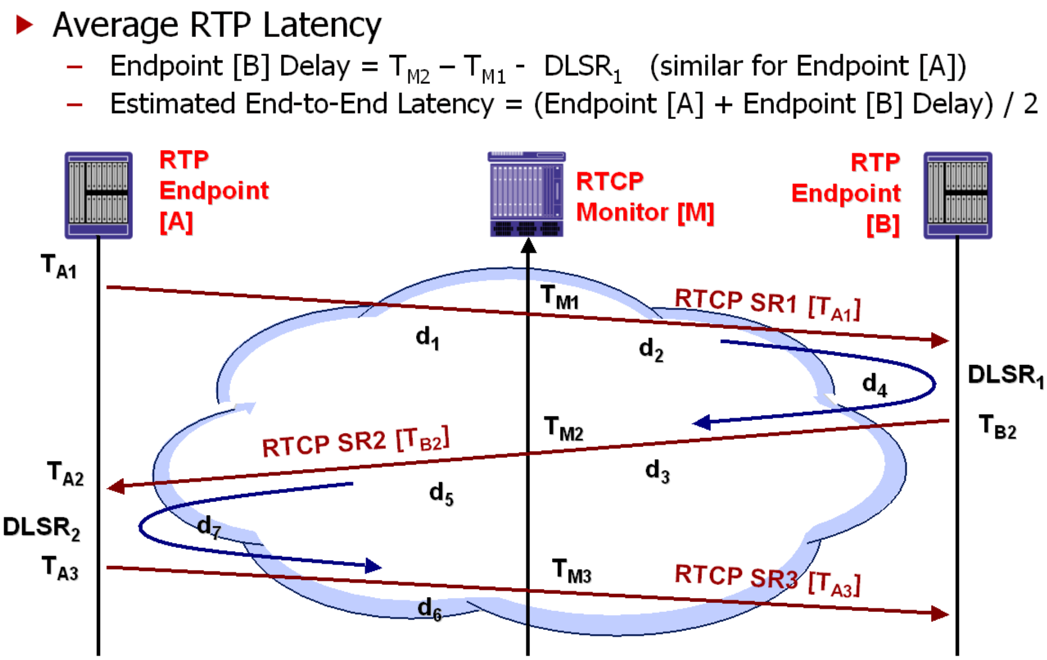
Description:

The Average RTP Delay KPI measures RTP Packet Latency during the Media Sessions during the measured period.

Σ RTP Delay / Σ Session Attempts

The Average RTP Delay KPI is determined in the signaling by calculating the RTCP "End to End Latency" value, and dividing it by 2 (please note that this KPI is available only if RTCP information is present). Media stream messages are illustrated in the call flow in Call Flow –Stream Illustration below.

Figure 10. Call Flow –Stream Illustration



Note: This KPI is available only if RTCP information is present.

Note: In direct RTP monitoring, the RTP Delay depends on where the probes are monitoring the streams.

Average RTP Jitter

Description:

The Ave RTP Jitter KPI measures the variation in arrival rates between individual packets on the stream during the measured period.

Σ RTP Jitter / Σ Session Attempts

Average RTP Jitter is determined in the signaling by measuring the variation in arrival rates between individual packets monitored in the stream. Media stream messages are illustrated in the call flow in Call Flow –Stream Illustration in Figure 5.

Note: In direct RTP monitoring, the RTP jitter depends on where the probes are monitoring the streams.

Average Kilobits (Sent)

Description:

Counts the number of RTP Kilobits that were sent during the measured period.

(Σ Octets Sent / 1024) / Σ Session Attempts

A Total Kilobits is determined in the signaling by counting the number of RTP Kilobits in the stream. The media stream messages are illustrated in the call flow in Call Flow –Stream Illustration in Figure 5.

Note: In direct RTP monitoring, the Kilobits Sent depends on where the probes are monitoring the streams.

Average MOS

Description:

The Average MOS (Mean Opinion Score) KPI measures the quality of the stream during the measured period using the MOS-CQ E-Model algorithm.

Σ MOS / Σ Session Attempts

The Average MOS KPI is determined in the signaling by applying the MOS-CQ E-Model algorithm to the measured Delay, Jitter and Packet Loss KPIs which is defined in Appendix. Media stream messages are illustrated in the call flow in Call Flow –Stream Illustration in Figure 5.

Note: In direct RTP monitoring, the MOS depends on where the probes are monitoring the streams.

Minimum MOS

Description:

The Minimum MOS (Mean Opinion Score) KPI measures the Minimum Ave MOS of the stream during the measured period using the MOS-CQ E-Model algorithm.

MINVALUE(MOS)

The Minimum MOS for a given measure period records the Minimum Ave MOS for the worst 5-minute period during the measured period.

For example, if during a 24-hour period there was a 5-minte interval where the Ave MOS was 1, and this was the worst score for the 24-hour period, then the Minimum MOS for that 24-hour period was 1.

Average R-Factor

Description:

The Average R-Factor KPI measures the quality of the stream during the measured period using the MOS-CQ/R-Factor E-Model algorithm.

Σ R-Factor / Σ Session Attempts

The Average R-Factor KPI is determined in the signaling by applying the R-Factor E-Model algorithm to the measured Delay, Jitter and Packet Loss KPIs that is defined in Appendix. Media stream messages are illustrated in the call flow in Call Flow –Stream Illustration Figure 5.

Note: In direct RTP monitoring, the R-Factor depends on where the probes are monitoring the streams.

Minimum R-Factor

Description:

The Minimum R-Factor KPI measures the Minimum Ave R-Factor of the stream during the measured period using the MOS-CQ/R-Factor E-Model algorithm.

MINVALUE(R-Factor)

The Minimum R-Factor for a given measure period records the Minimum Ave R-Factor for the worst 5-minute period during the measured period.

For example, if during a 24-hour period there was a 5-minte interval where the Ave R-Factor was 25, and this was the worst score for the 24-hour period, then the Minimum R-Factor for that 24-hour period was 25.

Bin Counts

Description:

The Bin Count KPIs count the number of Media Sessions where the MOS was between a certain range during the measured period.

Σ Session Attempts where MOS < X and MOS > Y

The Bin Counts KPI for a given measurement period records the number of Media Sessions where the MOS was between a certain ranges; the range is defined in Appendix . The available Bin KPIs are the following:

- Excellent (Total and percentage calculated as ExcellentBin/SessionAttempts)
- Good (Total and percentage calculated as GoodBin/SessionAttempts)
- Fair (Total and percentage calculated as FairBin/SessionAttempts)
- Poor(Total and percentage calculated as PoorBin/SessionAttempts)
- Unacceptable (Total and percentage calculated as UnacceptableBin/SessionAttempts)

Note: In direct RTP monitoring, the MOS depends on where the probes are monitoring the streams.

IPI RTCP KEY PERFORMANCE INDICATORS

We will refer to the following call flow diagrams through out the KPI discussions. Please note that these call flows have been simplified for the KPI discussion. For example, often the Reply message is omitted if its inclusion is not necessary in the relevant discussion. KPIs are calculated on Source and Destination nodes considering the direction of the stream.

Figure 11. Call Flow - Successful Call

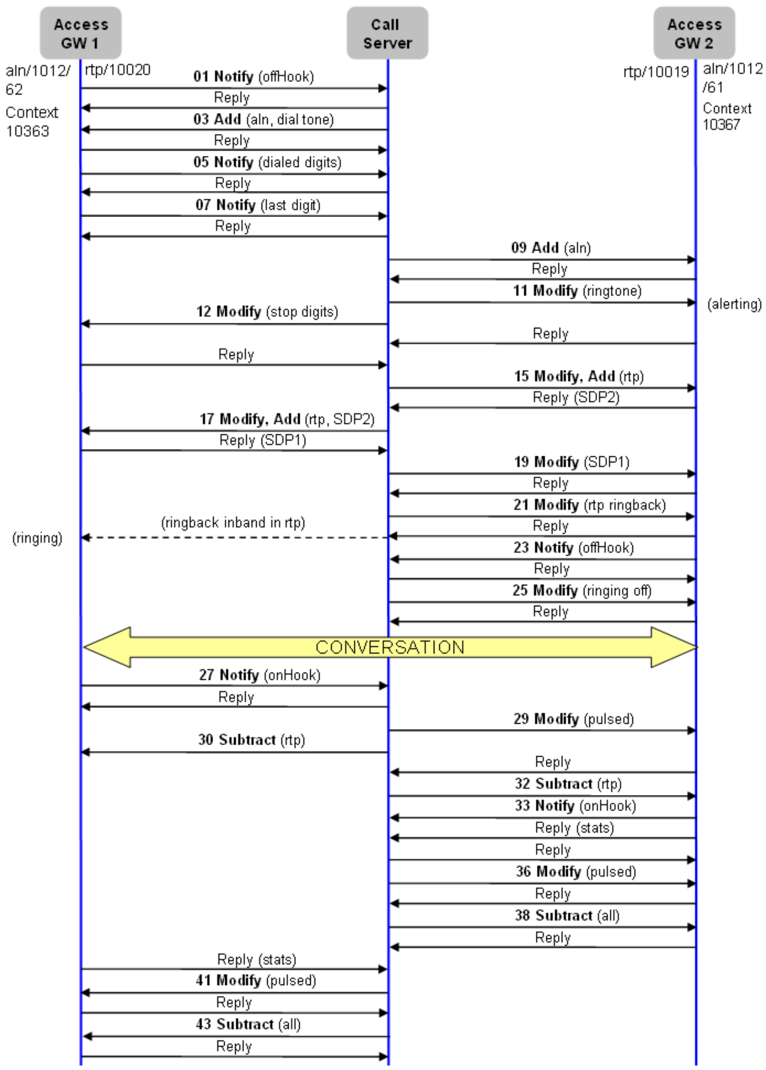
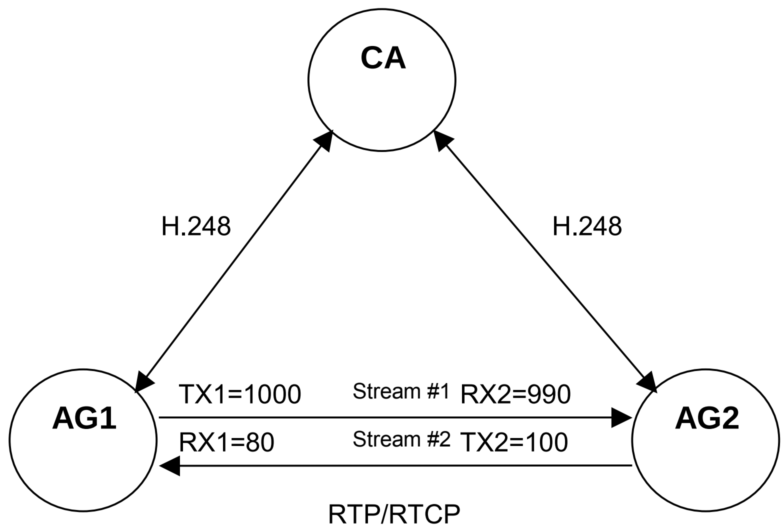
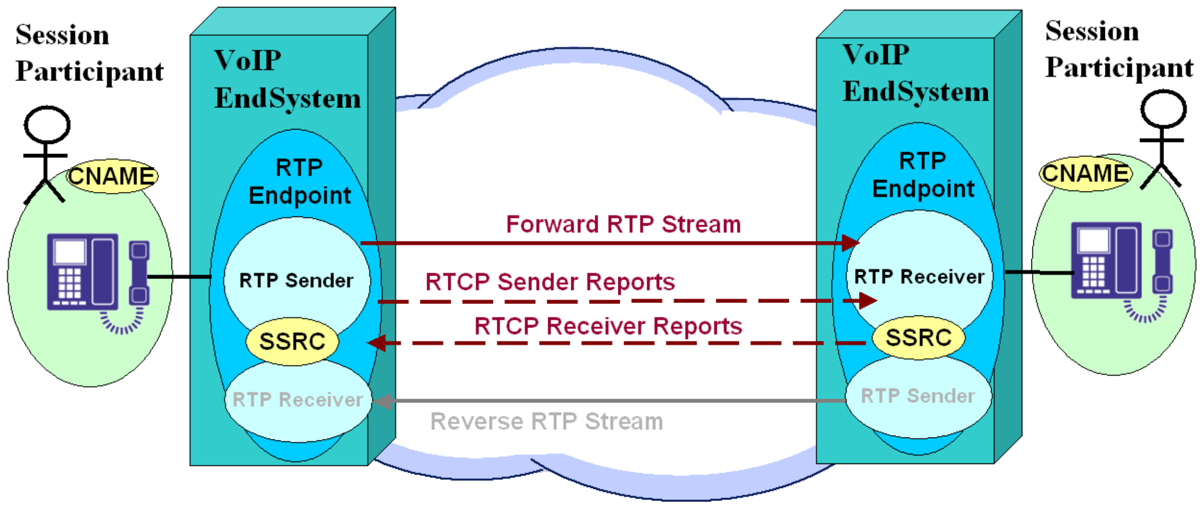


Figure 12. Call Flow – Stream Illustration



Originating Node	Terminating Node	Protocol	Sent	Received	Lost	From Call Leg
AG1 (Reporting)	AG2	H.248 EOCQ	1000	80	20	AG1 H.248 Call Leg
AG2 (Reporting)	AG1	H.248 EOCQ	100	990	10	AG2 H.248 Call Leg
AG1	AG2	RTCP	1000	990	10	RTP Stream #1
AG2	AG1	RTCP	100	80	20	RTP Stream #2

Figure 13. RTCP Sender and Receiver Reports



Call QoS KPIs

Session Attempts

Description:

Counts the number of Media Streams during the measured period.

Σ Session Attempts

A Session Attempt is determined in the signaling by the RTCP Receiver Reports (RRs) and Sender Reports (SRs) for a stream. The media stream messages are illustrated in the Call Flow –Stream Illustration. If there is a stream in the direction of AG1 to AG2 then this would count as a Session Attempt.

AG1 would be Source of one session attempt while AG2 would be Destination of one session attempt from monitoring the RTCP Receiver Reports (RRs) and Sender Reports (SRs) for Stream #1 in the call flow in the Call Flow –Stream Illustration.

AG2 would be Source of one session attempt while AG1 would be Destination of one session attempt from monitoring the RTCP Receiver Reports (RRs) and Sender Reports (SRs) for Stream #2 in the call flow in the Call Flow –Stream Illustration.

Session Completions

Description:

Counts the number of Media Streams that completed normally during the measured period.

Σ Media Session Completed

A Media Session Completion is determined in the signaling by looking at the reported GeoProbe Timeouts, and the Media Response Code on the Media Leg. The Application can be configured so that all RTCP Response Codes and Timeouts can be mapped to Success, Failure, or Timeout. If the Response Code or Timeout on the Media Leg is 200 Normal Completion, then the Call is determined to be a Completion.

In the successful call case, media stream messages are illustrated in the call flow in the Call Flow –Stream Illustration, there is no Response Code reported in the RTP session. In this case the 200 Normal Completion response code is reported.

NOTE: If the average MOS reported in the RTCP Receiver Reports (RRs) for the stream falls within the range of “Unacceptable” or “Poor” (see MOS bin definitions in Appendix) then the Session will be mapped to 8000 or 8001 respectively and counted as Failure.

Session Failures

Description:

Counts the number of Media Streams that failed during the measured period.

Σ Media Session Failed

A Media Session Failure is determined in the signaling by looking at the reported GeoProbe Timeouts, and the Media Response Code on the Media stream going “into” the AG. The Application can be configured so that all RTP Response Codes and Timeouts can be mapped to Success, Failure, or Timeout. If the Response Code or Timeout on the Media Leg is mapped to Failure, then the Call is determined to be a Failure.

In the successful call case, media stream messages are illustrated in the Call Flow –Stream Illustration, there is no Response Code reported in the RTP session. In this case the 200 Normal Completion response code is reported.

NOTE: If the average MOS reported in the RTCP Receiver Reports (RRs) for the stream falls within the range of “Unacceptable” or “Poor” (see MOS bin definitions in Appendix) then the Session will be mapped to 8000 or 8001 respectively and counted as a Failure.

Session Failures Percentage

Description:

Computes the percentage of Media Streams that failed during the measured period.

(Σ Failures) / (Σ Session Attempts) * 100

Session Timeouts

Description:

Counts the number of Media Streams that timed out during the measured period.

Σ Media Session Timeouts

A Media Session Timeout is determined in the signaling by looking at the reported GeoProbe Timeouts, and the Media Response Code on the Media stream going “into” the AG. The Application can be configured so that all RTP Response Codes and Timeouts can be mapped to Success, Failure, or Timeout. If the Response Code or Timeout on the Media Leg is mapped to timeout, then the Call is determined to be a timeout.

Session Timeouts Percentage

Description:

Computes the percentage of Media Streams that timed out during the measured period.

(Σ Timeouts) / (Σ Session Attempts) * 100

Total Packets (Sent)

Description:

Counts the number of RTP Packets that were Sent, as reported by RTCP, during the measured period.

Σ Packets Sent

Total Packets Sent is determined in the signaling by reporting the number of RTP Packets sent, as reported by RTCP in the Sender Reports (SRs). The media stream messages are illustrated in the call flow in the Call Flow –Stream Illustration.

Total Packets Lost

Description:

Total Packets Lost reports the number of RTP Packets that were Lost, as reported by RTCP, during the measured period.

Σ Packets Lost

Total Packets Lost is determined in the signaling by counting the number of RTP Packets missing in the stream, as reported by the RTCP in the Receiver Reports (RRs). RTCP determines the number of missing packets by looking at the sequence numbers in the RTP headers. The media stream messages are illustrated in the call flow in the Call Flow –Stream Illustration.

Note: In direct RTP monitoring, the number of lost depends on where the probes are monitoring the streams.

Total Packets Lost Percentage

Description:

Computes the Ratio of the number of Total Packets Lost during the measured period.

Σ Total Packets Lost / Σ Total Packets Sent * 100

Total Packets Lost Percentage is determined by dividing Total Packets Lost by the number of Total Packets Sent. Media stream messages are illustrated in the Call Flow –Stream Illustration.

Total Octets (Sent)

Description:

Counts the number of RTP Octets that were Sent, as reported by RTCP, during the measured period.

Σ Octets Sent

Total Octets Sent is determined in the signaling by reporting the number of RTP Octets sent, as reported by RTCP in the Sender Reports (SRs). The media stream messages are illustrated in the call flow in the Call Flow –Stream Illustration.

Average RTP Delay

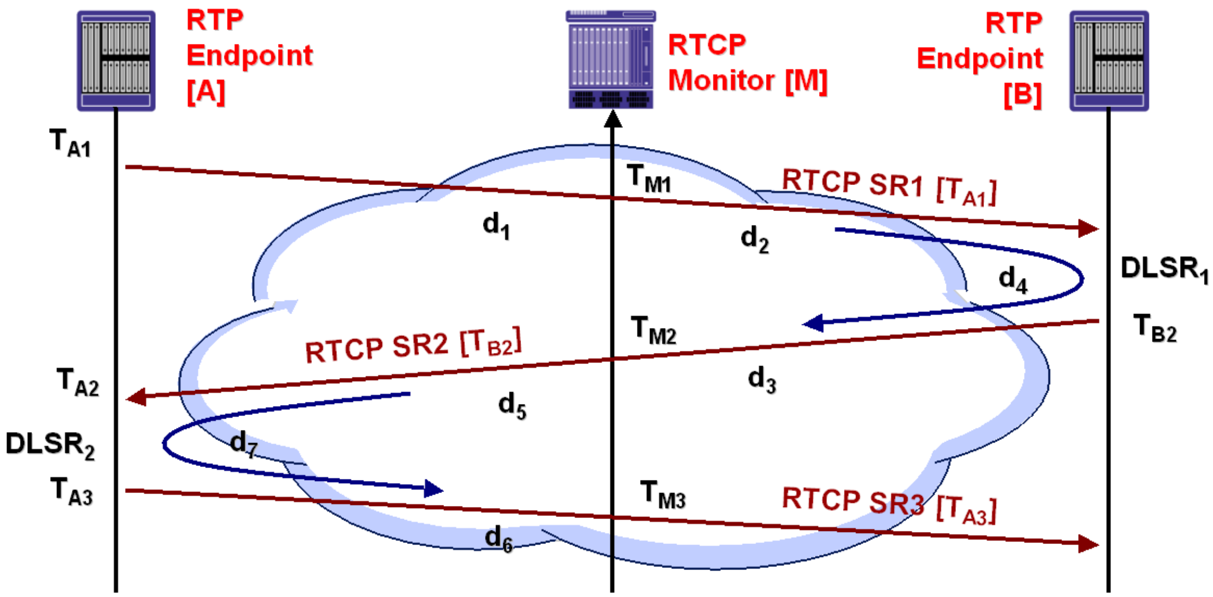
Description:

The Average RTP Delay KPI measures RTP Packet Latency, as reported by RTCP, during the measured period.

Σ RTP Delay / Σ Session Attempts

The Average RTP Delay KPI is determined in the signaling by calculating the RTCP “End to End Latency” value, and dividing it by 2 (please note that this KPI is available only if RTCP information is present). Media stream messages are illustrated in the Call Flow –Stream Illustration.

- Average RTP Latency
  - Endpoint [B] Delay = $T_{M2} - T_{M1} -$ $DLSR_1$ (similar for Endpoint [A])
  - Estimated End-to-End Latency = (Endpoint [A] + Endpoint [B] Delay) / 2



Note: This KPI is available only if RTCP information is present.

Note: In direct RTP monitoring, the RTP Delay depends on where the probes are monitoring the streams.

Average RTP Jitter

Description:

The Ave RTP Jitter KPI reports the variation in arrival rates between individual packets on the stream, as reported by RTCP, during the measured period.

Σ RTP Jitter / Σ Session Attempts

Average RTP Jitter is determined in the signaling by reporting the RTP Jitter, as reported by RTCP in the Receiver Reports (RRs). Media stream messages are illustrated in the call flow in the Call Flow –Stream Illustration.

Average Kilobits (Sent)

Description:

Counts the number of RTP Kilobits that were Sent, as reported by RTCP, during the measured period.

(Σ Octets Sent / 1024) / Σ Session Attempts

Total Kilobits Sent is determined in the signaling by reporting the number of RTP Kilobits sent, as reported by RTCP in the Sender Reports (SRs). The media stream messages are illustrated in the Call Flow –Stream Illustration.

Average MOS

Description:

The Average MOS (Mean Opinion Score) KPI measures the quality of the stream, as reported by RTCP, during the measured period using the MOS-CQ E-Model algorithm.

Σ MOS / Σ Session Attempts

The Average MOS KPI is determined in the signaling by applying the MOS-CQ E-Model algorithm to the measured Delay, Jitter and Packet Loss KPIs from the RTCP Receiver Reports (RRs). Media stream messages are illustrated in the Call Flow –Stream Illustration.

Minimum MOS

Description:

The Minimum MOS (Mean Opinion Score) KPI measures the Minimum Ave MOS of the stream, as reported by RTCP, during the measured period using the MOS-CQ E-Model algorithm.

MINVALUE(MOS)

The Minimum MOS for a given measure period records the Minimum Ave MOS for the worst 5-minute period during the measured period.

For example, if during a 24-hour period there was a 5-minte interval where the Ave MOS was 1, and this was the worst score for the 24-hour period, then the Minimum MOS for that 24-hour period was 1.

Average R-Factor

Description:

The Average R-Factor KPI measures the quality of the stream, as reported by RTCP, during the measured period using the MOS-CQ/ R-Factor E-Model algorithm.

Σ R-Factor / Σ Session Attempts

The Average R-Factor KPI is determined in the signaling by applying the R-Factor E-Model algorithm to the measured Delay, Jitter and Packet Loss KPIs from the RTCP Receiver Reports (RRs). Media stream messages are illustrated in the Call Flow –Stream Illustration.

Note: In direct RTP monitoring, the R-Factor depends on where the probes are monitoring the streams.

Minimum R-Factor

Description:

The Minimum R-Factor KPI reports the Minimum Ave R-Factor of the stream, as reported by RTCP, during the measured period using the MOS-CQ/R-Factor E-Model algorithm.

MINVALUE(R-Factor)

The Minimum R-Factor for a given measure period records the Minimum Ave R-Factor for the worst 5-minute period during the measured period.

For example, if during a 24-hour period there was a 5-minte interval where the Ave R-Factor was 25, and this was the worst score for the 24-hour period, then the Minimum R-Factor for that 24-hour period was 25.

Bin Counts

Description:

The Bin Count KPIs count the number of Media Sessions where the MOS was between a certain range during the measured period.

Σ Session Attempts where MOS < X and MOS > Y

The Bin Counts KPI for a given measurement period records the number of Media Sessions where the MOS was between a certain range. The available Bin KPIs are the following:

- Excellent (Total and percentage calculated as ExcellentBin/SessionAttempts)
- Good (Total and percentage calculated as GoodBin/SessionAttempts)
- Fair (Total and percentage calculated as FairBin/SessionAttempts)
- Poor(Total and percentage calculated as PoorBin/SessionAttempts)
- Unacceptable (Total and percentage calculated as UnacceptableBin/SessionAttempts)

Note: In direct RTP monitoring, the MOS depends on where the probes are monitoring the streams.

IPI RTCP-XR KEY PERFORMANCE INDICATOR

RTCP-XR is an extended set of statistics described in RFC3611. IPI provides the Worst and Average value for the set below described of the available parameters defined in the RFC.

Worst and Average value of the metrics defined in the IETF document are provided for Local and Remote as detailed in the following section of this document for each metric.

For a specific “X” metric, the Average value is provided with the following formula:

= Σ (value of X in the selected period) / (Number of calls providing the X metric in the selected period).

The Worst value for the following Local or Remote metrics is calculated as the minimum value in the selected period:

- Signal Level,
- MOS LQ,
- MOS CQ,
- R Factor Listening Quality,
- R Factor Conversational Quality,
- External R Factor In,
- External R Factor Out

For all the other Remore or Local metrics, the Worst value is calculated as the maximum in the selected period.

Session Attempts

Description:

Counts the number of Media Streams reported during the measured period

Σ Session Attempts

Session Completions

Description:

Counts the number of Media Streams that completed normally during the measured period.

Σ Media Session Completed

A Media Session Completion is determined in the signaling by looking at the reported G10 Timeouts, and the SIP Response Code on the SIP Leg. The Application can be configured so that all SIP Response Codes and Timeouts can be mapped to Success, Failure, or Timeout.

NOTE: When a Session is successful based on response code, then it is classified based on MOS_LQ , and it is successful when:

- if Remote and Local MOS LQ are available, both are Fair, Good or Excellent
- if only Remote MOS LQ is available, it is Fair, Good or Excellent
- if only Local MOS LQ is available, it is Fair, Good or Excellent

Session Failures

Description:

Counts the number of Media Streams that failed normally during the measured period.

Σ Media Session Failed

A Media Session Failure is determined in the signaling by looking at the reported G10 Timeouts, and the SIP Response Code on the SIP Leg. The Application can be configured so that all SIP Response Codes and Timeouts can be mapped to Success, Failure, or Timeout.

NOTE: When a Session is successful based on response code, then it is classified based on MOS_LQ , and it is failed when:

- Remote or Local MOS LQ is Poor or Unacceptable.
- both Remote and Local MOS LQ IE are NOT available. In this case, the session will have response cause “No MOS Available”. By default, such value is mapped into failures, but it is possible to configure IPI to put such response cause as a success.

Session Failure Percentage (Session Failure %)

Description:

Computes the percentage of Media Streams that failed normally during the measured period.

(Σ Failures) / (Σ Session Attempts) * 100

Session Timeouts

Description:

Counts the number of Media Streams that timed out normally during the measured period.

Σ Media Session Timeouts

A Media Session Timeout is determined in the signaling by looking at the reported G10 Timeouts, and the SIP Response Code on the SIP Leg. The Application can be configured so that all SIP Response Codes and Timeouts can be mapped to Success, Failure, or Timeout.

Session Timeout Percentage (Session Timeout %)

Description:

Computes the percentage of Media Streams that timed out normally during the measured period.

(Σ Timeouts) / (Σ Session Attempts) * 100

MOS-LQ

The estimated mean opinion score for listening quality (MOS-LQ) is a voice quality metric on a scale from 1 to 5, in which 5 represents excellent and 1 represents unacceptable. This metric is defined as not including the effects of delay and can be compared to MOS scores obtained from listening quality (ACR) tests.

For this metric, IPI provides Average and Worst value for both Local and Remote.

MOS-CQ

The estimated mean opinion score for conversational quality (MOS-CQ) is defined as including the effects of delay and other effects that would affect conversational quality. The metric may be calculated by converting an R factor determined according to ITU-T G.107 [6] or ETSI TS 101 329-5 [3] into an estimated MOS using the equation specified in G.107. It is expressed as an integer in the range 10 to 50, corresponding to MOS x 10, as for MOS-LQ.

For this metric, IPI provides Average and Worst value for both Local and Remote.

R Factor Listening Quality

It refers to ListeningQualityR. This field reports the listening quality expressed as an R factor (per G.107). This does not include the effects of echo or delay. The range of R is 0-95 for narrowband calls and 0-120 for wideband calls. Algorithms for computing this value SHOULD be compliant with ITU-T recommendations P.564 [10] and G.107 [11].

For this metric, IPI provides Average and Worst value for both Local and Remote.

R Factor Conversational Quality

It refers to ConversationalQualityR . This field corresponds to "R factor" in RFC3611 in the VoIP Metrics Report Block. This parameter provides a cumulative measurement of voice quality from the start of the session to the reporting time.

The range of R is 0-95 for narrowband calls and 0-120 for wideband calls. Algorithms for computing this value SHOULD be compliant with ITU-T Recommendation P.564 and G.107.

For this metric, IPI provides Average and Worst value for both Local and Remote.

External R Factor In

This field corresponds to "ext. R factor" in RFC3611 in the VoIP Metrics Report Block. This parameter reflects voice quality as measured by the local endpoint for incoming connection on "other" side (refer to RFC3611 for a more detailed explanation). The range of R is 0-95 for narrowband calls and 0-120 for wideband calls. Algorithms for computing this value SHOULD be compliant with ITU-T Recommendation P.564 and G.107.

For this metric, IPI provides Average and Worst value for both Local and Remote.

External R Factor Out

It refers to ExternalR-Out.This field corresponds to "ext. R factor" in RFC3611 in the VoIP Metrics Report Block. Here, the value is copied from RTCP XR message received from the remote endpoint on "other" side of this endpoint refer to RFC3611 for a more detailed explanation). The range of R is 0-95 for narrowband calls and 0-120 for wideband calls.

For this metric, IPI provides Average and Worst value for both Local and Remote.

Jitter Buffer Max

The current maximum jitter buffer delay which corresponds to the earliest arriving packet that would not be discarded. In simple queue implementations this may correspond to the nominal size. In adaptive jitter buffer implementations, this value may dynamically vary.

Units: ms

It is extracted from the JB Maximum field

For this metric, IPI provides Average and Worst value for both Local and Remote.

Jitter Buffer Absolute Max (ms)

jitter buffer absolute maximum delay (JB abs max). This is the absolute maximum delay in milliseconds that the adaptive jitter buffer can reach under worst case conditions.

For this metric, IPI provides Average and Worst value for both Local and Remote.

Jitter Buffer Nominal (ms)

jitter buffer nominal delay (JB nominal). This is the current nominal jitter buffer delay in milliseconds, which corresponds to the nominal jitter buffer delay for packets that arrive exactly on time.

For this metric, IPI provides Average and Worst value for both Local and Remote.

Burst Loss Density

burst density. The fraction of RTP data packets within burst periods since the beginning of reception that were either lost or discarded.

For this metric, IPI provides Average and Worst value for both Local and Remote.

Burst Duration (ms)

The mean duration, expressed in milliseconds, of the burst periods that have occurred since the beginning of reception.

The duration of each period is calculated based upon the packets that mark the beginning and end of that period.

For this metric, IPI provides Average and Worst value for both Local and Remote.

Gap Loss Density

The fraction of RTP data packets within inter-burst gaps since the beginning of reception that were either lost or discarded. The value is expressed as a fixed point number with the binary point at the left edge of the field.

For this metric, IPI provides Average and Worst value for both Local and Remote.

Gap Duration (ms)

The mean duration, expressed in milliseconds, of the gap periods that have occurred since the beginning of reception.

The duration of each period is calculated based upon the packet that marks the end of the prior burst and the packet that marks the beginning of the subsequent burst.

For this metric, IPI provides Average and Worst value for both Local and Remote.

End System Delay (ms)

The most recently estimated end system delay, expressed in milliseconds.

For this metric, IPI provides Average and Worst value for both Local and Remote.

Network Packet Loss Rate (%)

The fraction of RTP data packets from the source lost since the beginning of reception

Units: %

It is extracted from the Loss Rate field

For this metric, IPI provides Average and Worst value for both Local and Remote.

JitterBufferDiscardRate (%)

The fraction of RTP data packets from the source that have been discarded since the beginning of reception, due to late or early arrival, under-run or overflow at the receiving jitter buffer.

Units: %

It is extracted from the Discard Rate field

For this metric, IPI provides Average and Worst value for both Local and Remote.

RoundTripDelay (ms)

The most recently calculated round trip time between RTPinterfaces.

Units: ms

It is extracted from the Round Trip Delay field.

For this metric, IPI provides Average and Worst value for both Local and Remote.

OneWayDelay (ms)

The most recently calculated delay in transporting a packet from the local party to the remote party.

Units: ms

For this metric, IPI provides Average and Worst value for both Local and Remote.

InterarrivalJitter (ms)

An estimate of the statistical variance of the RTP data packet interarrival time.

Units: ms

For this metric, IPI provides Average and Worst value for both Local and Remote.

Signal Level (dB)

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

Units: dB

It is extracted from the signal level field.

For this metric, IPI provides Average and Worst value for both Local and Remote.

Noise Level (dB)

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

Units: dB

It is extracted from the noise level field.

For this metric, IPI provides Average and Worst value for both Local and Remote.

Residual Echo Return Loss(dB)

This is the ratio between the original signal and the echo level after the echo suppression.

Units: bB

It is extracted from the RERL field.

For this metric, IPI provides Average and Worst value for both Local and Remote.

Mean Absolute Jitter (ms)

Correspond to the Mean absolute packet delay variation

Units: ms

It is extracted from the mean_jitter field.

For this metric, IPI provides Average and Worst value for Local.

Bin Counts

Description:

The Bin Count KPIs count the number of Media Sessions where the MOS LQ was between a certain range during the measured period.

Σ Session Attempts where MOS < X and MOS > Y

The Bin Counts KPI for a given measurement period records the number of Media Sessions where the MOS was between a certain range. The available Bin KPIs are the following:

- Excellent (Total number of sessions and percentage calculated as ExcellentBin/SessionAttempts)
- Good (Total number of sessions and percentage calculated as GoodBin/SessionAttempts)
- Fair (Total number of sessions and percentage calculated as FairBin/SessionAttempts)
- Poor (Total number of sessions and percentage calculated as PoorBin/SessionAttempts)
- Unacceptable (Total number of sessions and percentage calculated as UnacceptableBin/SessionAttempts)

RTSP ON IMS CORE NETWORK KPIS

The **Real Time Streaming Protocol** (**RTSP**) is used to control streaming media servers. It enables to establishe and control media sessions between end points. Clients of media servers issue VCR-like commands, such as *play* and *pause*, to facilitate real-time control of playback of media files from the server.

IPI supports KPIs on IMS core network analysed by Media Servers, giving visibility of issues related to such part of the network on delivering video services.

IPI calculates number of attempts, successes and failures, success and failure rate and average latency on the following Procedures of the Real Time Streaming Protocol (RTSP) defined by ietf RFC 2326:

- Announce
- Describe
- Extension
- Get Parameter
- Options
- Pause
- Play
- Record
- Redirect
- Set Parameter
- Setup
- Teardown
- Response With No Request

RTSP Response Codes can be configured to be related to successes or failures. The list of Response Code are provided in the Appendix.

Expression:

- Attempts: ∑ (# Procedure attempts)
- Successful Procedures: ∑ (# Procedure responses with successful response code, correlated with procedure attempts)
- Failures: ∑ (#Procedure responses with failure response code, correlated with procedure attempts)
- Success Rate: (Successful Procedures / Attempts) * 100%
- Failure Rate: (Failures / Attempts) * 100%
- Average Latency (ms) : ∑ (Total Latency for attempted procedures not terminated with a timeout condition) / (Attempts - Attempts terminated with timeout)

APPENDIX

H.248 Response Codes

Table 1 - H.248 Error & Reason Codes

Return Code & Mnemonic	Description (See Notes)
200 Normal Completion	Command completed normally. (See Notes)
400 Message Syntax Error	Syntax error in message
401 Protocol Error	Protocol Error
402 Unauthorized	Unauthorized
403 Transact Syntax Error	Syntax error in transaction request
406 Version Not Supported	Version Not Supported
410 Identifier Incorrect	Incorrect identifier
411 ContextId Unknown	The transaction refers to an unknown ContextId
412 ContextId Unavailable	No ContextIDs available
421 Action Illegal	Unknown action or illegal combination of actions
422 Action Syntax Error	Syntax Error in Action
430 TerminationId Unknown	Unknown TerminationID
431 TerminationId Mismatch	No TerminationID matched a wildcard
432 TerminationId Unavail	Out of TerminationIDs or No TerminationID available
433 TerminationId In Use	TerminationID is already in a Context
434 Terminations Exceeded	Maximum number of Terminations in a Context exceeded
435 TerminationId Context	Termination ID is not in specified Context
440 Package Unsupported	Unsupported or unknown Package
441 Descriptor Missing	Missing Remote or Local Descriptor

Return Code & Mnemonic	Description (See Notes)
442 Command Syntax Error	Syntax Error in Command
443 Command Unknown	Unsupported or Unknown Command
444 Descriptor Unknown	Unsupported or Unknown Descriptor
445 Property Unknown	Unsupported or Unknown Property
446 Parameter Unknown	Unsupported or Unknown Parameter
447 Descriptor Illegal	Descriptor not legal in this command
448 Descriptor Duplicate	Descriptor appears twice in a command
450 Property Undefined	No such property in this package
451 Event Undefined	No such event in this package
452 Signal Undefined	No such signal in this package
453 Statistic Undefined	No such statistic in this package
454 Parameter Undefined	No such parameter value in this package
455 Property Illegal	Property illegal in this Descriptor
456 Property Duplicate	Property appears twice in this Descriptor
457 Parameter Missing	Missing parameter in signal or event
458 Identifier Unexpected	Unexpected Event/Request ID
459 Profile Unknown	Unsupported or Unknown Profile
471 Implied Add Failure	Implied Add for Multiplex failure
500 Internal MG SW Failure	Internal software Failure in MG
501 Not Implemented	Not Implemented
502 Not Ready	Not ready
503 Service Unavailable	Service Unavailable
504 Command Unauthorized	Command Received from unauthorized entity
505 Transaction Seq Error	Transaction Request Received before a Service Change Reply has been received

Return Code & Mnemonic	Description (See Notes)
506 Transactions Exceeded	Number of Transaction Pendings Exceeded
510 Insufficient Resources	Insufficient resources
512 Event Unequipped	Media Gateway unequipped to detect requested Event
513 Signal Unequipped	Media Gateway unequipped to generate requested Signals
514 Announce Unequipped	Media Gateway cannot send the specified announcement
515 Media Unsupported	Unsupported Media Type
517 Mode Unsupported	Unsupported or invalid mode
518 Event Buffer Full	Event buffer full
519 Digit Map Full	Out of space to store digit map
520 Digit Map Undefined	Digit Map undefined in the MG
521 Termination Changing	Termination is "ServiceChangeing"
526 Insufficient Bandwidth	Insufficient bandwidth
529 Internal MG HW Failure	Internal hardware failure in MG
530 Network Temp Failure	Temporary Network failure
531 Network Perm Failure	Permanent Network failure
532 Audited Nonexistent	Audited Property, Statistic, Event or Signal does not exist
533 PDU Size Exceeded	Response exceeds maximum transport PDU size
534 Property Illegal	Illegal write or read only property
540 State Unexpected	Unexpected initial hook state
581 Does Not Exist	Does Not Exist
600 Announce Syntax Error	Illegal syntax within an announcement specification
601 Variable Unsupported	Variable type not supported
602 Variable Range Error	Variable value out of range
603 Category Unsupported	Category not supported

Return Code & Mnemonic	Description (See Notes)
604 Selector Type Error	Selector type not supported
605 Selector Value Error	Selector value not supported
606 SegmentId Unknown	Unknown segment ID
607 Provisioning Mismatch	Mismatch between play specification and provisioned data
608 Provisioning Error	Provisioning error
609 Offset Invalid	Invalid offset
610 SegmentId Unavailable	No free segment Ids
611 Segment Not Found	Temporary segment not found
612 Segment In Use	Segment in use
613 ISP Port Limit Overrun	ISP port limit overrun
614 Modem Unavailable	No modems available
615 Calling Nbr Unaccept	Calling number unacceptable
616 Called Nbr Unaccept	Called number unacceptable
617 Operation Terminated	Premature termination of operation.
618 Key Sequence Invalid	Invalid command key sequence detected.
619 Max Attempts Exceeded	Maximum attempts exceeded.
620 Digits Not Collected	No digits were collected after <MaxAttempts> prompts.
622 Speech Not Collected	No speech was collected after <MaxAttempts> prompts.
623 Storage Overrun	Out of storage.
624 Segment Delete Error	Unable to delete temporary audio segment.
900 Service Restored	Service Restored
901 Cold Boot	Cold Boot
902 Warm Boot	Warm Boot
903 MGC Directed Change	MGC Directed Change

Return Code & Mnemonic	Description (See Notes)
904 Termination Malfunc	Termination malfunctioning
905 Termination OutOfServ	Termination taken out of service
906 Connectivity Loss	Loss of lower layer connectivity (e.g. downstream sync)
907 Transmission Failure	Transmission Failure
908 MG Impending Failure	MG Impending Failure
909 MGC Impending Failure	MGC Impending Failure
910 Media Capab Failure	Media Capability Failure
911 Modem Capab Failure	Modem Capability Failure
912 Mux Capab Failure	Mux Capability Failure
913 Signal Capab Failure	Signal Capability Failure
914 Event Capab Failure	Event Capability Failure
915 State Loss	State Loss
916 Package Change	Packages Change
917 Capability Change	Capabilities Change
8000 Failed due to Unacceptable MOS	If the average MOS for the stream falls within the range of "Unacceptable" (see MOS bin definitions in Appendix 0) then the Session will be mapped to 8000.
8001 Failed due to Poor MOS	If the average MOS for the stream falls within the range of "Poor" (see MOS bin definitions in Appendix 0) then the Session will be mapped to 8000.

NOTES:

"Normal Completion" Result Code – The "Normal Completion" result code never appears in H.248 messaging. It is only used for Call Trace Result Code display and Message Statistics accumulation purposes. A value of 200 is used to represent "Normal Completion" wherever numeric encoding is needed (e.g. in xDR formatting).

Error Codes 4xx – 5xx – Error Codes in the range 400 through 599 are documented in H.248.8 [8].

Error Codes 6xx – Application-specific Error Codes in the range 600 – 699 are mainly taken from H.248.9 [9].

Reason Codes 9xx – Reason Codes in the range 900 – 999 are documented in H.248.8 [8].

Mnemonic & Description Handling – The Return Code + Mnemonic text shall be used for display of return values in the Compressed PDU Display (as well as Call Trace Call Summary windows & menus). The full Description text shall be used in the Expanded PDU Display.

H.248 Call Types

Table 2 - H.248 Call Types

Call Type	Description
Context	H.248 Context call opened by first Add transaction adding Termination to new Context (e.g. Context = $ {Add …} a.k.a. “CHOOSE Context” action) from Call Agent, closed by Subtract or Move transaction from Call Agent that removes last Termination from Context. Context call also comprises Modify, Move, Notify, AuditValue and AuditCapabilities CTTP transactions matching active non-Null Contexts.
Modify	H.248 Modify call comprising only Modify transaction from Call Agent for Terminations in the Null Context (i.e. Modify transactions for Terminations in active non-Null Contexts are correlated to matching Context call instead). A Modify call is opened and closed by a single Modify CTTP transaction.
Notify	H.248 Notify call comprising only Notify transaction from Media Gateway for Terminations in the Null Context (i.e. Notify transactions for Terminations in active non-Null Contexts are correlated to matching Context call instead). A Notify call is opened and closed by a single Notify CTTP transaction.
AuditValue	H.248 AuditValue call comprising only AuditValue transaction from Call Agent for Terminations in the Null Context (i.e. AuditValue transactions for Terminations in active non-Null Contexts are correlated to matching Context call instead). An AuditValue call is opened and closed by a single AuditValue CTTP transaction.
AuditCapabilities	H.248 AuditCapabilities call comprising only AuditCapabilities transaction from Call Agent for Terminations in the Null Context (i.e. AuditCapabilities transactions for Terminations in active non-Null Contexts are correlated to matching Context call instead). An AuditCapabilities call is opened and closed by a single AuditCapabilities CTTP transaction.
ServiceChange	H.248 ServiceChange call comprising only ServiceChange transaction from either Call Agent or Media Gateway.
WildcardSubtract	This call type is used only for the special where a Subtract command has “ALL” wildcards in both Context Id & Termination Id and has requested return of a wildcard (i.e. merged) Response. In this case, existing Context calls matching the wildcard values are marked with “Wildcard Subtract” status indication then closed if no more Terminations remain in the context. The messages for the invoking wildcard CTTP transaction are then placed in a separate WildcardContext call type (also marked with “Wildcard Subtract” status indicator) that is then immediately closed.

NOTES

(1) **CTTP Transaction** – The term "CTTP Transaction" is used in this table to distinguish a GeoProbe Call & Transaction Tracking Process (CTTP) Transaction (i.e. Command + Response pseudo-message pairs) from an H.248 Transaction.

MGCP Release Causes

RESPONSE_CODE	NAME
0	Response acknowledgment
100	Pending
101	Pending queued
200	OK
250	Connection deleted
400	Transient error
401	Phone off hook
402	Phone on hook
403	Insufficient resources
404	Insufficient bandwidth
405	Endpoint restarting
406	Transaction timeout
407	Transaction aborted
409	Internal overload
410	Endpoint unavailable
411	ContextId Unknown
412	ContextId Unavailable
421	Action Illegal
422	Action Syntax Error
430	TerminationId Unknown
431	TerminationId Mismatch

RESPONSE_CODE	NAME
432	TerminationId Unavail
433	TerminationId In Use
434	Terminations Exceeded
435	TerminationId Context
440	Package Unsupported
441	Descriptor Missing
442	Command Syntax Error
443	Command Unknown
444	Descriptor Unknown
445	Property Unknown
446	Parameter Unknown
447	Descriptor Illegal
448	Descriptor Duplicate
450	Property Undefined
451	Event Undefined
452	Signal Undefined
453	Statistic Undefined
454	Parameter Undefined
455	Property Illegal
456	Property Duplicate
457	Parameter Missing
458	Identifier Unexpected
459	Profile Unknown
471	Implied Add Failure
500	Endpoint unknown
501	Endpoint not ready
502	Insufficient resources
503	Wildcard too complex

RESPONSE_CODE	NAME
504	Command unknown
505	Descriptor unknown
506	Descriptor conflict
507	Unsupported functionality
508	Quarantine unknown
509	Descriptor error
510	Protocol error
511	Unrecognized extension
512	Event unsupported
513	Signal unsupported
514	Announcement invalid
515	Connection unknown
516	Call id unknown
517	Mode invalid
518	Package invalid
519	Digit map unavailable
520	Endpoint restarting
521	Endpoint redirected
522	Event unknown
523	Action unknown
524	Internal inconsistency
525	Extension unknown
526	Insufficient bandwidth
527	Missing descriptor
528	Incompatible version
529	Internal failure
530	CAS signaling error
531	Trunk grouping failure

RESPONSE_CODE	NAME
532	Options unsupported
533	Response too large
534	Negotiation failure
535	Packet period invalid
536	Method unknown
537	Digit map ext unknown
538	Parameter error
539	Parameter invalid
540	Connections exceeded
541	Options Invalid
581	Does Not Exist
600	Announce Syntax Error
601	Variable Unsupported
602	Variable Range Error
603	Category Unsupported
604	Selector Type Error
605	Selector Value Error
606	SegmentId Unknown
607	Provisioning Mismatch
608	Provisioning Error
609	Offset Invalid
610	SegmentId Unavailable
611	Segment Not Found
612	Segment In Use
613	ISP Port Limit Overrun
614	Modem Unavailable
615	Calling Nbr Unaccept
616	Called Nbr Unaccept

RESPONSE_CODE	NAME
617	Operation Terminated
618	Key Sequence Invalid
619	Max Attempts Exceeded
620	Digits Not Collected
622	Speech Not Collected
623	Storage Overrun
624	Segment Delete Error
800	Package specific 800
801	Package specific 801
802	Package specific 802
803	Package specific 803
804	Package specific 804
805	Package specific 805
806	Package specific 806
807	Package specific 807
808	Package specific 808
809	Package specific 809
900	Service Restored
901	Cold Boot
902	Warm Boot
903	MGC Directed Change
904	Termination Malfunc
905	Termination OutOfServ
906	Connectivity Loss
907	Transmission Failure
908	MG Impending Failure
909	MGC Impending Failure
910	Media Capab Failure

RESPONSE_CODE	NAME
911	Modem Capab Failure
912	Mux Capab Failure
913	Signal Capab Failure
914	Event Capab Failure
915	State Loss
916	Package Change
917	Capability Change
1000	CRCX Ack
1001	MDCX Ack
1002	DLCX Ack
1003	RQNT Ack
1004	NTFY Ack
1005	EPCF Ack
1006	AUEP Ack
1007	AUCX Ack
1008	RSIP Ack
1009	RESP Ack
1010	Pending wait Ack
1011	MESG Ack
1028	Tran timeout
1029	Wait DLCX
2000	GeoProbe timeout
9003	Poor or unacceptable MOS
9999	UNKNOWN

MGCP Timeout

RESPONSE_CODE	NAME
1000	CRCX Ack
1001	MDCX Ack
1002	DLCX Ack
1003	RQNT Ack
1004	NTFY Ack
1005	EPCF Ack
1006	AUEP Ack
1007	AUCX Ack
1008	RSIP Ack
1009	RESP Ack
1010	Pending wait Ack
1011	MESG Ack
1028	Tran timeout
1029	Wait DLCX
2000	GeoProbe timeout

Direct RTP Response Codes

Table 03 – Direct RTP Result Codes

Result Code & Mnemonic	Description
200 Normal Completion	RTP Stream terminated normally by detection of RTCP BYE command with no Reason given.
201 Normal with Reason	RTP Stream terminated normally by detection of RTCP BYE command with Reason given.
202 Inactivity Completion	RTP Stream terminated due to Inactivity Timeout.
8000 Failed due to Unacceptable MOS	If the average MOS for the stream falls within the range of “Unacceptable” (see MOS bin definitions in Appendix) then the Session will be mapped to 8000.
8001 Failed due to Poor MOS	If the average MOS for the stream falls within the range of “Poor” (see MOS bin definitions in Appendix) then the Session will be mapped to 8000.

NOTES:

Phase 1 Result Codes – For Phase 1 of the GeoProbe feature, only result code 202 “Inactivity Completion” will be possible since PsyVoIP does not factor in RTCP BYE signals into RTP Session End determination.

Direct RTP Timeouts

Table 4 - Direct RTP Timeouts

Timeout Event	Call Impact	Description
RTP Stream Inactivity Timeout	The Stream is closed, call impact is command dependent	Active RTP Stream monitor records with no RTP Packets detected within “RTP Stream Inactivity Timeout” seconds are closed.

RTSP Release Causes

RESPONSE_CODE	RESPONSE CODE NAME
250	Low on Storage Space
451	Parameter Not Understood
452	Conference Not Found
453	Not Enough Bandwidth
454	Session Not Found
455	Method Not Valid in This State
456	Header Field Not Valid for Resource
457	Invalid Range
458	Parameter Is Read-Only
459	Aggregate Operation Not Allowed
460	Only Aggregate Operation Allowed
461	Unsupported Transport
462	Destination Unreachable
551	Option Not Supported
70000	Timeout
70002	Reserved
70004	Transaction Incomplete
100	Continue
400	Bad Request
401	Unauthorized
402	Payment Required

RESPONSE_CODE	RESPONSE CODE NAME
403	Forbidden
404	Not Found
405	Method Not Allowed
406	Not Acceptable
407	Proxy Authentication Required
408	Request Timeout
410	Gone
411	Length Required
412	Precondition Failed
413	Request Entity Too Large
414	Request-URI Too Large
415	Unsupported Media Type
300	Multiple Choices
301	Moved Permanently
302	Moved Temporarily
303	See Other
304	Not Modified
305	Use Proxy
500	Internal Server Error
501	Not Implemented
502	Bad Gateway
503	Service Unavailable
504	Gateway Timeout
505	RTSP Version Not Supported

RESPONSE_CODE	RESPONSE CODE NAME
200	OK
201	Created
70001	No Release Cause
70003	Reserved Success
70099	Unhandled failure

E-Model R-Factor

Overview of E-Model R-Factor

One standardized approach to Voice Quality measurement is defined by the so-called "E-Model". The E-Model and its associated R-Factor score are specified in ITU-T G.107 [4] and associated applications document ITU-T G.108 [5]. In addition, ITU-T G.109 [6] elaborates on the definition and meaning of the R-Factor speech scores.

In particular, G.108 Annex B section B.3 gives examples of how the E-Model would be applied to Voice carried over impaired IP networks using different types of Speech codecs. In addition, ITU-T G.113 [7] Appendix I contains up to date values for "Equipment Impairment Factor" and Appendix II has guidance for "Advantage Factor A" to be used for VoIP networks. The GeoProbe Projected Voice Quality Measurements uses these algorithms and factors to calculate R-Factor for each monitored call.

Finally, G.107 Annex B specifies how MOS scores are to be calculated given an R-Factor rating. This algorithm is used by GeoProbe to report MOS scores from RTP QoS measurements

E-Model R-Factor Calculations

Overview:

According to G.107 (03/2003) [4] Section 3.1, overall R-Factor calculation is the sum of 5 components:

$R = Ro – Is – Id – Ie,eff + A$

where

Ro	= Basic Signal-to-Noise Ratio
Is	= Simultaneous Impairment Factor
Id	= Delay Impairment Factor
Ie,eff	= Effective Equipment Impairment Factor
A	= Advantage Factor

Each of these factors represents a compound calculation of its own derived from a series of network characteristics and measured parameters. Since many of the underlying parameters are not measurable by the feature and those that are measurable are subject to interpretation, the following sections document how each of these factors should be calculated. In general, where parameters are not measurable by the GeoProbe system, default values specified in Table 2/G.107 [4] will apply, sometimes configurable by the GeoProbe user.

Basic Signal-to-Noise Ratio (*Ro*)

According to G.107 (03/2003) [4] Section 3.2, *Ro* is a complex calculation involving Send Loudness Rating (SLR), Receive Loudness Rating (RLR), Listener Sidetone Rating (LSTR) and various Noise factors including

Circuit Noise (Nc), Sender Room Noise (Nos), Receiver Room Noise (Nor), and Receiver Noise Floor (Nfo). Since none of these factors is independently measurable or predictable by GeoProbe, all applicable default values in Table 2/G.107 [4] are used. This yields a fixed value for *Ro* of

Ro = *93.2* (Basic Signal-to-Noise Ratio)

Note that this value for *Ro* incorporates all default values in Table 2/G.107 [4] , including default values that affect *Is*, *Id* and *Ie*.

Simultaneous Impairment Factor (*Is*)

According to G.107 (03/2003) [4] Section 3.3, *Is* is the sum of 3 components:

Is = Iolr + Ist + Iq

where

Iolr = Too-low Overall Loudness Rating

Ist = Non-optimal Listener Sidetone

Iq = Quantization Distortion

Each of these factors is a complex calculation involving Sidetone Masking Rating (STLR), Mean One-way Echo Path Delay (T), Talker Echo Loudness Rating (TELR) and Quantization Distortion Units (QDU). Most of these components are not measurable or predictable by GeoProbe and so the default values in Table 2/G.107 [4] are used.

The one exception is Quantization Distortion Units. Beginning with G.113 (02/2001) [7], QDU apply only to PCM-based equipment (e.g. Terminals, Gateways etc using G.711 encoding) and not to Low Bitrate Codecs commonly found in packet networks (these latter use the Equipment Impairment method instead – see Section 0 below). One QDU should be assigned for each G.711 codec pair encountered in the speech path. For example, a packet network with all G.729A codecs in the speech path is assigned 0 QDU, but a packet network using G.711 terminals inter-working with a TDM Gateway with Transcoder would be assigned 2 QDU. In principle the GeoProbe could be aware of these factors and assign variable QDU values (e.g. by inspection of the RTP Payload Type code and knowing which RTP Endpoints represented Transcoding Gateways. In practice, the effect of QDU on overall R-Factor score is small enough that this refinement is held out as a possible future enhancement.

Thus *Is* in the current release is determined by taking all applicable defaults in Table 2/G.107 [4] . The default value of QDU = 1 is reasonable given that a large percentage of traffic will encounter at least one TDM Gateway in a typical packet network. However since QDU = 1 has already been accounted for in the *Ro* value above, the resulting *Is* value is

Is = *0.0* (Simultaneous Impairment Factor)

Delay Impairment Factor (*Id*)

According to G.107 (03/2003) [4] Section 3.4, *Id* is the sum of 3 components:

Id = Idte + Idle + Idd

where

Idte = Talker Echo

Idle = Listener Echo

Idd = Too-long Mean One-Way Delay (*Ta*)

Talker Echo and Listener Echo components depend on parameters that are not measurable or predictable by GeoProbe and so the default values in Table 2/G.107 [4] are used. Since these defaults have already been accounted for in *Ro* above, the contribution from *Idte* and *Idle* is 0.

Too-long Mean One-Way Delay (*Idd*) is dependent solely on *Ta* (Mean End-to-End One-Way Delay) which is measurable by the GeoProbe. As demonstrated in G.108 [5] Annex B.3, *Ta* is a function of the Codec Type and GeoProbe network measurements as follows:

Ta = Tp + Tn

where

Tp = One-Way Packetization Delay (from Table 5 for the applicable Codec Type)

Tn = Mean One-Way Network Delay (average RTP packet delay measured by GeoProbe)

Thus Id is calculated from equation (27) in G.107 (which is a function of *Ta*) as follows:

Id = Idd = Function_(27)/G.107(Ta) (Delay Impairment Factor)

Equipment Impairment Factor (*Ie,eff*)

According to G.107 (03/2003) [4] Section 3.5, the Effective Equipment Impairment Factor *Ie,eff* is calculated as follows:

Ie,eff = Ie + (95 – Ie) x *(Ppl / (Ppl + Bpl))*

where

Ie = Baseline Equipment Impairment Factor (from Table 5 for the applicable Codec Type)

Ppl = Probability of Packet Loss (Packets Lost % measured by GeoProbe)

Bpl = Packet-loss Robustness Factor (from Table 5 for the applicable Codec Type)

Note that in this formula, **it is imperative to use percent value for *Ppl*, not fractional value!** For example, if the packet loss measured by GeoProbe is 5%, then Ppl = 5 in the formula above, NOT 0.05. Also note that for the Call Summary display, *Ppl* should be taken from the periodic RTCP Receiver Report Block “Fraction Lost” field, whereas for the Average R-Factor score in RTP QoS Statistics and RTCP DR data records *Ppl* should be the Total RTP Packets Lost divided by Total RTP Packets Sent.

Table 5 – E-Model Codec Calculation Parameters

RTP Payload Type Code	Codec Name	Reference	Bit Rate (Kb/s)	Ie Value	Bpl Value	# Frames per Packet	One-Way Packet Delay (*Tp*) (ms)
0	PCMU	G.711 u-Law w/o PLC	64	0	4.3	1	20
		G.711 u-Law with PLC	64	0	25.1	1	20
3	GSM	GSM 06.10, Full-Rate	13	20	10.0	1	20

RTP Payload Type Code	Codec Name	Reference	Bit Rate (Kb/s)	Ie Value	Bpl Value	# Frames per Packet	One-Way Packet Delay (*Tp*) (ms)
4	G723	G.723.1 +VAD	5.3	19	16.1	1	67.5
			6.3	15	16.1	1	67.5
5	DVI4	RFC 3551		0	10.0	1	20
6	DVI4	RFC 3551		0	10.0	1	20
7	LPC	RFC 3551		0	10.0	1	20
8	PCMA	G.711 A-Law w/o PLC	64	0	4.3	1	20
		G.711 A-Law with PLC	64	0	25.1	1	20
9	G722	G.722		0	10.0	1	20
10	L16	RFC 3551		0	10.0	1	20
11	L16	RFC 3551		0	10.0	1	20
12	QCELP	IS-96a	8	21	10.0	1	20
13	CN	RFC 3551		0	10.0	1	20
14	MPA	RFC 3551		0	10.0	1	20
15	G728	G.728	16	7	10.0	1	20
			12.8	20	10.0	1	20
16	DVI4	RFC 3551		0	10.0	1	20
17	DVI4	RFC 3551		0	10.0	1	20
18	G729	G.729	8	11	19.0	1	25
		G.729A + VAD	8	20	19.0	1	25
dyn	G726-40	G.726, G.727	40	2	10.0	1	20
dyn	G726-32	G.726, G.727	32	7	10.0	1	20
dyn	G726-24	G.726, G.727	24	25	10.0	1	20

RTP Payload Type Code	Codec Name	Reference	Bit Rate (Kb/s)	Ie Value	Bpl Value	# Frames per Packet	One-Way Packet Delay (*Tp*) (ms)
dyn	G726-16	G.726, G.727	16	50	10.0	1	20
dyn	G729D	G.729 Annex D	8	20	10.0	1	25
dyn	G729E	G.729 Annex E	8	20	10.0	1	25
dyn	GSM-EFR	GSM 06.60, Enh Full-Rate	12.2	5	10.0	1	20
dyn	L8	RFC 3551		0	10.0	1	20
dyn	RED	RFC 3551		0	10.0	1	20
dyn	VDVI	RFC 3551		0	10.0	1	20
dyn	VSELP	IS-54	8	20	10.0	1	20
dyn	ACELP	IS-641	7.4	10	10.0	1	20
dyn	RCELP	IS-127	8	6	10.0	1	20
dyn	VSELP	Japan PDC	6.7	24	10.0	1	20
dyn	VSELP	GSM 06.20, Half-Rate	5.6	23	10.0	1	20

NOTES on Table 5

This table represents a composite view of Table I.1 and Table I.3/G.113 [8], Table B.7/G.108 [5], and Table 4/RFC3551 [2].

Shaded values for *Ie* and *Bpl* have not been established in G.113 Appendix I [8]. Values listed are recommended default values that the user must be able to reconfigure as needed.

For PCMU (Payload Type = 0) and PCMA (Payload Type = 8), PCL is "Packet Loss Concealment" per G.711 Appendix I. Since these alternatives have the same RTP Payload Type value, the user must be able to configure the *Bpl* values that apply for their network.

RTP Payload Types listed as "dyn" is to be "dynamically assigned" (see RFC3551 [2]), the exact method being outside the scope of RFC3551. Until a standard method is identified, the user must be able to configure these RTP Payload Type codes to values that apply for their network.

All values listed in this table are intended to be configurable by the user, including the ability to add new Payload / Codec Type entries as needed.

Advantage Factor (*A*)

According to G.107 (03/2003) [4] Section 3.6, the Advantage Factor *A* takes on values between 0 to 20 and is used to improve the R-Factor score for those networks where subscribers are willing to tolerate lower speech quality in exchange for “Access Advantage”. For example, it is fairly well established the Mobile Phone subscribers will tolerate poorer voice quality than they expect from their Fixed Network telephone since Mobile network access is much more convenient.

In the GeoProbe implementation, it is recommended that the user be able to configure the Advantage Factor *A* that applies for their network. However it is recommended that the Advantage Factor **not be applied to the R-Factor score**, but rather be used to improve the MOS value accordingly. The rational is that the raw R-Factor score is needed for general usage (e.g. xDR data feeds for Unified Assurance), but the MOS score gives a quick view of “user acceptability” of the call on the Call Summary display.

MOS (Mean Opinion Score)

Overview of Mean Opinion Score (MOS)

Standardized methodology for Subjective Speech Quality scoring is specified in ITU-T P.830 [9].

Subjective testing is the most widely used method of assessing the performance of digital codecs. When the transmission path is digital and/or non-linear, simple objective measurements, such as those specified in Recommendation G.712, are insufficient to ensure adequate transmission performance. The aim of a subjective testing methodology is to measure the degradation contributed by the non-linear part of the transmission path, and hence to ensure that the performance of the complete system is satisfactory. To be suitable for this purpose, the measurements must be:

a) Reliable; and

b) Carried out in a way that takes account of major interactions between the non-linear part and the other parts of the transmission system.

This implies both the ability to assign a unique numerical contribution to each digital process and the ability to use this assigned contribution in conjunction with other impairments to estimate telephone connection performance.

For most applications the ITU-T recommends the use of the Absolute Category Rating (ACR) method using the Listening Quality scale. However, there are times when other scales and rating methods are more suitable and appropriate (e.g. the Listening Effort scale has been found useful when interest centers on good correlation with measures of conversational performance), and these are used as well in this Recommendation. Only where there is a deviation from the use of the ACR method using the Listening Quality scale will it be stated.

MOS Score Calculation

G.107 (03/2003) [4] Annex B specifies how the R-Factor score is to be converted to MOS score for conversational speech calls. Specifically, Equation (B-4) is used for this conversion as follows:

$$MOS = 1 + (0.035 \times R) + R \times (R - 60) \times (100 - R) \times 7E\text{-}6$$

Where

R = Raw R-Factor score + Advantage Factor A

This is shown graphically in Figure 1. Note that as the Advantage Factor increases, the MOS scores improve for any given R-Factor score.

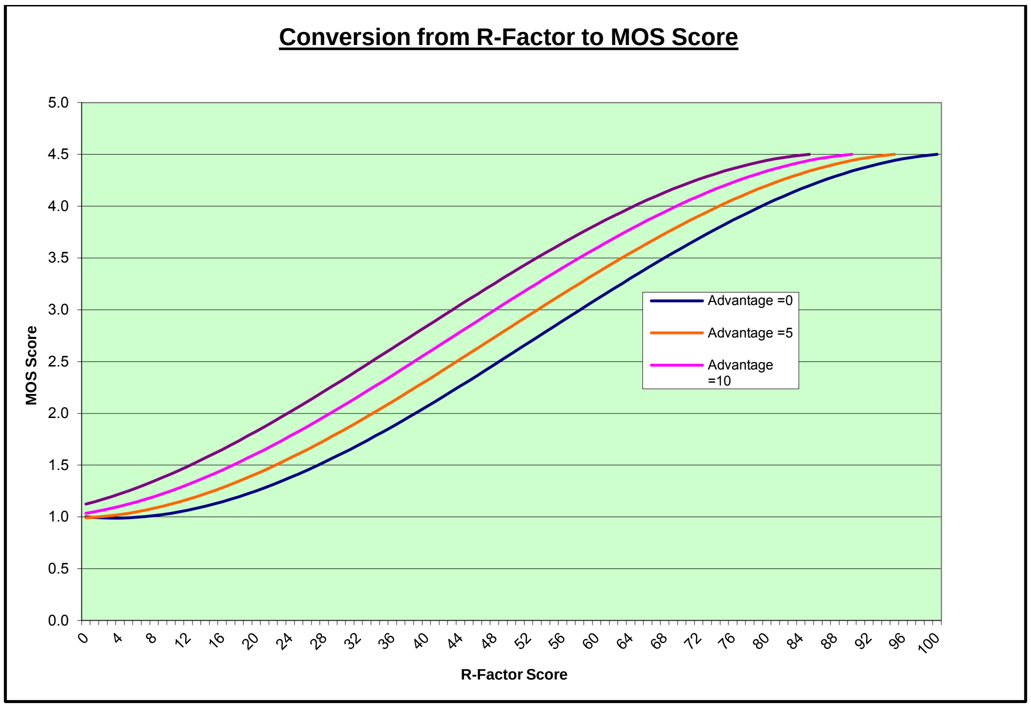


Figure 1 - Conversion from R-Factor to MOS Score

MOS Bin Definition Defaults

Bin	Range
Unacceptable	1.00-2.59
Poor	2.60-3.19
Fair	3.20-3.99
Good	4.00-4.49
Excellent	4.50-5.00
Unknown	0.00-0.99 or 5+

Industry Standards

H.248 Industry Standards

[1] H.248.1 “Gateway control protocol: Version 2”, May 2002, ITU-T.
[2] H.248.2 “Gateway control protocol: Facsimile, text conversation and call discrimination packages”, November 2000, ITU-T.
[3] H.248.3 “Gateway control protocol: User interface elements and actions packages”, November 2000, ITU-T.
[4] H.248.4 “Gateway control protocol: Transport over Stream Control Transmission Protocol (SCTP)”, November 2000, ITU-T.
[5] H.248.5 “Gateway control protocol: Transport over ATM”, November 2000, ITU-T.
[6] H.248.6 “Gateway control protocol: Dynamic Tone Definition package”, November 2000, ITU-T.
[7] H.248.7 “Gateway control protocol: Generic announcement package”, November 2000, ITU-T.
[8] H.248.8 “Gateway control protocol: Error code and service change reason description”, March 2002, ITU-T.
[9] H.248.9 “Gateway control protocol: Advanced media server packages”, March 2002, ITU-T.
[10] H.248.10 “Gateway control protocol: Media gateway resource congestion handling package”, July 2001, ITU-T.
[11] H.248.11 “Gateway control protocol: Media gateway overload control package”, November 2002, ITU-T.
[12] H.248.12 “Gateway control protocol: H.248.1 packages for H.323 and H.324 interworking”, July 2001, with “Amendment 1: New Annex A: Extended H.324, H.245 command and H.245 indication packages”, November 2002, ITU-T.
[13] H.248.13 “Gateway control protocol: Quality Alert Ceasing package”, March 2002, ITU-T.
[14] H.248.14 “Gateway control protocol: Inactivity timer package”, March 2002, ITU-T.
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[16] H.248.16 “Gateway control protocol: Enhanced digit collection packages and procedures”, November 2002, ITU-T.
[17] H.248.17 “Gateway control protocol: Line test packages”, November 2002, ITU-T.
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[19] H.248.19 “Gateway control protocol: Decomposed multipoint control unit, audio, video and data conferencing packages”, March 2004, ITU-T.
[20] H.248.20 “Gateway control protocol: The use of local and remote descriptors with H.221 and H.223 multiplexing”, November 2002, ITU-T.
[21] H.248.21 “Gateway control protocol: Semi-permanent connection handling package”, March 2004, ITU-T.
[22] H.248.22 “Gateway control protocol: Shared Risk Group package”, July 2003, ITU-T.
[23] H.248.23 “Gateway control protocol: Enhanced Alerting packages”, July 2003, ITU-T.
[24] H.248.24 “Gateway control protocol: Multi-frequency tone generation and detection packages”, July 2003, ITU-T.
[25] H.248.25 “Gateway control protocol: Basic CAS packages”, July 2003, ITU-T.
[26] H.248.26 “Gateway control protocol: Enhanced analog lines packages”, July 2003, ITU-T.
[27] H.248.27 “Gateway control protocol: Supplemental tones packages”, July 2003, ITU-T.
[28] H.248.28 “Gateway control protocol: International CAS packages”, March 2004, ITU-T.
[29] H.248.29 “Gateway control protocol: International CAS compelled register signalling packages”, January 2005, ITU-T.
[30] H.248.30 “RTCP extended performance metrics packages”, March 2004, ITU-T.
[31] H.248.31 “Adaptive Jitter Buffer Package”, April 2004, ITU-T.
[32] H.248.32 “Gateway control protocol: Detailed congestion reporting package”, January 2005, ITU-T.
[33] H.248.33 “Gateway control protocol: PCM frame spare bit package”, January 2005, ITU-T.

RTP Industry Standards

[1] RFC 3550 "RTP: A Transport Protocol for Real-Time Applications", July 2003, Internet Engineering Task Force.
[2] RFC 3551 "RTP Profile for Audio and Video Conferences with Minimal Control", July 2003, Internet Engineering Task Force.
[3] RFC 3611 "RTP Control Protocol Extended Reports (RTCP XR)", November 2003, Internet Engineering Task Force.
[4] G.107 "The E-model, a computational model for use in transmission planning", March 2003, ITU-T.
[5] G.108 "Application of the E-model: A planning guide", September 1999, ITU-T.
[6] G.109 "Definition of categories of speech transmission quality", September 1999, ITU-T.
[7] G.113 "Transmission impairments due to speech processing", February 2001, ITU-T.
[8] G.113 Appendix I,
"Provisional planning values for the equipment impairment factor Ie and packet-loss robustness factor Bpl", May 2002, ITU-T.
[9] G.114 "One-way transmission time", May 2003, ITU-T.
[10] P.830 "Subjective Performance Assessment Of Telephone-Band and Wideband Digital Codecs", February 1996, ITU-T.
[11] P.862 "Perceptual evaluation of speech quality (PESQ): An objective method for end-to-end speech quality assessment of narrow-band telephone networks and speech codecs", February 2001, ITU-T.
[12] TR41.1.2/00-05-004 "Voice Quality Recommendations for IP Telephony", Draft 4, 16 May 2000, TIA Standards Project PN-4689.

Acronyms

Following is a definition of the acronyms used in this document.

AG/AGW	Access Gateway
CCA	Call Control Agent
CS	Circuit Switched
KPI	Key Performance Indicator
KQI	Key Quality Indicator
MG	Media Gateway
NRT	Near Real-Time (5-minute delay)
SS7	Signaling System 7
VoIP	Voice over Internet Protocol
xDR	x Data Record (i.e. Data Record of any type)
BC	Bearer Control (H.245)
CEI	Connection Endpoint Identifier
CES	Connection Endpoint Suffix
CPS	Calls Per Second
CS	Call Setup (H.225.0)
DA	Destination Address
IE	Information Element
IP	Internet Protocol
LAN	Local Area Network
MSAN	Media Services Access Node
MOS	Mean Opinion Score
MPC	Multi-Protocol Correlation
MTA	Multimedia Terminal Adapter
PBX	Private Branch Exchange
PDU	Protocol Data Unit
PSTN	Public Switched Telephone Network
QoS	Quality of Service
RTCP	Real-time Transfer Control Protocol
RAS	Registration, Admission, and Status (H.225.0)
RTP	Real-time Transfer Protocol
SAPI	Service Access Point Identifier

SCTP	Stream Control Transmission Protocol
SG	Signaling Gateway
SIC	Service Indicator Code
SS7	Signaling System 7
TA	Terminal adapter
TCP	Transmission Control Protocol
TEI	Terminal Endpoint Identifier
TS	Time Stamp
UDP	User Datagram Protocol
UK	United Kingdom
VoIP	Voice over Internet Protocol