

SIPPING Working Group  
Internet Draft

A. Pendleton  
Nortel  
A. Clark  
Telchemy  
A. Johnston  
Avaya  
H. Sinnreich  
Adobe

Intended status: Informational  
Expires: April 11, 2009

October 7, 2008

## **Session Initiation Protocol Package for Voice Quality Reporting Event**

[draft-ietf-sipping-rtcp-summary-05](#)

### Status of this Memo

By submitting this Internet-Draft, each author represents that any applicable patent or other IPR claims of which he or she is aware have been or will be disclosed, and any of which he or she becomes aware will be disclosed, in accordance with [Section 6 of BCP 79](#).

Internet-Drafts are working documents of the Internet Engineering Task Force (IETF), its areas, and its working groups. Note that other groups may also distribute working documents as Internet-Drafts.

Internet-Drafts are draft documents valid for a maximum of six months and may be updated, replaced, or obsoleted by other documents at any time. It is inappropriate to use Internet-Drafts as reference material or to cite them other than as "work in progress."

The list of current Internet-Drafts can be accessed at <http://www.ietf.org/ietf/1id-abstracts.txt>

The list of Internet-Draft Shadow Directories can be accessed at <http://www.ietf.org/shadow.html>

This Internet-Draft will expire on April 11, 2009.

### Copyright Notice

Copyright (C) The IETF Trust (2008).

### Abstract

This document defines a SIP event package that enables the collection and reporting of metrics that measure the quality for Voice over Internet Protocol (VoIP) sessions.

Pendleton

[Page 1]

[draft-ietf-sipping-rtcp-summary](#)

7 October 2008

## Table of Contents

<a href="#">1.</a>	<a href="#">Introduction.....</a>	<a href="#">2</a>
<a href="#">1.1.</a>	<a href="#">Usage Scenarios.....</a>	<a href="#">2</a>
<a href="#">2.</a>	<a href="#">Terminology.....</a>	<a href="#">4</a>
<a href="#">3.</a>	<a href="#">SIP Events for Voice over IP Performance Reporting.....</a>	<a href="#">5</a>
<a href="#">3.1.</a>	<a href="#">SUBSCRIBE/NOTIFY method.....</a>	<a href="#">5</a>
<a href="#">3.2.</a>	<a href="#">PUBLISH method.....</a>	<a href="#">5</a>
<a href="#">3.3.</a>	<a href="#">Multi-Party and Multi-Segment Calls.....</a>	<a href="#">5</a>
<a href="#">3.4.</a>	<a href="#">Overload Avoidance.....</a>	<a href="#">6</a>
<a href="#">4.</a>	<a href="#">Event Package Formal Definition.....</a>	<a href="#">6</a>
<a href="#">4.1.</a>	<a href="#">Event Package Name.....</a>	<a href="#">6</a>
<a href="#">4.2.</a>	<a href="#">Event Package Parameters.....</a>	<a href="#">6</a>
<a href="#">4.3.</a>	<a href="#">SUBSCRIBE Bodies.....</a>	<a href="#">6</a>
<a href="#">4.4.</a>	<a href="#">Subscription Duration.....</a>	<a href="#">6</a>
<a href="#">4.5.</a>	<a href="#">NOTIFY and PUBLISH Bodies.....</a>	<a href="#">6</a>
<a href="#">4.6.</a>	<a href="#">Voice Quality Event Syntax and Semantics.....</a>	<a href="#">7</a>
<a href="#">4.6.1.</a>	<a href="#">ABNF Syntax Definition.....</a>	<a href="#">8</a>
<a href="#">4.6.2.</a>	<a href="#">Parameter Definitions and Mappings.....</a>	<a href="#">16</a>
<a href="#">4.7.</a>	<a href="#">Message Flow and Syntax Examples.....</a>	<a href="#">24</a>
<a href="#">4.7.1.</a>	<a href="#">End of Session Report using NOTIFY.....</a>	<a href="#">24</a>
<a href="#">4.7.2.</a>	<a href="#">Mid Session Threshold Violation using NOTIFY.....</a>	<a href="#">26</a>
<a href="#">4.7.3.</a>	<a href="#">End of Session Report using PUBLISH.....</a>	<a href="#">28</a>
<a href="#">4.7.4.</a>	<a href="#">Alert Report using PUBLISH.....</a>	<a href="#">30</a>
<a href="#">4.8.</a>	<a href="#">Configuration Dataset for vq-rtcpxr Events.....</a>	<a href="#">31</a>
<a href="#">5.</a>	<a href="#">IANA Considerations.....</a>	<a href="#">32</a>
<a href="#">5.1.</a>	<a href="#">SIP Event Package Registration.....</a>	<a href="#">32</a>
<a href="#">5.2.</a>	<a href="#">application/vq-rtcp-xr MIME Registration.....</a>	<a href="#">32</a>
<a href="#">6.</a>	<a href="#">Security Considerations.....</a>	<a href="#">33</a>
<a href="#">7.</a>	<a href="#">Contributors.....</a>	<a href="#">33</a>
<a href="#">8.</a>	<a href="#">Normative References.....</a>	<a href="#">33</a>
	<a href="#">Author's Addresses.....</a>	<a href="#">34</a>
	<a href="#">Intellectual Property Statement.....</a>	<a href="#">35</a>
	<a href="#">Disclaimer of Validity.....</a>	<a href="#">35</a>

## [1. Introduction](#)

Real time communications over IP networks use SIP for signaling with RTP/RTCP for media transport and reporting respectively. These protocols are very flexible and can support an extremely wide spectrum of usage scenarios. For this reason, extensions to these protocols must be specified in the context of a specific usage scenario. In this memo, extensions to SIP are proposed to support the reporting of Real-Time Control Protocol Extended Reports [4] metrics.

## **1.1. Usage Scenarios**

Pendleton

[Page 2]

[draft-ietf-sipping-rtcp-summary](#)

7 October 2008

The usage scenarios addressed in this memo are situations where a SIP user agent can easily report the voice quality since it communicates with a small number of other endpoints:

1. Point-to-point VoIP conversations. These can include small telephony type multiparty scenarios, such as when using call transfer.
2. Conference calls using a central conferencing server when each SIP endpoint can report on the quality of their leg to the central conferencing server.
3. Multicast VoIP calls using source specific multicast (SSM). This is somewhat similar to the central conferencing scenario.

Distributed conferences with audio mixing in the endpoints may require reporting on too many call legs and may be therefore less practical if there are more than 3-4 participants.

The usage scenarios 1, 2, and 3 provide voice quality reports that are most closely related to the user experience, since the reporting application resides in the endpoints, such as in SIP UAs (UA). Many SIP UAs however may have limitations as to the footprint of the software and as a result frugal reporting capabilities are preferable.

RTCP reports are usually sent to other participating endpoints in a session which can make collection of performance information by administration or management systems too complex. In the usage scenarios addressed in this memo, the data contained in RTCP XR VoIP metrics reports ([RFC3611](#)[4]) are forwarded to a central collection server systems using SIP.

Applications residing in the server or elsewhere can aid in network management to alleviate bandwidth constraints and also to support customer service by identifying and acknowledging calls of poor quality. Specifying such applications are however beyond the scope

of this paper.

### Keeping it Simple

There is a large portfolio of quality parameters that can be associated with VoIP, but only a minimal necessary number of parameters are included on the RTCP-XR reports:

1. The codec type, as resulting from the SDP offer-answer negotiation in SIP,
2. The burst gap loss density and max gap duration, since voice cut-outs are the most annoying quality impairment in VoIP,
3. Round trip delay because it is critical to conversational

Pendleton

[Page 3]

[draft-ietf-sipping-rtcp-summary](#)

7 October 2008

quality,

4. Conversational quality as a catch-all for other voice quality impairments, such as random distributed packet loss, jitter, annoying silent suppression effects, etc.

In specific usage scenarios where other parameters are required, designers can include other parameters beyond the scope of this paper.

RTCP reports are best effort only, and though very useful have a number of limitations as discussed in [3]. This must be considered when using RTCP reports in managed networks.

This document defines a new SIP event package, `vq-rtcpxr`, and a new MIME type, `application/vq-rtcpxr`, that enable the collection and reporting of metrics that measure quality for RTP [3] sessions. The definitions of the metrics used in the event package are based on RTCP Extended Reports [4] and RTCP [3]; a mapping between the SIP event parameters and the parameters within the forementioned RFC's is defined within this document in [section 4.6.2](#).

Monitoring of voice quality is believed to be the highest priority for usage of this mechanism and as such, the metrics in the event package are largely tailored for voice quality measurements. The event package is designed to be extensible. However the negotiation of such extensions is not defined in this document.

The event package supports reporting both the voice quality metrics for both inbound and outbound directions. Voice quality metrics for the inbound direction can generally be computed locally by the reporting endpoint however voice quality metrics for the outbound

direction are computed by the remote endpoint and sent to the reporting endpoint using the RTCP Extended Reports [4].

Configuration of usage of the event package is not covered in this document. It is the recommendation of this document that the SIP configuration framework [8] be used. The authors have defined a configuration dataset that would facilitate this support in [section 5.8](#).

The event package SHOULD be used with the SUBSCRIBE/NOTIFY method however it MAY be also used with the PUBLISH method for backward compatibility with some existing implementations. Message flow examples for both methods are provided in this document.

## **2. Terminology**

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in [RFC 2119](#) [1].

Pendleton

[Page 4]

[draft-ietf-sipping-rtcp-summary](#)

7 October 2008

## **3. SIP Events for Voice over IP Performance Reporting**

This document defines a SIP events package [5] for Voice over IP performance reporting. A SIP UA can send these events to an entity which can make the information available to other applications. For purposes of illustration, the entities involved in SIP vq-rtcprr event reporting will be referred to as follows:

- o REPORTER is an entity involved in the measurement and reporting of media quality i.e. the SIP UA involved in a media session.
- o COLLECTOR is an entity that receives SIP vq-rtcprr events. A COLLECTOR may be a proxy server or another entity that is capable of supporting SIP vq-rtcprr events.

### **3.1. SUBSCRIBE/NOTIFY method**

The REPORTER SHOULD send the voice quality metric reports using the NOTIFY method. The COLLECTOR SHALL send a SUBSCRIBE to the REPORTER to explicitly establish the relationship. Configuration of an address of the COLLECTOR is not needed as explained below.

The REPORTER MUST NOT send any vq-rtcprr events if a COLLECTOR address has not been configured.

The REPORTER SHALL populate the Request-URI of the NOTIFY method

with the address of the COLLECTOR.

The COLLECTOR MAY send a SUBSCRIBE to a SIP Proxy acting on behalf of the reporting SIP UA's.

### **3.2. PUBLISH method**

A SIP UA that supports this specification MAY also send the service quality metric reports using the PUBLISH method, however this approach SHOULD NOT be used in unmanaged Internet services. The Publish method MAY be supported for backward compatibility with existing implementations.

The REPORTER MAY therefore populate the Request-URI of the PUBLISH method with the address of the COLLECTOR. To ensure security of SIP proxies and the COLLECTOR, the REPORTER MUST be configured with the address of the COLLECTOR, preferably using the SIP UA configuration framework [8], as described in [section 5.8](#).

It is recommended that the REPORTER send an OPTIONS message to the COLLECTOR to ensure support of the PUBLISH message.

### **3.3. Multi-Party and Multi-Segment Calls**

A voice quality metric report may be sent for each session terminating at the REPORTER and may contain multiple report bodies. For a multi-party call the report MAY contain report bodies for the

Pendleton

[Page 5]

[draft-ietf-sipping-rtcp-summary](#)

7 October 2008

session between the reporting endpoint and each remote endpoint for which there was an RTP session during the call.

Multi-party services such as call hold and call transfer can result in the user participating in a series of concatenated sessions, potentially with different choices of codec or sample rate, although these may be perceived by the user as a single call. A REPORTER MAY send a voice quality metric report at the end of each session or MAY send a single voice quality metric report containing a report body for each segment of the call.

### **3.4. Overload Avoidance**

To avoid overload of SIP Proxies or COLLECTORS it is important to do capacity planning and to minimize the number of reports that are sent.

Approaches to avoiding overload include:

- a. Send only one report at the end of each call

- b. Use interval reports only on "problem" calls that are being closely monitored

Limit the number of alerts that can be sent to a maximum of one per call

Additionally, it is recommended that COLLECTORS that receive these reports use the 503 'Service Unavailable' error response code to limit unwanted reports and include the Retry-after header with an appropriate time delay, depending on the needs of the COLLECTOR.

## **4. Event Package Formal Definition**

### **4.1. Event Package Name**

This document defines a SIP Event Package as defined in [RFC 3265](#) [5].

### **4.2. Event Package Parameters**

No event package parameters are defined.

### **4.3. SUBSCRIBE Bodies**

No SUBSCRIBE bodies are described by this specification.

### **4.4. Subscription Duration**

Subscriptions to this event package MAY range from minutes to weeks. Subscriptions in hours or days are more typical and are RECOMMENDED. The default subscription duration for this event package is one hour.

### **4.5. NOTIFY and PUBLISH Bodies**

Pendleton

[Page 6]

[draft-ietf-sipping-rtcp-summary](#)

7 October 2008

There are three notify bodies: a Session report, an Interval session report, and an Alert report.

The Session report SHOULD be used for reporting when a voice media session terminates or when a media change occurs, such as a codec change or a session forks and MUST NOT be used for reporting at arbitrary points in time. This report MUST be used for cumulative metric reporting and the report timestamps MUST be from the start of a media session to the time at which the report is generated.

The Interval report SHOULD be used for periodic or interval reporting and MUST NOT be used for reporting for the complete media session. This report is intended to capture short duration

metric reporting and the report intervals SHOULD be non-overlapping time windows.

The Alert report MAY be used when voice quality degrades during a session. The time window to which an Alert report relates MAY be a short time interval or from the start of the call to the point the alert is generated; this time window SHOULD be selected to provide the most useful information to support problem diagnosis.

Session, Interval and Alert reports MUST populate the metrics with values that are measured over the interval explicitly defined by the "start" and "stop" timestamps.

This specification defines a new MIME type application/vq-rtcpxr which is a text encoding of the RTCP and RTCP-XR statistics with some additional metrics and correlation information.

#### **4.6. Voice Quality Event Syntax and Semantics**

This section describes the syntax extensions required for event publication in SIP. The formal syntax definitions described in this section are expressed in the Augmented BNF [6] format used in SIP [2], and contains references to elements defined therein. Additionally, the definition of the timestamp format is provided in [7]. Note that most of the parameters are optional. In practice, most implementations will send a subset of the parameters. It is not the intention of this document to define what parameters may or may not be useful for monitoring the quality of a voice session, but to enable reporting of voice quality. As such, the syntax allows the implementer to choose which metrics are most appropriate for their solution. As there are no "invalid", "unknown", or "not applicable" values in the syntax, the intention is to exclude any parameters for which values are not available, not applicable, or unknown.

Additionally, the authors recognize that implementers may need to add new parameter lines to the reports and new metrics to the existing parameter lines. The extension tokens are intended to fulfill this need.

##### **4.6.1. ABNF Syntax Definition**

VQReportEvent = AlertReport / SessionReport / IntervalReport

SessionReport = "VQSessionReport" [COLON CallTerm] CRLF  
                  LocalMetrics [CRLF RemoteMetrics]  
                  [DialogID]

; CallTerm indicates the final report of a session.



```

IntervalReport = "VQIntervalReport" CRLF
                LocalMetrics [CRLF RemoteMetrics]
                [DialogID]

LocalMetrics   = "LocalMetrics" COLON CRLF Metrics

RemoteMetrics = "RemoteMetrics" COLON CRLF Metrics

AlertReport   = "VQAlertReport" COLON
                MetricType WSP Severity WSP Direction CRLF
                "Metrics:" CRLF
                Metrics
                [CRLF "RemoteMetrics:" CRLF Metrics]
                [DialogID]

Metrics = TimeStamps CRLF
        [SessionDescription CRLF]
        CallID CRLF
        FromID CRLF
        ToID CRLF
        LocalAddr CRLF
        RemoteAddr CRLF
        [JitterBuffer CRLF]
        [PacketLoss CRLF]
        [BurstGapLoss CRLF]
        [Delay CRLF]
        [Signal CRLF]
        [QualityEstimates CRLF]
        *(Extension CRLF)

; Timestamps are provided in Coordinated Universal Time (UTC)
; using the ABNF format provided in RFC3339,
; "Date and Time on the Internet: Timestamps"
; These timestamps SHOULD reflect, as closely as
; possible, the actual time during which the media session
; was running to enable correlation to events occurring
; in the network infrastructure and to accounting records

TimeStamps = "Timestamps" COLON StartTime WSP StopTime
StartTime  = "START" EQUAL date-time
StopTime   = "STOP" EQUAL date-time

; SessionDescription provides a shortened version of the

```

```

; session SDP but contains only the relevant parameters for
; session quality reporting purposes

```

```

SessionDescription = "SessionDesc" COLON
    [PayloadType WSP]
    [PayloadDesc WSP]
    [SampleRate WSP]
    [FrameDuration WSP]
    [FrameOctets WSP]
    [FramesPerPacket WSP]
    [PacketsPerSecond WSP]
    [FmtpOptions WSP]
    [PacketLossConcealment WSP]
    [SilenceSuppressionState]
    *(WSP Extension)

; PayloadType provides the PT parameter used in the RTP packets
; i.e. the codec used for decoding received RTP packets
; IANA registered values SHOULD be used where possible.

PayloadType = "PT" EQUAL (1*3DIGIT)

; PayloadDesc provides a text description of the codec
; The abbreviated Standard name for the codec SHOULD be used
; (for example "G.729A")

PayloadDesc = "PD" EQUAL (word / DQUOTE word-plus DQUOTE)

; SampleRate reports the rate at which voice was sampled
; in the case of narrowband codecs, this value will typically
; be 8000.
; For codecs that are able to change sample rates the lowest and
; highest sample rates MUST be reported (e.g. 8000;16000).

SampleRate = "SR" EQUAL (1*5DIGIT) *(SEMI (1*5DIGIT))

; FrameDuration can be combined with the FramesPerPacket
; to determine the packetization rate; the units for
; FrameDuration are milliseconds.

FrameDuration = "FD" EQUAL (1*3DIGIT)

; FrameOctets provides the number of octets in each frame
; This MAY be used where FrameDuration is not available

FrameOctets = "FO" EQUAL (1*4DIGIT)

; FramesPerPacket provides the number of frames in each RTP
; packet

FramesPerPacket = "FPP" EQUAL (1*2DIGIT)

; Packets per second provides the average number of packets

```

; that are transmitted per second.

PacketsPerSecond = "PPS" EQUAL (1\*5DIGIT)

; FMTP options from SDP. Note that the parameter is delineated  
; by " " to avoid parsing issues in transitioning between SDP and  
; SIP parsing

FmtpOptions = "FMTP" EQUAL DQUOTE word-plus DQUOTE

; PacketLossConcealment indicates whether a PLC algorithm was  
; or is being used for the session. The values follow the same  
; numbering convention as [RFC 3611](#)[4].  
; 0 - unspecified  
; 1 - disabled  
; 2 - enhanced  
; 3 - standard

PacketLossConcealment = "PLC" EQUAL ("0" / "1" / "2" / "3")

; SilenceSuppressionState indicates whether silence suppression,  
; also known as Voice Activity Detection (VAD) is enabled.

SilenceSuppressionState = "SSUP" EQUAL ("on" / "off")

; CallID provides the call id from the SIP header

CallID = "CallID" COLON Call-ID-Parm

; FromID provides the identification of the reporting endpoint  
; of the media session [2].

FromID = "FromID" COLON from-spec

; ToID provides the identification of the remote endpoint  
; of the media session [2].

ToID = "ToID" COLON (name-addr/addr-spec)

; LocalAddr provides the IP address, port and ssrc of the  
; endpoint/UA which is the receiving end of the stream being  
; measured.

LocalAddr = "LocalAddr" COLON IPAddress WSP Port WSP Ssrc

; RemoteAddr provides the IP address, port and ssrc of the  
; the source of the stream being measured.

RemoteAddr = "RemoteAddr" COLON IPAddress WSP Port WSP Ssrc

; For clarification, the LocalAddr in the LocalMetrics report  
; MUST be the RemoteAddr in the RemoteMetrics report.

Pendleton

[Page 10]

[draft-ietf-sipping-rtcp-summary](#)

7 October 2008

IPAddress = "IP" EQUAL IPv6address / IPv4address  
Port = "PORT" EQUAL 1\*DIGIT  
Ssrc = "SSRC" EQUAL ( '0x' 1\*8HEXDIG )

JitterBuffer = "JitterBuffer" COLON

[JitterBufferAdaptive WSP]  
[JitterBufferRate WSP]  
[JitterBufferNominal WSP]  
[JitterBufferMax WSP]  
[JitterBufferAbsMax]  
\*(WSP Extension)

; JitterBufferAdaptive indicates whether the jitter buffer in the  
; endpoint is adaptive, static, or unknown.  
; The values follow the same numbering convention as [RFC3611](#).  
; For more details, please refer to that document.  
; 0 - unknown  
; 1 - reserved  
; 2 - non-adaptive  
; 3 - adaptive

JitterBufferAdaptive = "JBA" EQUAL ("0" / "1" / "2" / "3")

; JitterBuffer metric definitions are provided in [RFC3611](#)

JitterBufferRate = "JBR" EQUAL (1\*2DIGIT) ;0-15  
JitterBufferNominal = "JBN" EQUAL (1\*5DIGIT) ;0-65535  
JitterBufferMax = "JBM" EQUAL (1\*5DIGIT) ;0-65535  
JitterBufferAbsMax = "JBX" EQUAL (1\*5DIGIT) ;0-65535

; PacketLoss metric definitions are provided in [RFC3611](#)

PacketLoss = "PacketLoss" COLON  
[NetworkPacketLossRate WSP]  
[JitterBufferDiscardRate]  
\*(WSP Extension)

NetworkPacketLossRate =

"NLR" EQUAL (1\*3DIGIT ["." 1\*2DIGIT]) ;percentage

JitterBufferDiscardRate =

"JDR" EQUAL (1\*3DIGIT [". " 1\*2DIGIT]) ;percentage

; BurstGapLoss metric definitions are provided in [RFC3611](#) [4]

BurstGapLoss = "BurstGapLoss" COLON

[BurstLossDensity WSP]  
[BurstDuration WSP]  
[GapLossDensity WSP]

Pendleton

[Page 11]

[draft-ietf-sipping-rtcp-summary](#)

7 October 2008

[GapDuration WSP]  
[MinimumGapThreshold]  
\*(WSP Extension)

BurstLossDensity =

"BLD" EQUAL (1\*3DIGIT [". " 1\*2DIGIT]) ;percentage

BurstDuration =

"BD" EQUAL (1\*7DIGIT) ;0-3,600,000 -- milliseconds

GapLossDensity =

"GLD" EQUAL (1\*3DIGIT [". " 1\*2DIGIT]) ;percentage

GapDuration =

"GD" EQUAL (1\*7DIGIT) ;0-3,600,000 -- milliseconds

MinimumGapThreshold =

"GMIN" EQUAL (1\*3DIGIT) ;1-255

Delay = "Delay" COLON

[RoundTripDelay WSP]  
[EndSystemDelay WSP]  
[OneWayDelay WSP]  
[SymmOneWayDelay WSP]  
[InterarrivalJitter WSP]  
[MeanAbsoluteJitter]  
\*(WSP Extension)

; RoundTripDelay is recommended to be measured as defined in  
; [RFC3550](#) [3].

RoundTripDelay = "RTD" EQUAL (1\*5DIGIT) ;0-65535

; EndSystemDelay metric is defined in [RFC 3611](#) [4]

EndSystemDelay = "ESD" EQUAL (1\*5DIGIT) ;0-65535

; OneWayDelay is defined in [RFC2679](#)

OneWayDelay = "OWD" EQUAL (1\*5DIGIT) ;0-65535

; SymmOneWayDelay is defined as half the sum of RoundTripDelay

; and the EndSystemDelay values for both endpoints.

SymmOneWayDelay = 'SOWD' EQUAL (1\*5DIGIT); 0-65535

; Interarrival Jitter is measured as defined [RFC 3550](#)

InterarrivalJitter = "IAJ" EQUAL (1\*5DIGIT) ;0-65535

; Mean Absolute Jitter is measured as defined

Pendleton

[Page 12]

[draft-ietf-sipping-rtcp-summary](#)

7 October 2008

; by ITU-T G.1020 [\[10\]](#) where it is known as MAPDV

MeanAbsoluteJitter = "MAJ" EQUAL (1\*5DIGIT);0-65535

; Signal metrics definitions are provided in [RFC 3611](#)

Signal = "Signal" COLON  
[SignalLevel WSP]  
[NoiseLevel WSP]  
[ResidualEchoReturnLoss]  
\*(WSP Extension)

; SignalLevel will normally be a negative value  
; the absence of the negative sign indicates a positive value  
; where the signal level is negative, the sign MUST be included.  
; This metric applies to the speech signal decoded from the  
; received packet stream.

SignalLevel = "SL" EQUAL (["-"] 1\*2DIGIT)

; NoiseLevel will normally be negative and the sign MUST be  
; explicitly included.  
; The absence of a sign indicates a positive value  
; This metric applies to the speech signal decoded from the  
; received packet stream.

NoiseLevel = "NL" EQUAL (["-"] 1\*2DIGIT)

; Residual Echo Return Loss (RERL) the ratio between  
; the original signal and the echo level as measured after  
; echo cancellation or suppression has been applied.

```
; Expressed in decibels (dB). This is typically a positive
; value.
; This metric relates to the proportion of the speech signal
; decoded from the received packet stream that is reflected
; back in the encoded speech signal output in the transmitted
; packet stream (i.e. will affect the REMOTE user's conversational
; quality). To support the diagnosis of echo related problems
; experienced by the local user of the device generating a report
; according to this document, the value of RERL reported via
; the RTCP XR VoIP Metrics payload SHOULD be reported in the
; RemoteMetrics set of data.
```

```
ResidualEchoReturnLoss = "RERL" EQUAL (1*3DIGIT)
```

```
; Voice Quality estimation metrics
; Each quality estimate has an optional associated algorithm.
; These fields permit the implementation to use a variety
; of different calculation methods for each type of metric
```

```
QualityEstimates = "QualityEst" COLON
    [ListeningQualityR WSP]
    [RLQEstAlg WSP]
```

Pendleton

[Page 13]

[draft-ietf-sipping-rtcp-summary](#)

7 October 2008

```
[ConversationalQualityR WSP]
[RCQEstAlg WSP]
[ExternalR-In WSP]
[ExtRInEstAlg WSP]
[ExternalR-Out WSP]
[ExtROutEstAlg WSP]
[MOS-LQ WSP]
[MOSLQEstAlg WSP]
[MOS-CQ WSP]
[MOSCQEstAlg WSP]
[QoEEstAlg]
*(WSP Extension)
```

```
ListeningQualityR = "RLQ" EQUAL (1*3DIGIT) ; 0 - 120
```

```
RLQEstAlg = "RLQEstAlg" EQUAL word ; "P.564", or other
```

```
ConversationalQualityR = "RCQ" EQUAL (1*3DIGIT) ; 0 - 120
```

```
RCQEstAlg = "RCQEstAlg" EQUAL word ; "P.564", or other
```

```
; ExternalR-In is measured by the local endpoint for incoming
; connection on "other" side of this endpoint
; e.g. Phone A <---> Bridge <----> Phone B
; ListeningQualityR = quality for Phone A <----> Bridge path
```

```

;   ExternalR-In = quality for Bridge <---- Phone B path

ExternalR-In = "EXTRI" EQUAL (1*3DIGIT) ; 0 - 120

ExtRInEstAlg = "ExtRIEstAlg" EQUAL word ; "P.564" or other

; ExternalR-Out is copied from RTCP XR message received from the
; remote endpoint on "other" side of this endpoint
;   e.g. Phone A <--> Bridge <----> Phone B
;   ExternalR-Out = quality for Bridge <----> Phone B path

ExternalR-Out = "EXTR0" EQUAL (1*3DIGIT) ; 0 - 120

ExtROutEstAlg = "ExtROEstAlg" EQUAL word ; "P.564" or other

MOS-LQ = "MOSLQ" EQUAL (DIGIT ["." 1*3DIGIT]) ; 0.0 - 4.9

MOSLQEstAlg = "MOSLQEstAlg" EQUAL word ; "P.564" or other

MOS-CQ = "MOSCQ" EQUAL (DIGIT ["." 1*3DIGIT]) ; 0.0 - 4.9

MOSCQEstAlg = "MOSCQEstAlg" EQUAL word ; "P.564" or other

; alternative to the separate estimation algorithms
; for use when the same algorithm is used for all measurements

QoEEstAlg = "QoEEstAlg" EQUAL word; "P.564" or other

```

```

; DialogID provides the identification of the dialog with
; which the media session is related. This value is taken
; from the SIP header.

DialogID = "DialogID" COLON Call-ID-Parm *(SEMI did-parm)

did-parm = to-tag / from-tag / word

to-tag = "to-tag" EQUAL token

from-tag = "from-tag" EQUAL token

; MetricType provides the metric on which a notification of
; threshold violation was based. The more commonly used metrics
; for alerting purposes are included here explicitly and the
; token parameter allows for extension

MetricType = "Type" EQUAL "RLQ" / "RCQ" / "EXTR" /
             "MOSLQ" / "MOSCQ" /
             "BD" / "NLR" / "JDR" /

```



```

"RTD" / "ESD" / "IAJ" /
"RERL" / "SL" / "NL" / Extension

Direction = "Dir" EQUAL "local" / "remote"

Severity = "Severity" EQUAL "Warning" / "Critical" /
"Clear"

Call-ID-Parm = word [ "@" word ]

; General ABNF notation from RFC5234

CRLF = %x0D.0A
DIGIT = %x30-39
WSP = SP / HTAB ; white space
SP = " "
HTAB = %x09 ; horizontal tab
HEXDIG = DIGIT / "A" / "B" / "C" / "D" / "E" / "F" /
"a" / "b" / "c" / "d" / "e" / "f"
DQUOTE = %x22 ; " (Double Quote)
ALPHA = %x41-5A / %x61-7A ; A-Z / a-z

; ABNF notation from RFC3261

alphanum = ALPHA / DIGIT
LWS = [*WSP CRLF] 1*WSP ; linear whitespace
SWS = [LWS] ; sep whitespace
SEMI = SWS ";" SWS ; semicolon
EQUAL = SWS "=" SWS ; equal
COLON = SWS ":" SWS ; colon
token = 1*(alphanum / "-" / "." / "!" / "%" / "*" /
"/" / "_" / "+" / "`" / "'" / "~" )

```

```

IPv4address = 1*3DIGIT "." 1*3DIGIT "." 1*3DIGIT "." 1*3DIGIT

IPv6address = hexpart [ ":" IPv4address ]
hexpart = hexseq / hexseq ":@" [ hexseq ] / "::"
[ hexseq ]
hexseq = hex4 *( ":" hex4)
hex4 = 1*4HEXDIG

```

; ABNF notation from [RFC3339](#)

```

date-fullyear = 4DIGIT ; e.g. 2006
date-month = 2DIGIT ; e.g. 01 or 11
date-mday = 2DIGIT ; e.g. 02 or 22
time-hour = 2DIGIT ; e.g. 01 or 13

```

```

time-minute      = 2DIGIT ; e.g. 03 or 55
time-second      = 2DIGIT ; e.g. 01 or 59
time-secfrac     = "." 1*DIGIT
time-numoffset   = ("+" / "-") time-hour ":" time-minute
time-offset      = "Z" / time-numoffset
partial-time     = time-hour ":" time-minute ":" time-second
[time-secfrac]
full-date        = date-fullyear "-" date-month "-" date-mday
full-time        = partial-time time-offset
date-time        = full-date "T" full-time

;
; Miscellaneous definitions
;

```

Extension = word-plus

```

word = 1*(alphanum / "-" / "." / "!" / "%" / "*" /
  "_" / "+" / "`" / "'" / "~" /
  "(" / ")" / "<" / ">" /
  ":" / "\" / DQUOTE /
  "/" / "[" / "]" / "?" )

word-plus = 1*(alphanum / "-" / "." / "!" / "%" / "*" /
  "_" / "+" / "`" / "'" / "~" /
  "(" / ")" / "<" / ">" / ":" /
  "\" / "/" / "[" / "]" / "?" /
  "{" / "}" / "=" / " ")

```

#### **4.6.2. Parameter Definitions and Mappings**

##### **4.6.2.1. General mapping percentages from 8 bit, fixed point numbers**

[RFC3611](#) uses an 8 bit, fixed point number with the binary point at the left edge of the field. This value is calculated by dividing the total number of packets lost by the total number of packets expected and multiplying the result by 256, and taking the integer part.

Pendleton

[Page 16]

[draft-ietf-sipping-rtcp-summary](#)

7 October 2008

For any RTCP XR parameter in this format, to map into the equivalent SIP vq-rtcpxr parameter, simply reverse the equation i.e. divide by 256 and taking the integer part.

##### **4.6.2.2. Timestamps**

Following SIP and other IETF convention, timestamps are provided in Coordinated Universal Time (UTC) using the ABNF format provided in

IETF [RFC3339](#) [7]. These timestamps SHOULD reflect, as closely as possible, the actual time during which the media session was running to enable correlation to related events occurring in the network and to accounting or billing records.

#### **4.6.2.3. SessionDescription**

The parameters in this field provide a shortened version of the session SDP(s), containing only the relevant parameters for session quality reporting purposes.

##### **Payload Type**

This is the "payload type" parameter used in the RTP packets i.e. the codec. This field can also be mapped from the SDP "rtpmap" attribute field "payload type". IANA registered types SHOULD be used.

##### **Payload Desc**

This parameter provides a text name for the codec used in this session.

##### **Sample Rate**

This parameter is mapped from the SDP "rtpmap" attribute field "clock rate". The field provides the rate at which voice was sampled, measured in Hertz (Hz).

##### **Frame Duration**

This parameter is not contained in RTP or SDP but can usually be obtained from the device codec. The field reflects the amount of voice content in each frame within the RTP payload, measured in milliseconds. Note this value can be combined with the FramesPerPacket to determine the packetization rate.

##### **Frame Octets**

This parameter is not contained in RTP or SDP but is usually provided by the device codec. The field provides the number of octets in each frame within the RTP payload. This field is usually not provided when FrameDuration is provided.

##### **Frames Per Packet**

This parameter is not contained in RTP or SDP but can usually be obtained from the device codec. This field provides the number of

frames in each RTP packet. Note this value can be combined with the `FrameDuration` to determine the packetization rate.

#### Packets Per Second

This parameter is not contained in RTP or SDP but can usually be obtained from the device codec. `Packets per second` provides the (rounded) number of RTP packets that are transmitted per second.

#### FMTMP Options

This parameter is taken directly from the SDP attribute `"fmtmp"`.

#### Silence Suppression State

This parameter does not correspond to SDP, RTP, or RTCP XR. It indicates whether silence suppression, also known as Voice Activity Detection (VAD) is enabled for the identified session.

#### Packet Loss Concealment

This value corresponds to `"PLC"` in [RFC3611](#) in the VoIP Metrics Report Block. The values defined by [RFC3611](#) are reused by this recommendation and therefore no mapping is required.

#### **[4.6.2.4. LocalAddr](#)**

This field provides the IP address, port and synchronization source (SSRC) for the session from the perspective of the endpoint that is measuring performance. The IP address can be IPv4 or IPv6 format. The SSRC is taken from SDP, RTCP, or RTCP XR input parameters.

In the presence of NAT, the MAPPED-ADDRESS as reported by the STUN [\[9\]](#) server ([RFC 3489](#)) MUST be reported, since the internal IP address is not visible to the network operator.

#### **[4.6.2.5. RemoteAddr](#)**

This field provides the IP address, port and ssrc of the session peer from the perspective of the remote endpoint measuring performance. In the presence of NAT, the MAPPED-ADDRESS as reported by the STUN [\[9\]](#) server ([RFC 3489bis](#)) MUST be reported, since the internal IP address is not visible to the network operator.

#### **[4.6.2.6. Jitter Buffer Parameters](#)**

##### Jitter Buffer Adaptive

This value corresponds to `"JBA"` in [RFC3611](#) in the VoIP Metrics Report Block. The values defined by [RFC3611](#) are unchanged and therefore no mapping is required.

#### Jitter Buffer Rate

This value corresponds to "JB rate" in [RFC3611](#) in the VoIP Metrics Report Block. The parameter does not require any conversion.

#### Jitter Buffer Nominal

This value corresponds to "JB nominal" in [RFC3611](#) in the VoIP Metrics Report Block. The parameter does not require any conversion.

#### Jitter Buffer Max

This value corresponds to "JB maximum" in [RFC3611](#) in the VoIP Metrics Report Block. The parameter does not require any conversion.

#### Jitter Buffer Abs Max

This value corresponds to "JB abs max" in [RFC3611](#) in the VoIP Metrics Report Block. The parameter does not require any conversion.

### **[4.6.2.7. Packet Loss Parameters](#)**

#### Network Packet Loss Rate

This value corresponds to "loss rate" in [RFC3611](#) in the VoIP Metrics Report Block. For conversion, see "General mapping percentages from 8 bit, fixed point numbers".

#### Jitter Buffer Discard Rate

This value corresponds to "discard rate" in [RFC3611](#) in the VoIP Metrics Report Block. For conversion, see "General mapping percentages from 8 bit, fixed point numbers".

### **[4.6.2.8. Burst/Gap Parameters](#)**

#### Burst Loss Density

This value corresponds to "burst density" in [RFC3611](#) in the VoIP Metrics Report Block. For conversion, see "General mapping percentages from 8 bit, fixed point numbers".

#### Burst Duration

This value corresponds to "burst duration" in [RFC3611](#) in the VoIP Metrics Report Block. This value requires no conversion; the exact value sent in an RTCP XR VoIP Metrics Report Block can be included in

the SIP vq-rtcpxr parameter.

#### Gap Loss Density

Pendleton

[Page 19]

[draft-ietf-sipping-rtcp-summary](#)

7 October 2008

This value corresponds to "gap density" in [RFC3611](#) in the VoIP metrics Report Block.

#### Gap Duration

This value corresponds to "gap duration" in [RFC3611](#) in the VoIP Metrics Report Block. This value requires no conversion; the exact value sent in an RTCP XR VoIP Metrics Report Block can be reported.

#### Minimum Gap Threshold

This value corresponds to "Gmin" in [RFC3611](#) in the VoIP Metrics Report Block. This value requires no conversion; the exact value sent in an RTCP XR VoIP Metrics Report Block can be reported.

### **[4.6.2.9. Delay Parameters](#)**

#### Round Trip Delay

This value corresponds to "round trip delay" in [RFC3611](#) in the VoIP Metrics Report Block and may be measured using the method defined in [RFC3550](#). The parameter is expressed in milliseconds.

#### End System Delay

This value corresponds to "end system delay" in [RFC3611](#) in the VoIP Metrics Report Block. This parameter does not require any conversion. The parameter is expressed in milliseconds.

#### 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.

#### One Way Delay

This value SHOULD be measured using the methods defined in IETF [RFC2679](#). The parameter is expressed in milliseconds.

#### Interarrival Jitter

Inter-arrival jitter is defined in [RFC 3550](#). The parameter is expressed in milliseconds.

## Mean Absolute Jitter

It is recommended that MAJ be measured as defined in ITU-T G.1020[10]. This parameter is often referred to as MAPDV. The parameter is expressed in milliseconds.

### **4.6.2.10. Signal-related Parameters**

#### Signal Level

Pendleton

[Page 20]

[draft-ietf-sipping-rtcp-summary](#)

7 October 2008

This field corresponds to "signal level" in [RFC3611](#) in the VoIP Metrics Report Block. This field provides the voice signal relative level is defined as the ratio of the signal level to a 0 dBm0 reference, expressed in decibels. This value can be used directly without extra conversion.

#### Noise Level

This field corresponds to "noise level" in [RFC3611](#) in the VoIP Metrics Report Block. This field provide the ratio of the silent period background noise level to a 0 dBm0 reference, expressed in decibels. This value can be used directly without extra conversion.

#### Residual Echo Return Loss (RERL)

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. This value can be used directly without extra conversion.

### **4.6.2.11. Quality Scores**

#### 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 [[11](#)] and G.107 [[12](#)].

#### RLQEstAlg

This field provides a text name for the algorithm used to estimate ListeningQualityR.

#### 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. Within [RFC3611](#) a reported R factor of 127 indicates that this parameter is unavailable; in this case the ConversationalQualityR parameter MUST be omitted from the vq-rtcpxr event.

#### RCQEstAlg

This field provides a text name for the algorithm used to estimate ConversationalQualityR.

Pendleton

[Page 21]

[draft-ietf-sipping-rtcp-summary](#)

7 October 2008

#### ExternalR-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. Within [RFC3611](#) a reported R factor of 127 indicates that this parameter is unavailable; in this case the ConversationalQualityR parameter MUST be omitted from the vq-rtcpxr event.

#### ExtRInEstAlg

This field provides a text name for the algorithm used to estimate ExternalR-In.

#### 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. Algorithms for computing this value SHOULD be compliant with ITU-T Recommendation P.564 and G.107. Within [RFC3611](#) a reported R factor of 127 indicates that this parameter is unavailable; in this case the ConversationalQualityR parameter SHALL be omitted from the vq-rtcpxr event.



## ExtR0utEstAlg

This field provides a text name for the algorithm used to estimate ExternalR-Out.

Conversion of [RFC3611](#) reported MOS scores for use in reporting MOS-LQ and MOS-CQ MUST be performed by dividing the [RFC3611](#) reported value by 10 if this value is less than or equal to 50 or omitting the MOS-xQ parameter if the [RFC3611](#) reported value is 127 (which indicates unavailable).

## 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 [[11](#)].

## MOSLQEstAlg

Pendleton

[Page 22]

[draft-ietf-sipping-rtcp-summary](#)

7 October 2008

This field provides a text name for the algorithm used to estimate MOS-LQ.

## 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.

## MOSCQEstAlg

This field provides a text name for the algorithm used to estimate MOS-CQ.

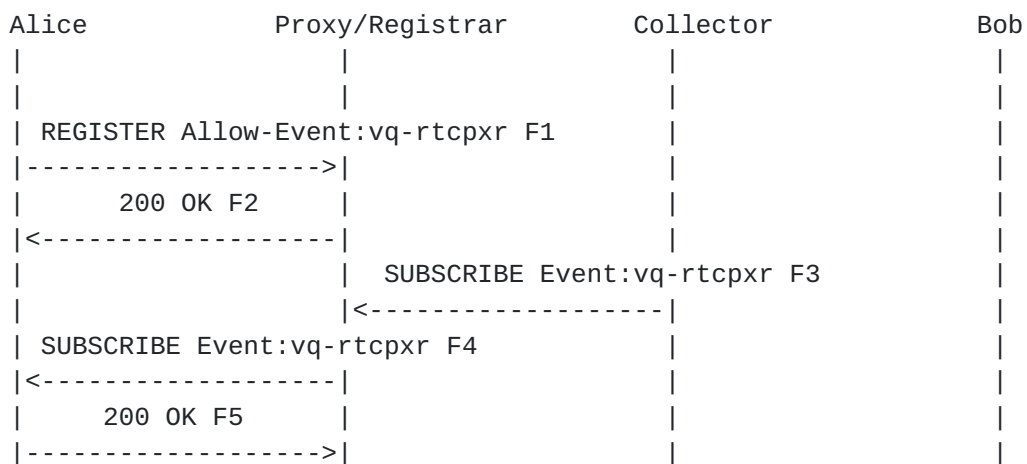
## QoEEstAlg

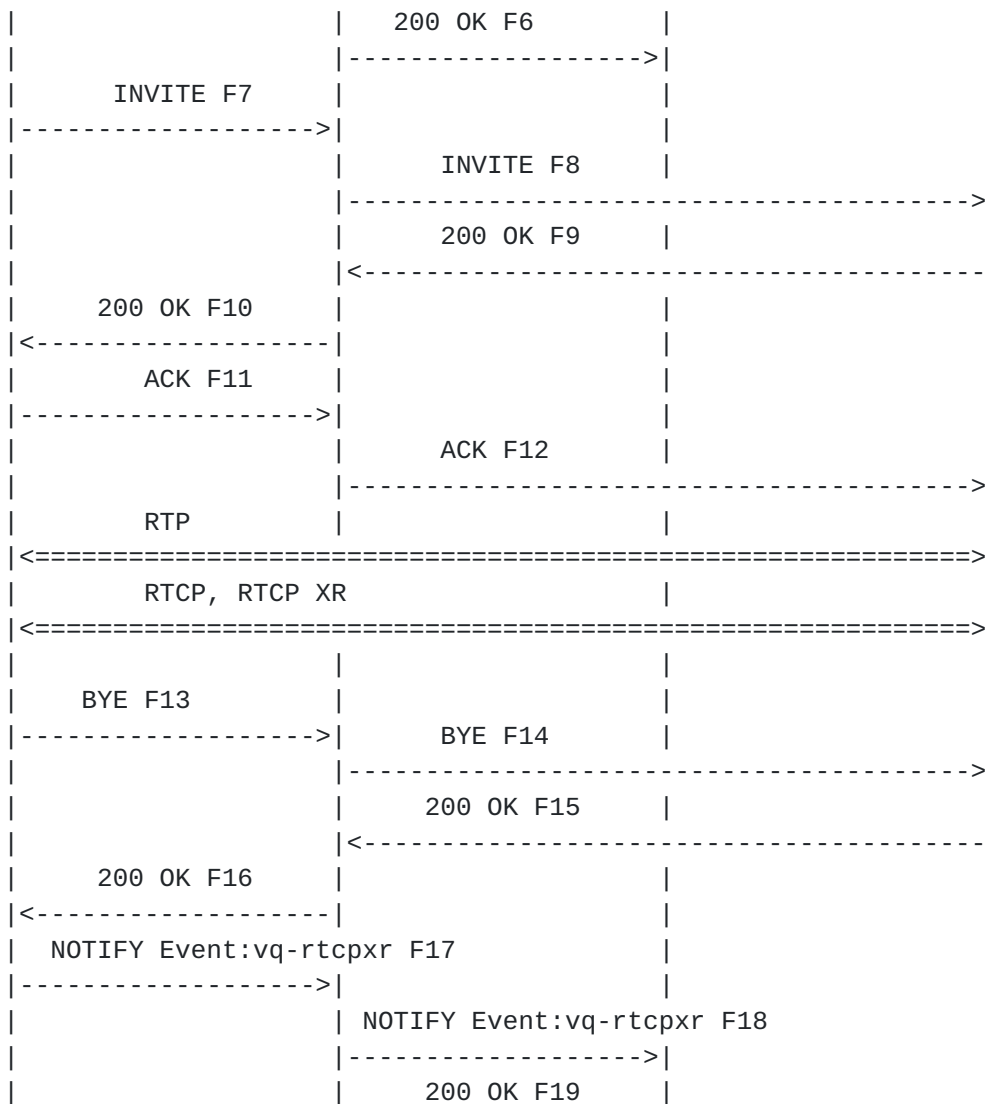
This field provides a text description of the algorithm used to estimate all voice quality metrics. This parameter is provided as an alternative to the separate estimation algorithms for use when the same algorithm is used for all measurements.

#### **4.7. Message Flow and Syntax Examples**

This section shows a number of message flow examples showing how the event package works.

##### **4.7.1. End of Session Report using NOTIFY**





Pendleton

[Page 24]

[draft-ietf-sipping-rtcp-summary](#)

7 October 2008

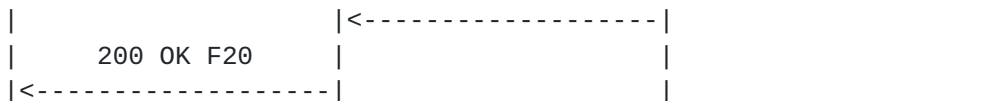


Figure 1. Summary report with NOTIFY sent after session termination.

In the call flow depicted in Figure 1, the following message format is sent in F17:

```

NOTIFY sip:collector@example.org SIP/2.0
Via: SIP/2.0/UDP pc22.example.org;branch=z9hG4bK3343d7
Max-Forwards: 70
To: <sip:collector@example.org>;tag=43524545
From: Alice <sip:alice@example.org>;tag=a3343df32
Call-ID: 1890463548@alice.example.org
  
```

CSeq: 4321 NOTIFY  
Contact: <sip:alice@pc22.example.org>  
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER,  
SUBSCRIBE, NOTIFY  
Event: vq-rtcpxr  
Accept: application/sdp, message/sipfrag  
Subscription-State: active;expires=3600  
Content-Type: application/vq-rtcpxr  
Content-Length: ...

VQSessionReport : CallTerm

LocalMetrics:

Timestamps:START=2004-10-10T18:23:43Z STOP=2004-10-01T18:26:02Z

SessionDesc:PT=0 PD=PCMU SR=8000 FD=20 FO=160 FPP=1 PPS=50

PLC=3 SSUP=on

CallID:1890463548@alice.example.org

FromID: Alice <sip:alice@example.org>

ToID: Bill <sip:bill@elpmaxe.org>

LocalAddr:IP=10.10.1.100 PORT=5000 SSRC=1a3b5c7d

RemoteAddr:IP=11.1.1.150 PORT=5002 SSRC=0x2468abcd

JitterBuffer:JBA=3 JBR=2 JBN=40 JBM=80 JBX=120

PacketLoss:NLR=5.0 JDR=2.0

BurstGapLoss:BLD=0 BD=0 GLD=2.0 GD=500 GMIN=16

Delay:RTD=200 ESD=140 SOWD=200 IAJ=2 MAJ=10

Signal:SL=-18 NL=-50 RERL=55

QualityEst:RLQ=88 RCQ=85 EXTRI=90 MOSLQ=4.1 MOSCQ=4.0

QoEEstAlg=P.564

RemoteMetrics:

Timestamps:START=2004-10-10T18:23:43Z STOP=2004-10-01T18:26:02Z

SessionDesc:PT=0 PD=PCMU SR=8000 FD=20 FO=160 FPP=1 PPS=50

PLC=3 SSUP=on

CallID:1890463548@alice.example.org

LocalAddr:IP=11.1.1.150 PORT=5002 SSRC=0x2468abcd

RemoteAddr:IP=10.10.1.100 PORT=5000 SSRC=0x1a3b5c7d

JitterBuffer:JBA=3 JBR=2 JBN=40 JBM=80 JBX=120

PacketLoss:NLR=5.0 JDR=2.0

BurstGapLoss:BLD=0 BD=0 GLD=2.0 GD=500 GMIN=16

Delay:RTD=200 ESD=140 SOWD=200 IAJ=2 MAJ=10

Pendleton

[Page 25]

[draft-ietf-sipping-rtcp-summary](#)

7 October 2008

Signal:SL=-21 NL=-45 RERL=55

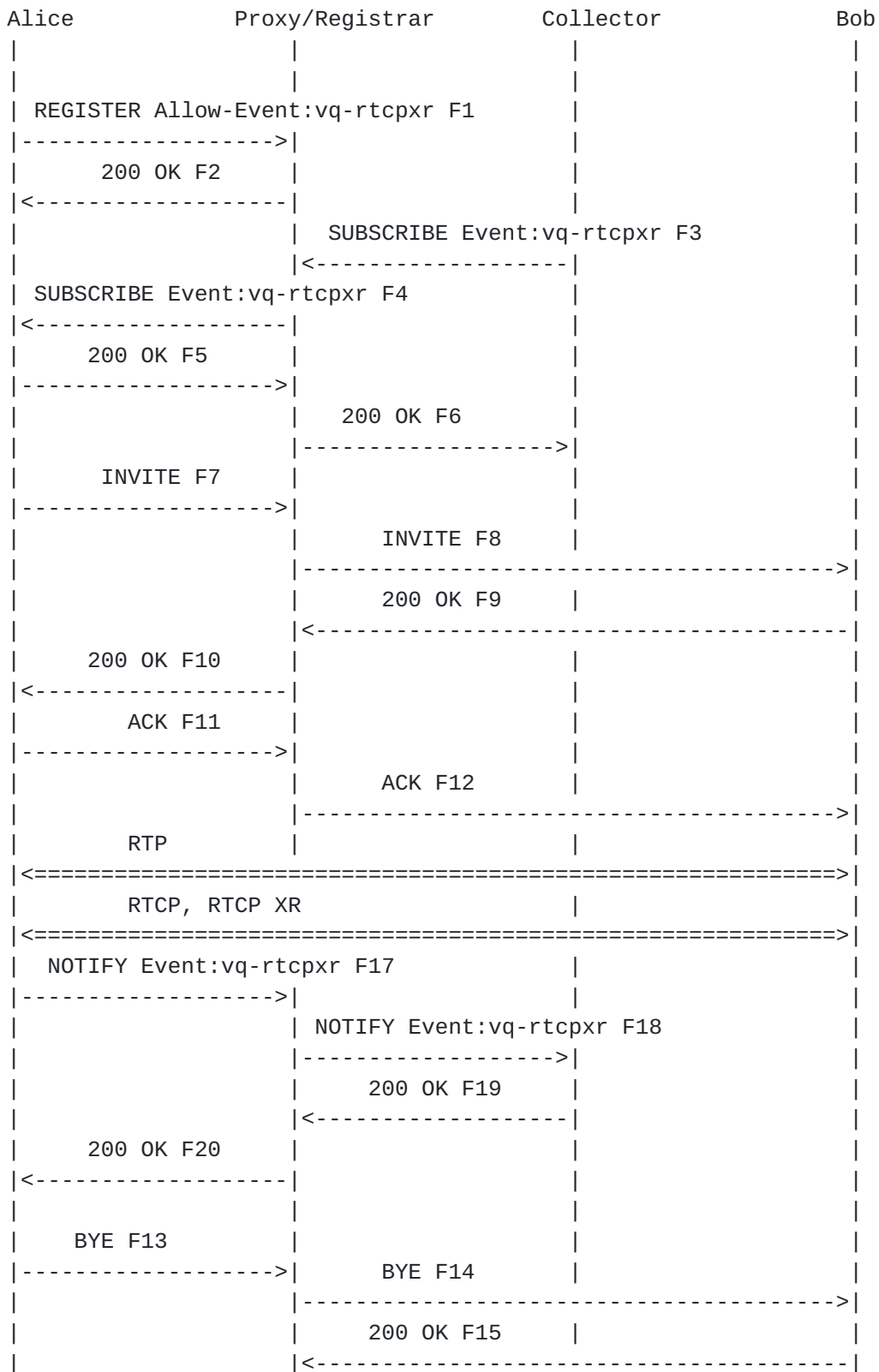
QualityEst:RLQ=90 RCQ=85 EXTRI=90 MOSLQ=4.3 MOSCQ=4.2

QoEEstAlg=P.564

DialogID:1890463548@alice.example.org;to-tag=8472761;

from-tag=9123dh311

#### **4.7.2. Mid Session Threshold Violation using NOTIFY**



200 OK F16			
------------	--	--	--

<-----		
NOTIFY Event:vq-rtcpxr F17		
----->		
	NOTIFY Event:vq-rtcpxr F18	
	----->	
	200 OK F19	
	<-----	
200 OK F20		
<-----		

Figure 2. Summary report sent during session with alert report.

In the call flow depicted in Figure 2, the following message format is sent in F17:

```

NOTIFY sip:collector@example.org SIP/2.0
Via: SIP/2.0/UDP pc22.example.org;branch=z9hG4bK3343d7
Max-Forwards: 70
To: <sip:collector@example.org>
From: Alice <sip:alice@example.org>;tag=a3343df32
Call-ID: 1890463548@alice.example.org
CSeq: 4321 PUBLISH
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER,
SUBSCRIBE, NOTIFY
Event: vq-rtcpxr
Accept: application/sdp, message/sipfrag
Content-Type: application/vq-rtcpxr
Content-Length: ...

VQAlertReport: Type=RLQ Severity=Warning Dir=local
Metrics:
Timestamps:START=2004-10-10T18:23:43Z STOP=2004-10-01T18:26:02Z
SessionDesc:PT=0 PD=PCMU SR=8000 FD=20 FO=160 FPP=1 PPS=50
          PLC=3 SSUP=on
CallID:1890463548@alice.example.org
FromID: Alice <sip:alice@example.org>
ToID: Bill <sip:bill@elpmaxe.org>
LocalAddr:IP=10.10.1.100 PORT=5000 SSRC=0x1a3b5c7d
RemoteAddr:IP=11.1.1.150 PORT=5002 SSRC=0x2468abcd
JitterBuffer:JBA=3 JBR=2 JBN=40 JBM=80 JBX=120
PacketLoss:NLR=5.0 JDR=2.0
BurstGapLoss:BLD=0 BD=0 GLD=2.0 GD=500 GMIN=16
Delay:RTD=200 ESD=140 SOWD=200 IAJ=2 MAJ=10
Signal:SL=-12 NL=-30 RERL=55
QualityEst:RLQ=60 RCQ=55 EXTR=90 MOSLQ=2.4 MOSCQ=2.3
          QoEEstAlg=P.564
RemoteMetrics:
Timestamps:START=2004-10-10T18:23:43Z STOP=2004-10-01T18:26:02Z
SessionDesc:PT=0 PD=PCMU SR=8000 FD=20 FO=160 FPP=1 PPS=50
          PLC=3 SSUP=on
CallID:1890463548@alice.example.org

```

LocalAddr:IP=11.1.1.150 PORT=5002 SSRC=0x2468abcd

Pendleton

[Page 27]

[draft-ietf-sipping-rtcp-summary](#)

7 October 2008

RemoteAddr:IP=10.10.1.100 PORT=5000 SSRC=0x1a3b5c7d  
JitterBuffer:JBA=3 JBR=2 JBN=40 JBM=80 JBX=120  
PacketLoss:NLR=5.0 JDR=2.0  
BurstGapLoss:BLD=0 BD=0 GLD=2.0 GD=500 GMIN=10  
Delay:RTD=200 ESD=140 SOWD=200 IAJ=2 MAJ=10  
Signal:SL=-21 NL=-50 RERL=55  
QualityEst:RLQ=90 RCQ=85 EXTRI=90 MOSLQ=4.2 MOSCQ=4.1  
QoEEstAlg=P.564  
DialogID:1890463548@alice.example.org;to-tag=8472761;  
from-tag=9123dh31111

#### [4.7.3.](#) End of Session Report using PUBLISH

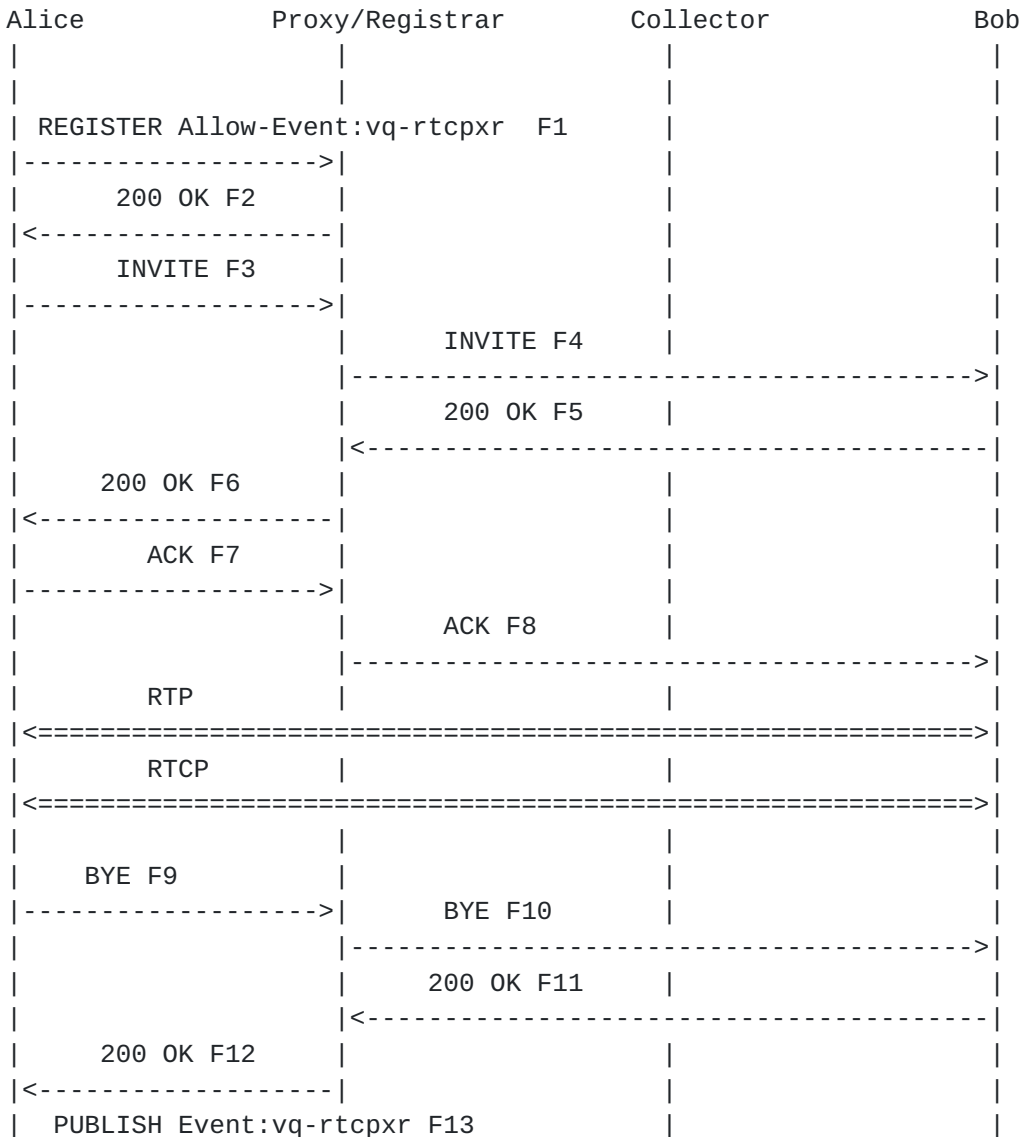




Figure 3. End of session report sent after session termination.

In the message flow depicted in Figure 3, the following message is sent in F13.

```

PUBLISH sip:collector@example.org SIP/2.0
Via: SIP/2.0/UDP pc22.example.org;branch=z9hG4bK3343d7
Max-Forwards: 70
To: <sip:proxy@example.org>
From: Alice <sip:alice@example.org>;tag=a3343df32
Call-ID: 1890463548@alice.example.org
CSeq: 4331 PUBLISH
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER,
SUBSCRIBE, NOTIFY
Event: vq-rtcpxr
Accept: application/sdp, message/sipfrag
Content-Type: application/vq-rtcpxr
Content-Length: ...
  
```

VQSessionReport : CallTerm

```

LocalMetrics:
Timestamps:START=2004-10-10T18:23:43Z STOP=2004-10-01T18:26:02Z
SessionDesc:PT=18 PD=G729 SR=8000 FD=20 FO=20 FPP=2 PPS=50
             FMTP="annexb=no" PLC=3 SSUP=on
CallID:1890463548@alice.example.org
FromID: Alice <sip:alice@example.org>
ToID: Bill <sip:bill@elpmaxe.org>
LocalAddr:IP=10.10.1.100 PORT=5000 SSRC=0x2468abcd
RemoteAddr:IP=11.1.1.150 PORT=5002 SSRC=1357efff
JitterBuffer:JBA=3 JBR=2 JBN=40 JBM=80 JBX=120
PacketLoss:NLR=5.0 JDR=2.0
BurstGapLoss:BLD=0 BD=0 GLD=2.0 GD=500 GMIN=16
Delay:RTD=200 ESD=140 SOWD=200 IAJ=2 MAJ=10
Signal:SL=-21 NL=-50 RERL=55
QualityEst:RLQ=90 RCQ=85 EXTRI=90 MOSLQ=4.2 MOSCQ=4.3
QoEEstAlg=P.564
  
```



RemoteMetrics:  
 Timestamps:START=2004-10-10T18:23:43Z STOP=2004-10-01T18:26:02Z  
 SessionDesc:PT=18 PD=G729 SR=8000 FD=20 FO=20 FPP=2 PPS=50  
               FMTP="annexb=no" PLC=3 SSUP=on  
 CallID:1890463548@alice.example.org  
 LocalAddr:IP=11.1.1.150 PORT=5002 SSRC=1357efff  
 RemoteAddr:IP=10.10.1.100 PORT=5000 SSRC=0x2468abcd  
 JitterBuffer:JBA=3 JBR=2 JBN=40 JBM=80 JBX=120  
 PacketLoss:NLR=5.0 JDR=2.0  
 BurstGapLoss:BLD=0 BD=0 GLD=2.0 GD=500 GMIN=16  
 Delay:RTD=200 ESD=140 SOWD=200 IAJ=2 MAJ=10  
 Signal:SL=-21 NL=-45 RERL=55  
 QualityEst:RLQ=90 RCQ=85 MOSLQ=4.3 MOSCQ=4.2 QoEEstAlg=P.564  
 DialogID:1890463548@alice.example.org;to-tag=8472761;  
           from-tag=9123dh311

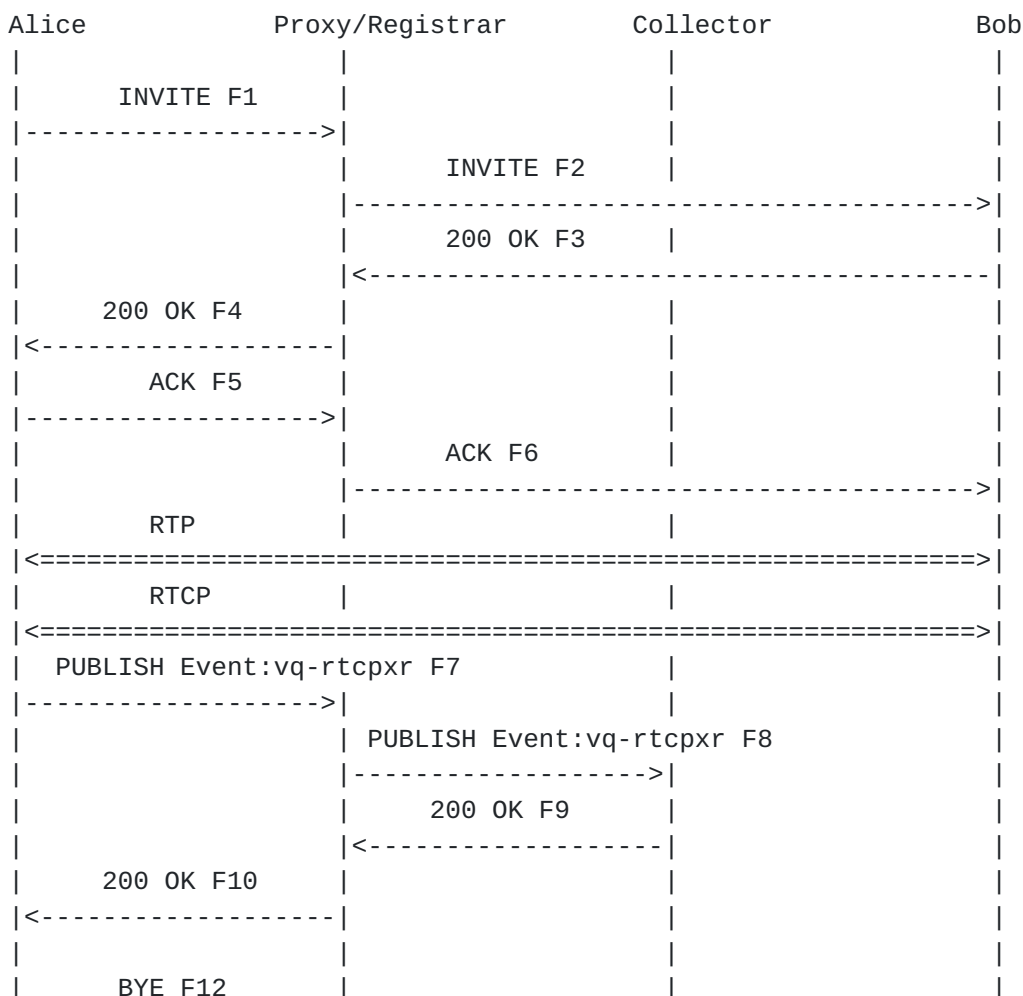
Pendleton

[Page 29]

[draft-ietf-sipping-rtcp-summary](#)

7 October 2008

#### **4.7.4. Alert Report using PUBLISH**



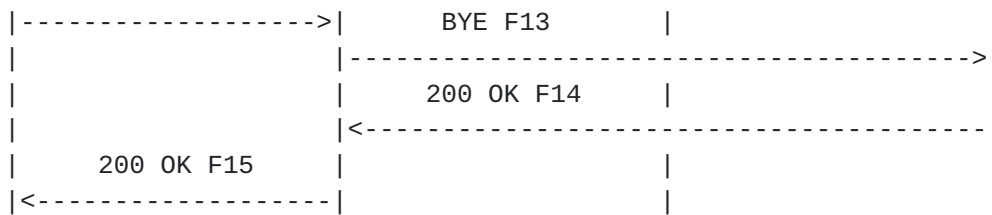


Figure 4. Alert report message flow

In the message flow depicted in Figure 4, the following message is sent in F7:

```

PUBLISH sip:collector@example.org SIP/2.0
Via: SIP/2.0/UDP pc22.example.org;branch=z9hG4bK3343d7
Max-Forwards: 70
To: <sip:collector@example.org>
From: Alice <sip:alice@example.org>;tag=a3343df32
Call-ID: 1890463548@alice.example.org
CSeq: 4321 PUBLISH
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER,
SUBSCRIBE, NOTIFY
  
```

Pendleton

[Page 30]

[draft-ietf-sipping-rtcp-summary](#)

7 October 2008

```

Event: vq-rtcpxr
Accept: application/sdp, message/sipfrag
Content-Type: application/vq-rtcpxr
Content-Length: ...
  
```

```

VQAlertReport: Type=RLQ Severity=Warning Dir=local
Metrics:
Timestamps:START=2004-10-10T18:23:43Z STOP=2004-10-01T18:26:02Z
SessionDesc:PT=0 PD=PCMU SR=8000 FD=20 FO=160 FPP=1 PPS=50
              PLC=3 SSUP=on
CallID:1890463548@alice.example.org
FromID: Alice <sip:alice@example.org>
ToID: Bill <sip:bill@elpmaxe.org>
LocalAddr:IP=10.10.1.100 PORT=5000 SSRC=2a4b6c8d
RemoteAddr:IP=11.1.1.150 PORT=5002 SSRC=9f7e5d3c
JitterBuffer:JBA=3 JBR=2 JBN=40 JBM=80 JBX=120
PacketLoss:NLR=5.0 JDR=2.0
BurstGapLoss:BLD=0 BD=0 GLD=2.0 GD=500 GMIN=16
Delay:RTD=200 ESD=140 SOWD=200 IAJ=2 MAJ=10
Signal:SL=-12 NL=-30 RERL=55
QualityEst:RLQ=60 RCQ=55 EXTR=90 MOSLQ=2.4 MOSCQ=2.3
          QoEEstAlg=P.564
  
```

RemoteMetrics:

Timestamps:START=2004-10-10T18:23:43Z STOP=2004-10-01T18:26:02Z  
 SessionDesc:PT=0 PD=PCMU SR=8000 FD=20 FO=160 FPP=1 PPS=50  
 PLC=3 SSUP=on  
 CallID:1890463548@alice.example.org  
 LocalAddr:IP=11.1.1.150 PORT=5002 SSRC=9f7e5d3c  
 RemoteAddr:IP=10.10.1.100 PORT=5000 SSRC=2a4b6c8d  
 JitterBuffer:JBA=3 JBR=2 JBN=40 JBM=80 JBX=120  
 PacketLoss:NLR=5.0 JDR=2.0  
 BurstGapLoss:BLD=0 BD=0 GLD=2.0 GD=500 GMIN=16  
 Delay:RTD=200 ESD=140 SOWD=200 IAJ=2 MAJ=10  
 Signal:SL=-23 NL=-60 RERL=55  
 QualityEst:RLQ=90 RCQ=85 EXTRI=90 MOSLQ=4.2 MOSCQ=4.3  
 QoEEstAlg=P.564  
 DialogID:1890463548@alice.example.org;to-tag=8472761;  
 from-tag=9123dh3111

#### 4.8. Configuration Dataset for vq-rtcpxr Events

It is the suggestion of the authors that the SIP configuration framework [8] be used to establish the necessary parameters for usage of vq-rtcpxr events. A dataset for this purpose is provided below:

```

<?xml version="1.0" encoding="utf-8"?>

<xs:schema targetNamespace="urn:ietf:params:xml:ns:vqrtcpxrdataset"
  xmlns:tns="urn:ietf:params:xml:ns:vqrtcpxrdataset"
  xmlns:xs="http://www.w3.org/2001/XMLSchema">

```

```

<xs:element name="rtcpxr-collector">
  <xs:complexType>
    <xs:sequence>
      <xs:element name="address" type="xs:string"/>
      <xs:element name="port" type="xs:integer"/>
    </xs:sequence>
  </xs:complexType>
</xs:element>

  <xs:element name="threshold-parameter-list">
<xs:complexType>
  <xs:sequence>
    <xs:element name="threshold-parameter">
      <xs:sequence>
        <xs:element name="parameter-name" type="xs:string"/>
        <xs:element name="warning-level" type="xs:integer"/>
        <xs:element name="excessive-level"

```

```

                                type="xs:integer"/>
        </xs:sequence>
    </xs:element>
</xs:sequence>
</xs:complexType>
</xs:element>

<xs:element name="session-report-settings">
    <xs:complexType>
        <xs:sequence>
            <xs:element name="enable" type="xs:boolean"/>
            <xs:element name="remote-report" type="xs:boolean"/>
        </xs:sequence>
    </xs:complexType>
</xs:element>

<xs:element name="interval-report-settings">
    <xs:complexType>
        <xs:sequence>
            <xs:element name="enable" type="xs:boolean"/>
            <xs:element name="remote-report" type="xs:boolean"/>
            <xs:element name="timer" type="xs:integer"/>
        </xs:sequence>
    </xs:complexType>
</xs:element>

</xs:schema>

```

## 5. IANA Considerations

This document registers a new SIP Event Package and a new MIME type.

### 5.1. SIP Event Package Registration

Pendleton

[Page 32]

[draft-ietf-sipping-rtcp-summary](#)

7 October 2008

Package name: vq-rtcp  
 Type: package  
 Contact: Amy Pendleton <asp@nortel.com>  
 Published Specification: This document

### 5.2. application/vq-rtcp-xr MIME Registration

MIME media type name: application  
 MIME subtype name: vq-rtcpxr  
 Mandatory parameters: none  
 Optional parameters: none

Encoding considerations: text  
Security considerations: See next section.  
Interoperability considerations: none.  
Published specification: This document.

Applications which use this media type: This document type is being used in notifications of VoIP quality reports.

Additional Information:

Magic Number: None  
File Extension: None  
Macintosh file type code: "TEXT"

Personal and email address for further information: Amy Pendleton  
<aspen@nortel.com>

Intended usage: COMMON

Author/Change controller: The IETF.

## **6. Security Considerations**

RTCP reports can contain sensitive information since they can provide information about the nature and duration of a session established between two or more endpoints. As a result, any third party wishing to obtain this information SHOULD be properly authenticated by the SIP UA using standard SIP mechanisms and according to the recommendations in [5]. Additionally the event content MAY be encrypted to ensure confidentiality; the mechanisms for providing confidentiality are detailed in [2].

## **7. Contributors**

The authors would like to thank Rajesh Kumar, Dave Oran, Tom Redman, Shane Holthaus and Jack Ford for their comments and input.

## **8. Normative References**

[1] Bradner, S., "Key words for use in RFCs to Indicate Requirement Levels", [BCP 14](#), [RFC 2119](#), March 1997.

Pendleton

[Page 33]

[draft-ietf-sipping-rtcp-summary](#)

7 October 2008

[2] Rosenberg, J., Schulzrinne, H., Camarillo, G., Johnston, A., Peterson, J., Sparks, R., Handley, M., and E. Schooler, "SIP: Session Initiation Protocol", [RFC 3261](#), June 2002.

[3] Schulzrinne, H., Casner, S., Frederick, R., and V. Jacobson,

"RTP: A Transport Protocol for Real-Time Applications", STD 64,  
[RFC 3550](#), July 2003.

[4] Friedman, T., Caceres, R., and A. Clark, "RTP Control Protocol  
Extended Reports (RTCP XR)", [RFC 3611](#), November 2003.

[5] Huitema, C., "Real Time Control Protocol (RTCP) attribute in  
Session Description Protocol (SDP)", [RFC 3605](#), October 2003.

[5] Roach, A., "Session Initiation Protocol (SIP)-Specific Event  
Notification", [RFC 3265](#), June 2002.

[6] Crocker, D., Ed. and P. Overell, "Augmented BNF for Syntax  
Specifications: ABNF", [RFC 5234](#), January 2008.

[7] Klyne, G. and C. Newman, "Date and Time on the Internet:  
Timestamps", [RFC 3339](#), July 2002.

[8] Petrie, D., "A Framework for Session Initiation Protocol User  
Agent Profile Delivery", [draft-ietf-sipping-config-framework-15](#),  
(work in progress), February 2008.

[9] Rosenberg, J., J. Weinberger, C. Huitema, and R. Mahy, "STUN-  
Simple Traversal of User Datagram Protocol (UDP) Through Network  
Address Translators (NATs)", [RFC 3489](#), March 2003.

[10] ITU-T Recommendation G.1020, Performance parameter  
definitions for quality of speech and other voiceband applications  
utilising IP networks

[11] ITU-T Recommendation P.564, Conformance Testing for Voice over  
IP transmission quality assessment models

[12] ITU-T Recommendation G.107, The E-model, a computational model  
for use in transmission planning

#### Author's Addresses

Amy Pendleton  
Nortel  
2380 Performance Drive  
Richardson, TX 75081, USA  
Email: [aspen@nortel.com](mailto:aspen@nortel.com)

Alan Clark  
Telchemy Incorporated  
2905 Premiere Parkway, Suite 280  
Duluth, GA 30097, USA  
Email: alan@telchemy.com

Alan Johnston  
Avaya  
St. Louis, MO 63124, USA  
Email: alan@sipstation.com

Henry Sinnreich  
Adobe Systems, Inc.  
601 Townsend Street, San Francisco, CA 94103, USA  
Email: henrys@adobe.com

## Intellectual Property Statement

The IETF takes no position regarding the validity or scope of any Intellectual Property Rights or other rights that might be claimed to pertain to the implementation or use of the technology described in this document or the extent to which any license under such rights might or might not be available; nor does it represent that it has made any independent effort to identify any such rights. Information on the procedures with respect to rights in RFC documents can be found in [BCP 78](#) and [BCP 79](#).

Copies of IPR disclosures made to the IETF Secretariat and any assurances of licenses to be made available, or the result of an attempt made to obtain a general license or permission for the use of such proprietary rights by implementers or users of this specification can be obtained from the IETF on-line IPR repository at <http://www.ietf.org/ipr>.

The IETF invites any interested party to bring to its attention any copyrights, patents or patent applications, or other proprietary rights that may cover technology that may be required to implement this standard. Please address the information to the IETF at [ietf-ipr@ietf.org](mailto:ietf-ipr@ietf.org).

## Disclaimer of Validity

This document and the information contained herein are provided on an "AS IS" basis and THE CONTRIBUTOR, THE ORGANIZATION HE/SHE REPRESENTS OR IS SPONSORED BY (IF ANY), THE INTERNET SOCIETY, THE IETF TRUST AND THE INTERNET ENGINEERING TASK FORCE DISCLAIM ALL WARRANTIES, EXPRESS OR IMPLIED, INCLUDING BUT NOT LIMITED TO ANY WARRANTY THAT THE USE OF THE INFORMATION HEREIN WILL NOT INFRINGE ANY RIGHTS OR ANY IMPLIED WARRANTIES OF MERCHANTABILITY OR FITNESS FOR A PARTICULAR PURPOSE.

## Copyright Statement

Pendleton

[Page 35]

[draft-ietf-sipping-rtcp-summary](#)

7 October 2008

Copyright (C) The IETF Trust (2008).

This document is subject to the rights, licenses and restrictions contained in [BCP 78](#), and except as set forth therein, the authors retain all their rights.

## Acknowledgment

Funding for the RFC Editor function is currently provided by the Internet Society.



