

SIPPING Working Group  
Internet-Draft  
Expires: November 1, 2007

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May 2007

Session Initiation Protocol Package for Voice Quality Reporting Event  
draft-ietf-sipping-rtcp-summary-02

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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.

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## [1.](#) Introduction

This document defines a new SIP event package, `vq-rtcpvr`, and a new MIME type, `application/vq-rtcpvr`, 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 measurement. However, the event package is designed to be extensible for use with any RTP application. The negotiation of such extensions is not defined in this recommendation.

The event package supports reporting both the local and remote versions of these statistics. It is expected that providing all views of voice quality will help facilitate multiple provider scenarios and faster problem resolution. Note that in multi-party calls, multiple reports need to be generated: either one per endpoint or one per media session.

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 can be used either with the SUBSCRIBE/NOTIFY methods or the PUBLISH method. Message flow examples for both mechanisms are provided in this document.

## [2.](#) Terminology

In this document, the key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "NOT

RECOMMENDED", "MAY", and "OPTIONAL" are to be interpreted as described in [BCP 14](#), [RFC 2119](#) [1] and indicate requirement levels for compliant implementations.

### [3.](#) SIP Events Approach

This document defines a new SIP events package [5]. A SIP UA can send these events using either the PUBLISH or SUBSCRIBE/NOTIFY methods to an entity which can make the information available to other applications. For purposes of illustration, the entities involved in SIP vq-rtcpvr event reporting will be referred to as follows:

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

The REPORTER shall be configured with one or more COLLECTOR's. The configuration process is out of scope for this recommendation, but it is suggested that the SIP configuration framework [8] be used for this purpose. A dataset should be defined for vq-rtcpvr following the suggestions in [section 5.8](#). The REPORTER shall not send any vq-rtcpvr events where a COLLECTOR has not been configured.

#### [3.1](#) PUBLISH Usage

A SIP UA that supports this specification may send the service quality metric reports using the PUBLISH method. The primary intention of using PUBLISH for this event is reduction of transaction processing.

The use of PUBLISH by this event is unique in that it does not require a soft or hard state to be maintained by either the REPORTER or the COLLECTOR. Furthermore the information that is provided in the vq-rtcpvr event is not expected to have an expiration, rather, the information is associated with the timestamps in the event itself.

The REPORTER shall populate the Request-URI of the PUBLISH method with the address of the resource (AOR) of the COLLECTOR. To ensure security of SIP proxies and the COLLECTOR, the REPORTER must be configured with the AOR 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.

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### [3.2](#) PUBLISH Overload Avoidance

A concern over the usage of the PUBLISH method is the potential overloading of servers receiving the events, particularly in the threshold reporting model. There are many approaches to solving this type of problem, but clearly the REPORTER's needs to adhere to some guidelines to reduce the probability of causing this overload condition. Some suggested solutions are:

- a) limit sending of one threshold report per metric per session
- b) limit sending of one threshold report per session regardless of the metric
- c) limit sending a new threshold report to when a metric state has been sustained for a reasonable amount of time, e.g.20-30 seconds.

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

### [3.3](#). SUBSCRIBE/NOTIFY Usage

The REPORTER may send the voice quality metric reports using the NOTIFY method. In this case, the COLLECTOR will send a SUBSCRIBE to the REPORTER to explicitly establish the relationship and the configuration of the AOR of the COLLECTOR is not needed, but may still be optionally supported.

The REPORTER shall populate the Request-URI of the PUBLISH method with the address of the resource (AOR) 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](#) [2]. The event-package token name for this package is:

"vq-rtcpxr"

#### [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

There are three notify bodies: a session report, an interval session report, and a alert report.

The session report is used for end of session reporting. This can be generated when a voice media session terminates or when a media change occurs, such as a codec change or a session forks. This report is intended to allow cumulative metric reporting. The session reports will populate the metrics with values that are measured over the interval explicitly defined by the "start" and "stop" timestamps.

The interval report is used for periodic or interval reporting. This report is intended to capture short duration metric reporting. Interval reports will populate the metrics with values that are measured over the interval explicitly defined by the "start" and "stop" timestamps.

The threshold report is used when voice quality degrades during a session. The session report parameters are also included in the alert report to provide all of the necessary diagnostic information. Like the interval report, the metrics in the threshold reports will be populated 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, along 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" CRLF
                LocalMetrics [CRLF RemoteMetrics]
                [DialogID]
```

```
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 "OtherDirMetrics:" CRLF Metrics]
[DialogID]
```

```
Metrics = TimeStamps CRLF
[SessionDescription CRLF]
CallID 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 or billing
; 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]
[FmtOptions 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
; It is recommended that IANA registered values are used
; where possible.
PayloadType = "PT" EQUAL (1*3DIGIT)

; PayloadDesc provides a text description of the codec
; It is recommended that IANA registered names are used
; where possible.
PayloadDesc = "PD" EQUAL word

; SampleRate provides the rate at which voice was sampled
; in the case of narrowband codecs, the value will typically be 8000
SampleRate = "SR" EQUAL (1*5DIGIT)

; FrameDuration can be combined with the FramesPerPacket to determine
; the packetization rate; the units for this are milliseconds.
FrameDuration = "FD" EQUAL (1*3DIGIT)

; FrameOctets provides the number of octets in each frame
; 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 number of packets, including one or
; more frames within each, that are transmitted per second
PacketsPerSecond = "PPS" EQUAL (1*5DIGIT)

; FMTP options from SDP. Note that the parameter is delimited
; 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. For more details,
; please refer to RFC 3611, RTCP XR

```

```

; 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

; LocalAddr provides the IP address, port and ssrc for the
; session from the perspective of the endpoint/UA which is
; sending the report
LocalAddr = "LocalAddr" COLON IPAddress WSP Port WSP Ssrc

; RemoteAddr provides the IP address, port and ssrc for the
; session from the perspective of the peer of the endpoint/UA
; that is sending the report
RemoteAddr = "RemoteAddr" COLON IPAddress WSP Port WSP Ssrc

IPAddress = "IP" EQUAL IPv6address / IPv4address
Port = "PORT" EQUAL 1*DIGIT
Ssrc = "SSRC" EQUAL 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 RFC 3611.
; 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 RTCP XR, RFC 3611
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 RTCP XR, RFC 3611
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 RTCP XR, RFC 3611
BurstGapLoss = "BurstGapLoss" COLON
  [BurstLossDensity WSP]
  [BurstDuration WSP]
  [GapLossDensity WSP]
  [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]
```

[InterarrivalJitter WSP]  
[MeanAbsoluteJitter]  
\*(WSP Extension)

---

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; RoundTripDelay is recommended to be measured as defined in  
; RTCP, [RFC 3550](#).

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

; EndSystemDelay metric is defined in RTCP XR, [RFC 3611](#)

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

; OneWayDelay is recommended to be measured according to  
; recommendations provided by the IPPM working group but may be  
; based on alternative measurement recommendations

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

; Interarrival Jitter is recommended to be measured as defined  
; in RTCP, [RFC 3550](#), but may be based on alternatives

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

; Mean Absolute Jitter is recommended to be measured as defined  
; by ITU-T G.1020 where it is known as MAPDV

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

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

Signal = "Signal" COLON

    [SignalLevel WSP]

    [NoiseLevel WSP]

    [ResidualEchoReturnLoss]

    \*(WSP Extension)

; SignalLevel will normally be a positive value  
; the absence of the negative sign indicates a positive value  
; where the signal level is negative, the sign must be included

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

; NoiseLevel will normally be negative but to align with the  
; the encoding of SignalLevel, the sign must be explicitly included  
; again, the absence of a sign indicates a positive value

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).  
ResidualEchoReturnLoss = "RERL" EQUAL (1\*3DIGIT)

---

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; Voice Quality estimation metrics  
; The definition of these metrics are provided in RTCP XR and  
; the new High Resolution proposal, RTCP HD.

; 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]  
[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 ; "PESQ", "G.107", or other

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

RCQEstAlg = "RCQEstAlg" EQUAL word ; "PESQ", "G.107", or other

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

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

```

ExtRInEstAlg = "ExtRIEstAlg" EQUAL word ; "PESQ" or other

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

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

ExtROutEstAlg = "ExtROEstAlg" EQUAL word ; "PESQ" or other

MOS-LQ = "MOSLQ" EQUAL (DIGIT ["." 1*2DIGIT]) ; 0.0 - 4.9

MOSLQEstAlg = "MOSLQEstAlg" EQUAL word ; "PESQ" or other

MOS-CQ = "MOSCQ" EQUAL (DIGIT ["." 1*2DIGIT]) ; 0.0 - 4.9

MOSCQEstAlg = "MOSCQEstAlg" EQUAL word ; "PESQ" or other

```

```

; alternative to the separate estimation algorithms
; for use when the same algorithm is used for all measurements
QoEEstAlg = "QoEEstAlg" EQUAL word; "PESQ" 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-Param *(SEMI did-param)
did-param = 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" / "IAD" /
    "RERL" / Extension

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

```

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

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

; miscellaneous needs for ABNF  
; some of these are pulled out of [RFC2234](#) or [RFC3261](#)  
; where this is the case, it is noted.

; taken from [RFC2234](#)

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

; taken 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

; DATE-TIME format

; taken from [RFC3339](#), refer for more information

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 for the syntax
;

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.

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.

#### [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](#) [x]. 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 "rtptime" attribute field "payload type". IANA registered types should be used.

##### Payload Desc

This parameter is not mapped from any specific SDP or RTP field; provides a text description of the Payload Type/codec.

##### Sample Rate

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

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 number of RTP packets that are transmitted per second.

#### FMTp

This parameter is taken directly from the SDP attribute "fmtp".

#### 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.

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#### [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 sending the event. The IPAddress can be IPv4 or IPv6 format. The SSRC is taken from SDP, RTCP, or RTCP XR input parameters.

#### [4.6.2.5](#) RemoteAddr

This field provides the IP address, port and ssrc of the session peer from the perspective of the endpoint sending the event.

#### [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 reused by this recommendation 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 mapping calculations.

##### Jitter Buffer Nominal

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

##### 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

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

See "General mapping percentages from 8 bit, fixed point numbers".

### 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 included in the SIP vq-rtcpxr parameter.

### 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 included in the SIP vq-rtcpxr parameter.

#### [4.2.6.10](#) Delay Parameters

##### Round Trip Delay

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

##### 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.

##### One Way Delay

This value may be measured based on IETF IPPM recommendations or may be calculated as described in [RFC3611](#):

$$\text{one way delay} = ( \text{RTD} + \text{ESD}(\text{A}) + \text{ESD}(\text{B}) ) / 2$$

The parameter is expected to be expressed in milliseconds.

#### Interarrival Jitter

It is recommended that IAJ be measured as defined in RTCP, [RFC 3550](#). The parameter is expected to be expressed in milliseconds.

#### Mean Absolute Jitter

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

### [4.2.6.11](#) Signal-related Parameters

#### Signal Level

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.

#### [4.2.6.13](#) Quality Scores

##### ListeningQualityR

This field does not have a direct mapping from [RFC3611](#) but is expected to be provided in the RTCP High Resolution (RTCP HR) draft [x]. The parameter reflects voice quality measured only from the listening related parameters i.e. does not include RERL, delay, signal level, or noise level. The scale used will typically be ITU-T G.107 compliant (0-100) but can be greater where wideband audio codecs are used.

##### RLQEstAlg

This field provides a text description of 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 including all metrics per ITU-T G.107 definition (but may be extended by vendor specific metrics as well.) The scale used will typically be ITU-T G.107 compliant (0-100) but can be greater where wideband audio codecs are used. Although in most cases the value does not need to be converted, a value of 127 indicates that this parameter is unavailable and should not be included in the vq-rtcpxr event.

##### RCQEstAlg

This field provides a text description of the algorithm used to estimate ConversationalQualityR

##### 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 scale used will typically be ITU-T G.107 compliant (0-100) but can be greater where

wideband audio codecs are used. Although in most cases the value does not need to be converted, a value of 127 indicates that this parameter is unavailable and should not be included in the vq-rtcpxr event.

#### ExtRInEstAlg

This field provides a text description of 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 scale used will typically be ITU-T G.107 compliant (0-100) but can be greater where wideband audio codecs are used. Although in most cases the value does not need to be converted, a value of 127 indicates that this parameter is unavailable and should not be included in the vq-rtcpxr event.

#### ExtROutEstAlg

This field provides a text description of the algorithm used to estimate ExternalR-Out.

#### 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. The [RFC3611](#) format is expressed as an integer in the range 10 to 50, corresponding to MOS x 10. Therefore the value should be divided by 10 to convert to vq-rtcpxr format. Additionally, a value of 127 in [RFC3611](#) indicates that the parameter is unavailable and should not be included in the vq-rtcpxr event.



#### MOSLQEstAlg

This field provides a text description of 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. The [RFC3611](#) format is expressed as an integer in the range 10 to 50, corresponding to MOS x 10. Therefore the value should be divided by 10 to convert to vq-rtcpvr format. Additionally, a value of 127 in [RFC3611](#) indicates that the parameter is unavailable and should not be included in the vq-rtcpvr event.

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#### MOSCQEstAlg

This field provides a text description of 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.

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#### [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 PUBLISH

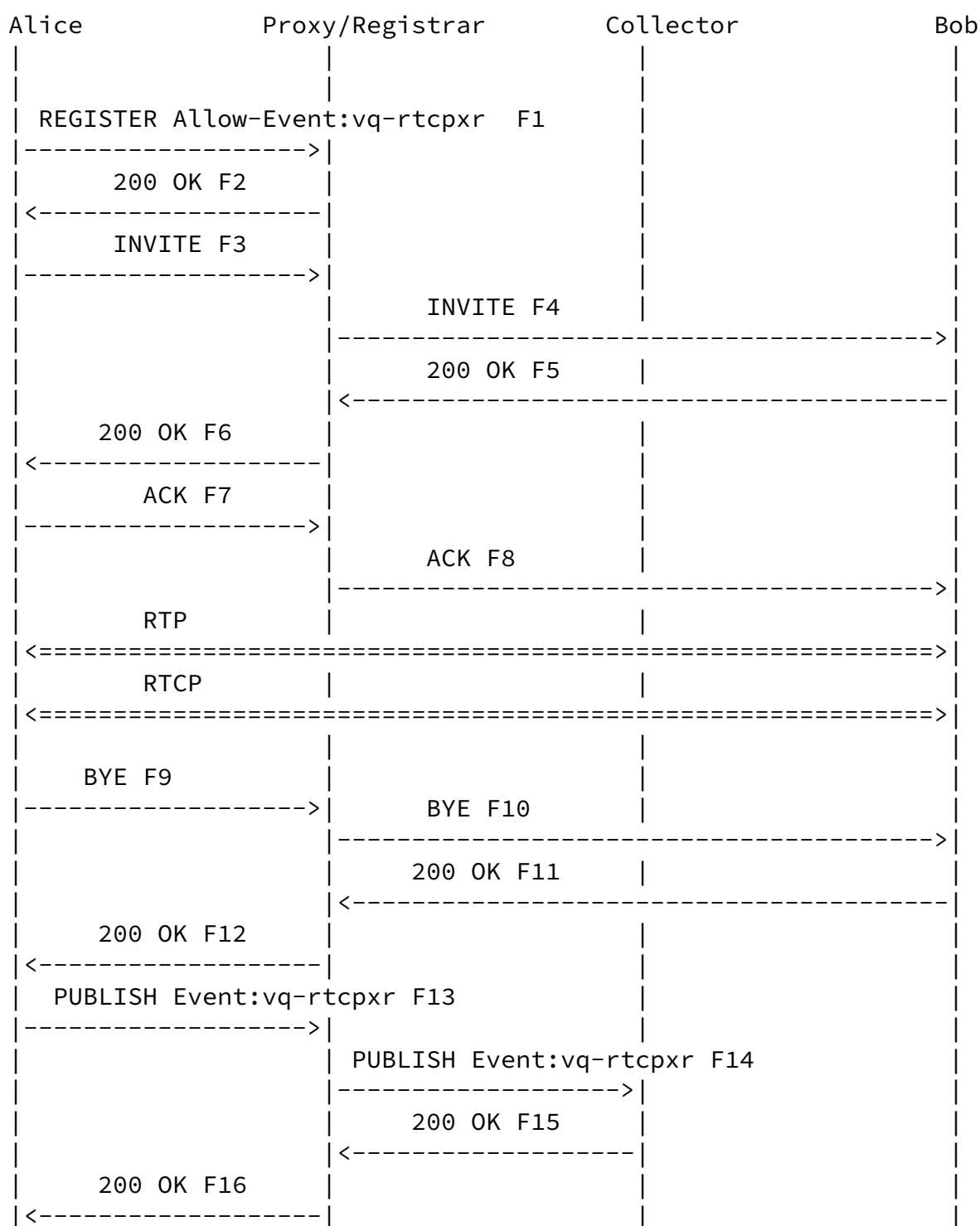


Figure 1. End of session report sent after session termination.

In the message flow depicted in Figure 1, 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

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

LocalAddr:IP=10.10.1.100 PORT=5000 SSRC=2468abcd

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 OWD=100 IAJ=2 MAJ=10

Signal:SL=2 NL=-10 RERL=55

QualityEst:RLQ=90 RCQ=85 EXTRI=90 MOSLQ=3.4 MOSCQ=3.3 QoEEstAlg=AlgX

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=2468abcd

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 OWD=100 IAJ=2 MAJ=10

Signal:SL=2 NL=-10 RERL=55

QualityEst:RLQ=90 RCQ=85 MOSLQ=3.4 MOSCQ=3.3 QoEEstAlg=AlgX

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

from-tag=9123dh311

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#### [4.7.2.](#) Alert Report using PUBLISH

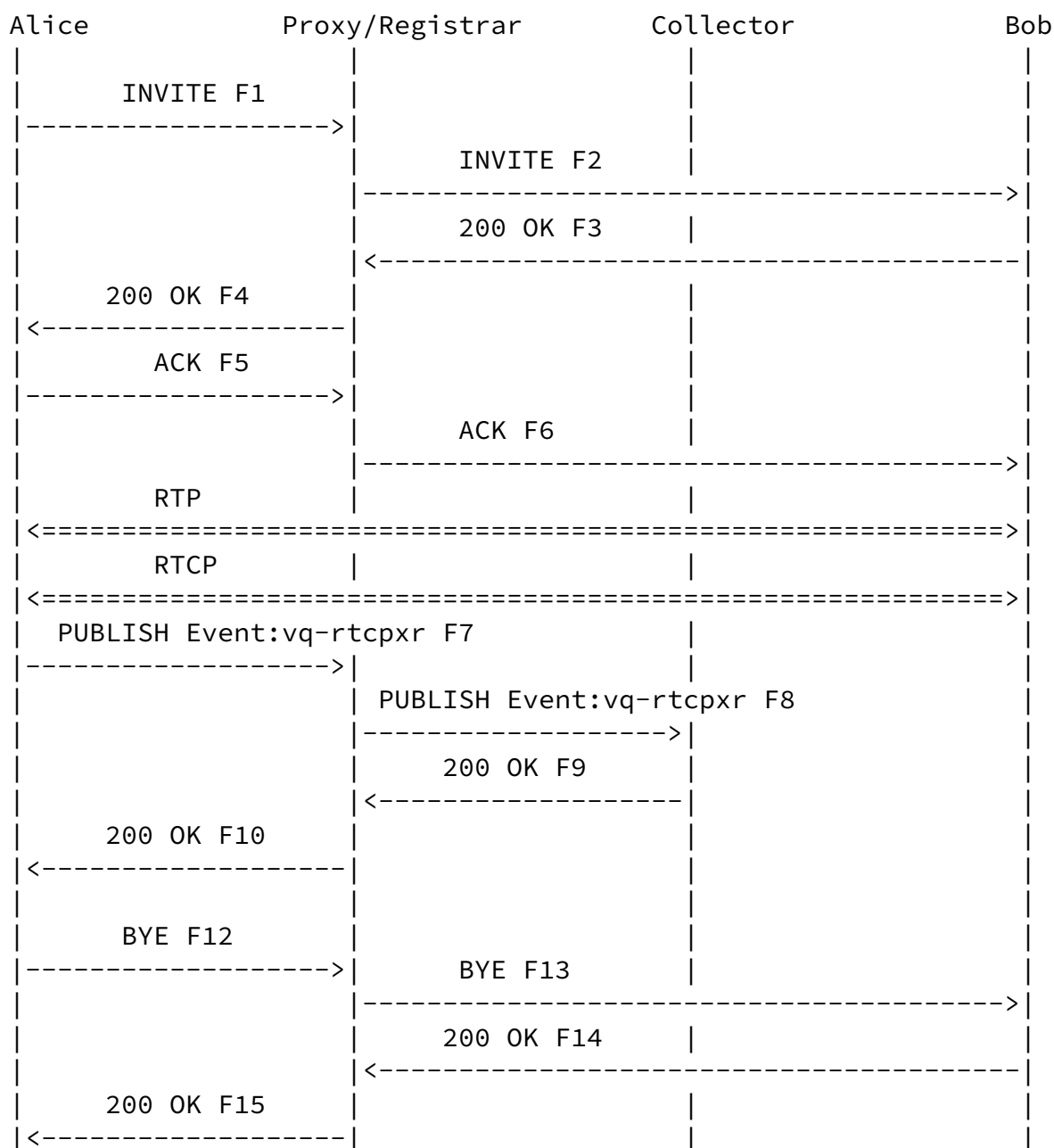


Figure 2. Alert report message flow

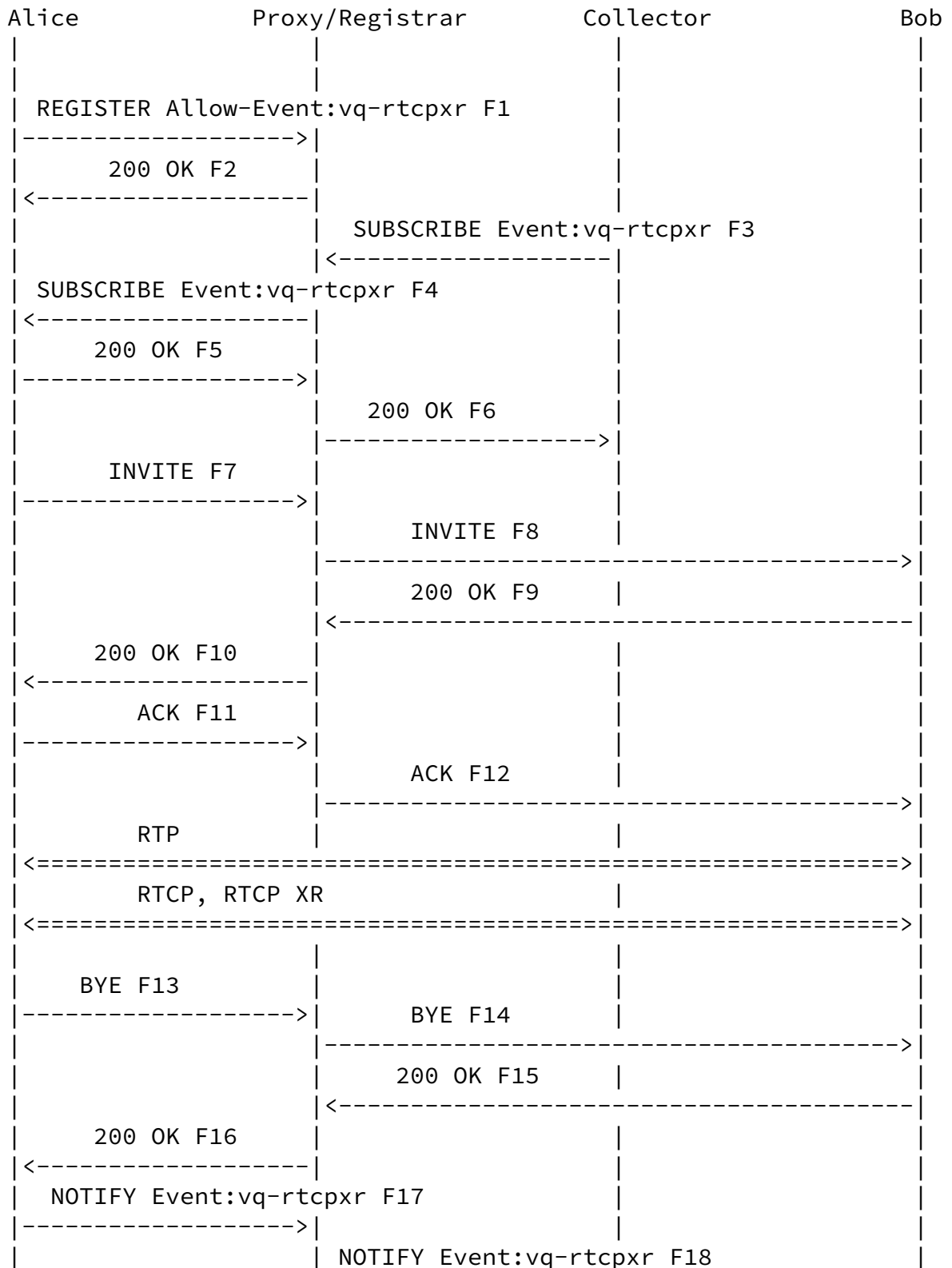
In the message flow depicted in Figure 2, the following message is sent in F7:

```
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
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
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 OWD=100 IAJ=2 MAJ=10
Signal:SL=2 NL=-10 RERL=55
QualityEst:RLQ=90 RCQ=85 EXTR=90 MOSLQ=3.4 MOSCQ=3.3 QoEEstAlg=AlgX
OtherDirMetrics:
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 OWD=100 IAJ=2 MAJ=10
Signal:SL=2 NL=-10 RERL=55
QualityEst:RLQ=90 RCQ=85 EXTRI=90 MOSLQ=3.4 MOSCQ=3.3 QoEEstAlg=AlgX
DialogID:1890463548@alice.example.org;to-tag=8472761;
        from-tag=9123dh3111
```

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



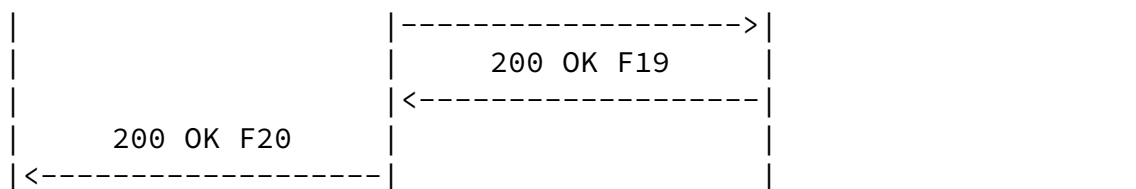


Figure 3. Summary report with NOTIFY sent after session termination.

In the call flow depicted in Figure 3, 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-rtcpvr
Accept: application/sdp, message/sipfrag
Subscription-State: active;expires=3600
Content-Type: application/vq-rtcpvr
Content-Length: ...
  
```

VQSessionReport

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

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

RemoteAddr:IP=11.1.1.150 PORT=5002 SSRC=2468abcd

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

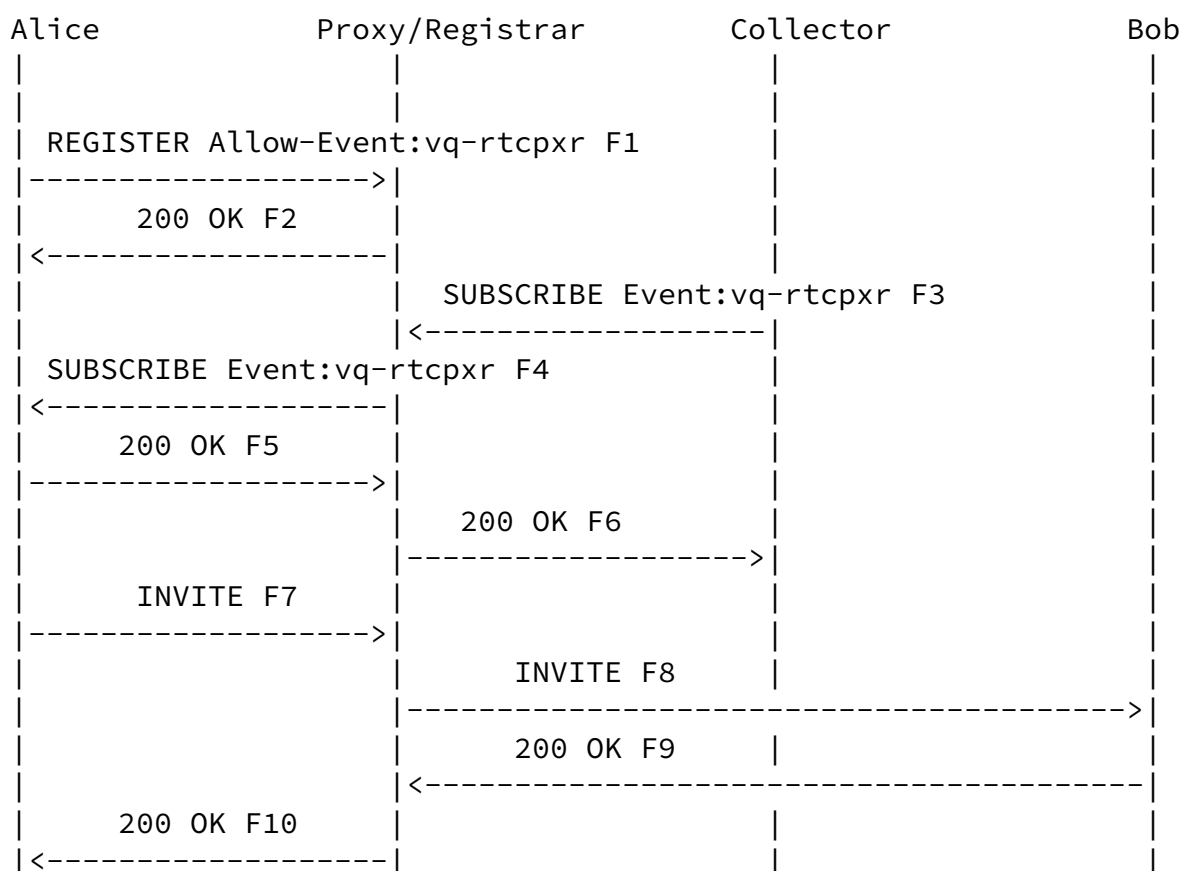
Delay:RTD=200 ESD=140 OWD=100 IAJ=2 MAJ=10

Signal:SL=2 NL=-10 RERL=55

QualityEst:RLQ=90 RCQ=85 EXTRI=90 MOSLQ=3.4 MOSCQ=3.3 QoEEstAlg=AlgX

RemoteMetrics:  
 Timestamps:START=2004-10-10T18:23:43Z STOP=2004-10-01T18:26:02Z  
 SessionDesc:PT=0 PD=PCMU SR=8000 FD=20 F0=160 FPP=1 PPS=50  
                   PLC=3 SSUP=on  
 CallID:1890463548@alice.example.org  
 LocalAddr:IP=11.1.1.150 PORT=5002 SSRC=2468abcd  
 RemoteAddr:IP=10.10.1.100 PORT=5000 SSRC=1a3b5c7d  
 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 OWD=100 IAJ=2 MAJ=10  
 Signal:SL=2 NL=-10 RERL=55  
 QualityEst:RLQ=90 RCQ=85 EXTRI=90 MOSLQ=3.4 MOSCQ=3.3 QoEEstAlg=AlgX  
 DialogID:1890463548@alice.example.org;to-tag=8472761;  
           from-tag=9123dh311

#### [4.7.4.](#) Mid Session Threshold Violation using NOTIFY





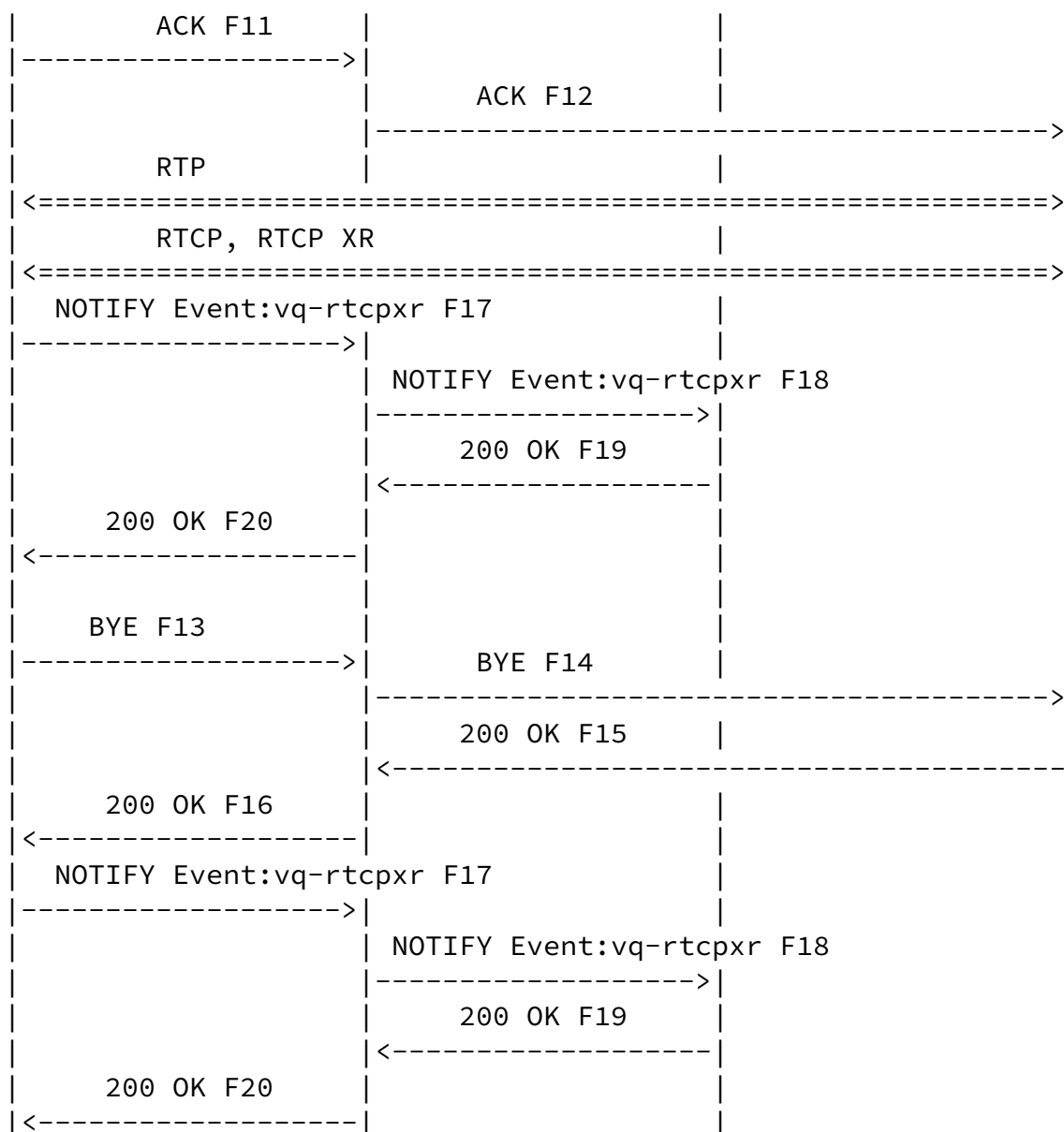


Figure 4. Summary report sent during session with threshold report.

In the call flow depicted in Figure 4, 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-rtcpvr  
Accept: application/sdp, message/sipfrag  
Content-Type: application/vq-rtcpvr  
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

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

RemoteAddr:IP=11.1.1.150 PORT=5002 SSRC=2468abcd

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 OWD=100 IAJ=2 MAJ=10

Signal:SL=2 NL=-10 RERL=55

QualityEst:RLQ=90 RCQ=85 EXTR=90 MOSLQ=3.4 MOSCQ=3.3 QoEEstAlg=AlgX

OtherDirMetrics:

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=2468abcd

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

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 OWD=100 IAJ=2 MAJ=10

Signal:SL=2 NL=-10 RERL=55

QualityEst:RLQ=90 RCQ=85 EXTRI=90 MOSLQ=3.4 MOSCQ=3.3 QoEEstAlg=AlgX

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

from-tag=9123dh31111

#### [4.8](#) Configuration Dataset for vq-rtcpvr 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-rtcpvr 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:schema>
  <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>

```

---

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## [5.](#) IANA Considerations

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

### [5.1.](#) SIP Event Package Registration

Package name: vq-rtcpxr

Type: package

Contact: Amy Pendleton <aspen@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.

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## 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 and Tom Redman for their discussions.

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