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# SIP Service Quality Reporting Event draft-ietf-sipping-rtcp-summary-00

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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 RTP-based VoIP sessions.

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## 1. Introduction

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]. 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 defined herein is designed to be extensible. The event package explicitly provides parameters for reporting both the local and remote versions of these statistics. Note that in multi-party calls, multiple reports will need to be generated either one for each endpoint or one for each session. Configuration of usage of the event package is not covered in this document. The event package can be used either with the SUBSCRIBE/NOTIFY methods or the PUBLISH method. Message flow examples for both mechanims are provided in this document.

#### 2. SIP Events Approach

This document defines a new SIP events package [6]. The intended methods to use for this event are PUBLISH and SUBSCRIBE/NOTIFY. A SIP UA can provide voice quality reports using either of these methods to an entity which can make the information available to other applications.

### 3. Use of PUBLISH Method

A SIP UA that supports this specification may send the service quality metric reports using the PUBLISH method. An application wishing to access this performance data maintains a State Agent for the vq-rtcpxr event package. The Request-URI of the PUBLISH method is set to the address of the resource for the VoIP application. The PUBLISH message is sent to the normal default outbound proxy server of the SIP UA or could be sent to other interested parties.

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 Event Publication Agent (EPA) or the Event State Compositor (ESC). Futhermore the information that is presented by the vq-rtcpxr event in a PUBLISH request is not expected to have an expiration, rather, the information is associated with the timestamps in the event itself. The primary intention of using PUBLISH for this event is reduction of transaction processing.

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A concern over the usage of the PUBLISH method is 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 EPA'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 such as 20-30 seconds.

- 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 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

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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

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 format used in SIP [2], and contains references to elements defined therein. 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. 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.

VQReportEvent = AlertReport / SessionReport / IntervalReport SessionReport = "VQSessionReport" CLRF LocalMetrics [CLRF RemoteMetrics] [DialogID] IntervalReport = "VQIntervalReport" CLRF LocalMetrics [CLRF RemoteMetrics] [DialogID] LocalMetrics = "LocalMetrics" COLON Metrics RemoteMetrics = "RemoteMetrics" COLON Metrics AlertReport = "VQAlertReport" COLON

```
Internet-Draft SIP Event for Voice Quality Reporting December 2005
                   MetricType WSP Severity WSP Direction CLRF
                   "Metrics:" CLRF
                   Metrics
                   [CLRF "OtherDir Metrics:" CLRF Metrics]
                   [DialogID]
   Metrics = TimeStamps CLRF
           [SessionDescription CLRF]
           CallID CLRF
          LocalAddr CRLF
          RemoteAddr CRLF
           [JitterBuffer CRLF]
           [PacketLoss CRLF]
           [BurstGapLoss CLRF]
           [Delay CLRF]
           [Signal CLRF]
           [QualityEstimates CLRF]
           *[Extension CLRF]
   ; Timestamps are measured inaccordance with RFC 3611
   TimeStamps = "Timestamps" COLON StartTime WSP StopTime
   StartTime = "START" EQUAL word
   StopTime = "STOP" EQUAL word
   ; 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]
           [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
   ; It is recommended that IANA registered values are used
   ; where possible.
   PayloadType = "PT" EQUAL (1*3DIGIT)
   ; PayloadDesc provides a text description of the codec
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                                                                [Page 6]
```

Internet-Draft SIP Event for Voice Quality Reporting December 2005 ; 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 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) ; FMTP options from SDP. Note that the parameter is deliniated ; by " " to avoid parsing issues in transitioning between SDP and ; SIP parsing FmtpOptions = "FMTP" EQUAL "word-plus" ; 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 = "CID" EQUAL 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 Pendleton, et al. Expires June 11, 2006 [Page 7]

```
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  ; 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" EQUAL text / notAvail
  Ssrc
  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
                      = "JBM" EQUAL (1*5DIGIT) ;0-65535
  JitterBufferMax
  JitterBufferAbsMax = "JBA" 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*3(DIGIT) ["." 1*2(DIGIT)]) ;percentage
  JitterBufferDiscardRate
   = "JDR" EQUAL (1*3(DIGIT) ["." 1*2(DIGIT)]) ;percentage
  ; BurstGapLoss metric definitions are provided in RTCP XR, RFC 3611
  BurstGapLoss = "BurstGapLoss" COLON
          [BurstLossDensity WSP]
          [BurstDuration WSP]
```

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```
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           [GapLossDensity WSP]
           [GapDuration WSP]
           [MinimumGapThreshold]
           *[WSP Extension]
   BurstLossDensity
    = "BLD" EQUAL (1*3(DIGIT) ["." 1*2(DIGIT)]) ;percentage
   BurstDuration
    = "BD" EQUAL (1*8DIGIT) ;0-3,600,000 -- milliseconds
   GapLossDensity
   = "GLD" EQUAL (1*3(DIGIT) ["." 1*2(DIGIT)]) ;percentage
   GapDuration
   = "GD" EQUAL (1*8DIGIT) ;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]
   ; 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
                     = "OWD" EQUAL (1*5DIGIT) ;0-65535
   OneWayDelay
   ; 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)
   ; Signal metrics definitions are provided in RTCP XR, RFC 3611
   Signal = "Signal" COLON
           [SignalLevel WSP]
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                                                                [Page 9]
```

```
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           [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)
   ResidualEchoReturnLoss = "RERL" EQUAL (1*3DIGIT)
   ; Voice Quality estimation metrics
   ; The definition of these metrics are provided in RTCP XR and
   ; the new High Resolution proposal, RTCP HD.
   ; Each quality estmiate 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
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                                                               [Page 10]
```

```
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   ;
      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", "G.107", 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
  ExtRInEstAlg = "ExtROEstAlg" EQUAL word ; "PESQ", "G.107", or other
  MOS-LQ = "MOSLQ" EQUAL (DIGIT ["." 1*2DIGIT]) ; 0.0 - 4.9
  MOSLQEstAlg = "MOSLQEstAlg" EQUAL word ; "PESQ", "G.107", or other
  MOS-CQ = "MOSCQ" EQUAL (DIGIT ["." 1*2DIGIT] ; 0.0 - 4.9
  MOSCQEstAlg = "MOSCQEstAlg" EQUAL word ; "PESQ", "G.107", or other
  ; alternative to the separate estimation algorithms
   ; for use when the same algorithm is used for all measurements
  QoEEstAlg = "QoEEstAlg'
  DialogID = "DialogID" COLON callid *(SEMI did-parm)
  did-parm = to-tag / from-tag / generic-param
  callid = Call-ID-Parm
   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 ]
```

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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
----	----	----	-----	----	---------	--------	-------	---------

Alice	Prox	y/Registrar	Collector	Bob
     REGISTE	R Allow-Even	   t:vq-rtcpxr F1		
20	0 OK F2			
<   IN 	VITE F3			
		INVITE F4	İ	
		200 OK F5		>
200	OK F6	<   		
	ACK F7			
		ACK F8		
	RTP			
<======	RTCP			>
<=======     BYE	F9	======================================		

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I CHAICCON,	CC UL.	DISPIECD 00	$x_{11} \subset x_{12}$	2000	LIUGC	ㅗㅗ

----->| BYE F10 \_\_\_\_\_ --> 200 OK F11 <-----200 OK F12 <-----PUBLISH Event:vq-rtcpxr F13 -----> PUBLISH Event:vq-rtcpxr F14 -----> 200 OK F15 <-----200 OK F16 <-----

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.com SIP/2.0 Via: SIP/2.0/UDP pc22.example.com;branch=z9hG4bK3343d7 Max-Forwards: 70 To: <sip:proxy@example.com> From: Alice <sip:alice@example.com>;tag=a3343df32 Call-ID: 1890463548@alice.chicago.com 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=10012004.18.23.43 STOP=10012004.18.26.02
SessionDesc:PT=18 PD=G729 SR=8000 FD=20 FPP=2 FMTP="annexb=no" PLC=3
               SSUP=on
CallID:1890463548@alice.uac.chicago.com
LocalAddr: IP=10.10.1.100 PORT=5000 SSRC=fjuekdn393k
RemoteAddr: IP=11.1.1.150 PORT=5002 SSRC=r3k3k99weid
JitterBuffer: JBA=3 JBR=2 JBN=40 JBM=80 JBA=120
PacketLoss:NLR=5.0 JDR=2.0
BurstGapLoss:BLD=0 BD=0 GLD=2.0 GD=500
Delay:RTD=200 ESD=140 IAD=2
```

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```
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   Signal:SL=2 NL=-10 RERL=14
  QualityEst:RLQ=90 RCQ=85 EXTRI=90 MOSLQ=3.4 MOSCQ=3.3 QoEEstAlg=G.107
  RemoteMetrics:
  TimeStamps:START=10012004.18.23.43 STOP=10012004.18.26.02
  SessionDesc:PT=18 PD=G729 SR=8000 FD=20 FPP=2 FMTP="annexb=no" PLC=3
                  SSUP=on
  CallID:1890463548@alice.uac.chicago.com
  LocalAddr: IP=11.1.1.150 PORT=5002 SSRC=r3k3k99weid
  RemoteAddr: IP=10.10.1.100 PORT=5000 SSRC=fjuekdn393k
  JitterBuffer:JBA=3 JBR=2 JBN=40 JBM=80 JBA=120
  PacketLoss:NLR=5.0 JDR=2.0
  BurstGapLoss:BLD=0 BD=0 GLD=2.0 GD=500
  Delay:RTD=200 ESD=140 IAJ=2
  Signal:SL=2 NL=-10 RERL=0
  QualityEst:RLQ=90 RCQ=85 MOSLQ=3.4 MOSCQ=3.3 QoEEstAlg=G.107
  DialogID:1890463548@alice.uac.chicago.com;to-tag=8472761;
                  from-tag=9123dh311
```





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<-----

Figure 2. Alert report message flow

200 OK F15 <-----|

This alert indicates that the quality of the call in progress, as measured by the Listening Quality R metric, has degraded to an unacceptable level. For futher troubleshooting of the problem, all metrics are populated, including the remote values obtained via RTCP XR (RFC 3611) in the endpoints.

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

PUBLISH sip:collector@chicago.example.com SIP/2.0 Via: SIP/2.0/UDP pc22.example.com;branch=z9hG4bK3343d7 Max-Forwards: 70 To: <sip:collector@example.com> From: Alice <sip:alice@example.com>;tag=a3343df32 Call-ID: 1890463548@alice.chicago.com 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=10012004.19.01.04 STOP=10012004.19.01.52
SessionDesc:PT=0 PD=PCMU SR=8000 FD=20 FPP=2 PLC=3 SSUP=on
CallID:1890463548@alice.chicago.com
LocalAddr: IP=10.10.1.100 PORT=5000 SSRC=fjuekdn393k
RemoteAddr: IP=11.1.1.150 PORT=5002 SSRC=r3k3k99weid
JitterBuffer: JBA=3 JBR=2 JBN=40 JBM=80 JBA=120
PacketLoss:NLR=5.0 JDR=2.0
BurstGapLoss:BLD=0 BD=0 GLD=2.0 GD=500
Delay:RTD=200 ESD=140 IAJ=2
Signal:SL=2 NL=-10 RERL=14
QualityEst:RLQ=90 RCQ=85 EXTR=90 MOSLQ=3.4 MOSCQ=3.3 QoEEstAlg=G.107
OtherDir Metrics:
```

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TimeStamps:START=10012004.18.23.43 STOP=10012004.18.26.02 SessionDesc:PT=0 PD=PCMU SR=8000 FD=20 FPP=2 PLC=3 SSUP=on CallID:1890463548@alice.uac.chicago.com LocalAddr: IP=11.1.1.150 PORT=5002 SSRC=r3k3k99weid RemoteAddr: IP=10.10.1.100 PORT=5000 SSRC=fjuekdn393k JitterBuffer: JBA=3 JBR=2 JBN=40 JBM=80 JBA=120 PacketLoss:NLR=5.0 JDR=2.0 BurstGapLoss:BLD=0 BD=0 GLD=2.0 GD=500 Delay:RTD=200 ESD=140 IAJ=2 Signal:SL=2 NL=-10 RERL=0 QualityEst:RLQ=90 RCQ=85 EXTRI=90 MOSLQ=3.4 MOSCQ=3.3 QoEEstAlg=G.107 DialogID:1890463548@alice.chicago.com;to-tag=8472761; from-tag=9123dh3111

Alice Proxy/Registrar Collector Bob REGISTER Allow-Event:vq-rtcpxr F1 -----> 200 OK F2 <-----SUBSCRIBE Event:vq-rtcpxr F3 <-----SUBSCRIBE Event:vq-rtcpxr F4 <-----200 OK F5 ----> 200 OK F6 -----> INVITE F7 ----> INVITE F8 -----> 200 OK F9 <-----200 OK F10 <-----ACK F11 ---->| ACK F12 RTP RTCP, RTCP XR 

4.7.3. End of Session Report using NOTIFY

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BYE F13 BYE F14 ---->| \_\_\_\_\_ -----> 200 OK F15 |<-----200 OK F16 /\_\_\_\_\_ NOTIFY Event:vq-rtcpxr F17 -----> NOTIFY Event:vq-rtcpxr F18 -----> 200 OK F19 <-----200 OK F20 <-----

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.com SIP/2.0 Via: SIP/2.0/UDP pc22.example.com;branch=z9hG4bK3343d7 Max-Forwards: 70 To: <sip:collector@example.com>;tag=43524545 From: Alice <sip:alice@example.com>;tag=a3343df32 Call-ID: 1890463548@alice.chicago.com CSeq: 4321 NOTIFY Contact: <sip:alice@pc22.example.com> 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 LocalMetrics: TimeStamps:START=10012004.18.23.43 STOP=10012004.18.26.02 SessionDesc:PT=0 PD=PCMU SR=8000 FD=20 FPP=2 PLC=3 SSUP=on CallID:1890463548@alice.uac.chicago.com LocalAddr:IP=10.10.1.100 PORT=5000 SSRC=fjuekdn393k RemoteAddr: IP=11.1.1.150 PORT=5002 SSRC=r3k3k99weid JitterBuffer: JBA=3 JBR=2 JBN=40 JBM=80 JBA=120 PacketLoss:NLR=5.0 JDR=2.0 BurstGapLoss:BLD=0 BD=0 GLD=2.0 GD=500

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```
Internet-Draft SIP Event for Voice Quality Reporting December 2005
  Delay:RTD=200 ESD=140 IAD=2
  Signal:SL=2 NL=-10 RERL=14
  QualityEst:RLQ=90 RCQ=85 EXTRI=90 MOSLQ=3.4 MOSCQ=3.3 QoEEstAlg=G.107
  RemoteMetrics:
  TimeStamps:START=10012004.18.23.43 STOP=10012004.18.26.02
  SessionDesc:PT=0 PD=PCMU SR=8000 FD=20 FPP=2 PLC=3 SSUP=on
  CallID:1890463548@alice.uac.chicago.com
  LocalAddr: IP=11.1.1.150 PORT=5002 SSRC=r3k3k99weid
  RemoteAddr: IP=10.10.1.100 PORT=5000 SSRC=fjuekdn393k
  JitterBuffer:JBA=3 JBR=2 JBN=40 JBM=80 JBA=120
  PacketLoss:NLR=5.0 JDR=2.0
  BurstGapLoss:BLD=0 BD=0 GLD=2.0 GD=500
  Delay:RTD=200 ESD=140 IAJ=2
  Signal:SL=2 NL=-10 RERL=0
  QualityEst:RLQ=90 RCQ=85 EXTRI=90 MOSLQ=3.4 MOSCQ=3.3 QoEEstAlg=G.107
  DialogID:1890463548@alice.chicago.com;to-tag=8472761;
           from-tag=9123dh311
```

4.7.4. Mid Session Threshold Violation using NOTIFY

Alice	Proxy	/Registrar	Collector	Bob
REGISTER A     200 O	  >  K F2	∶vq-rtcpxr Fl		
<			nt	
		SUBSCRIBE EVE	enc.vq-rtcpxr FS	
SUBSCRIBE	Event:vq-r	tcpxr F4		
<		1		
200 OK	F5			İ
	>			
		200 OK F6		
   INVIT	E F7   >		>	
		INVITE F8	3	
		 200 ok F9	·	>
		<	 	
200 OK	F10			
< ACK	 F11   >			
		ACK F12		

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RTP	 	>
RTCP, RTCP XI	======================================	
NOTIFY Event:vq-rt	======================================	
>	NOTIFY Event:vq-rtcr	pxr F18
	200 OK F19	
200 OK F20	<	
<>   BYE F13  >	BYE F14	
	200 OK F15	>
200 OK F16	<	
NOTIFY Event:vq-rt	 cpxr F17	
>	NOTIFY Event:vq-rtcr	oxr F18
	200 OK F19	
200 OK F20	<   	

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

This alert indicates that the quality of the call in progress, based on Listening Quality transmission rating R , has degraded to an unacceptable level. For futher troubleshooting of the problem, all metrics are populated, including the remote values obtained via RTCP XR [4].

In the call flow depicted in Figure 4, the following message format is sent in F17:

NOTIFY sip:collector@chicago.example.com SIP/2.0
Via: SIP/2.0/UDP pc22.example.com;branch=z9hG4bK3343d7
Max-Forwards: 70
To: <sip:collector@example.com>
From: Alice <sip:alice@example.com>;tag=a3343df32
Call-ID: 1890463548@alice.chicago.com
CSeq: 4321 PUBLISH

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Internet-Draft SIP Event for Voice Quality Reporting December 2005 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=10012004.19.01.04 STOP=10012004.19.01.52 SessionDesc:PT=0 PD=PCMU SR=8000 FD=20 FPP=2 PLC=3 SSUP=on CallID:1890463548@alice.chicago.com LocalAddr: IP=10.10.1.100 PORT=5000 SSRC=fjuekdn393k RemoteAddr: IP=11.1.1.150 PORT=5002 SSRC=r3k3k99weid JitterBuffer: JBA=3 JBR=2 JBN=40 JBM=80 JBA=120 PacketLoss:NLR=5.0 JDR=2.0 BurstGapLoss:BLD=0 BD=0 GLD=2.0 GD=500 Delay:RTD=200 ESD=140 IAJ=2 Signal:SL=2 NL=-10 RERL=14 QualityEst:RLQ=90 RCQ=85 EXTR=90 MOSLQ=3.4 MOSCQ=3.3 QoEEstAlg=G.107 OtherDir Metrics: TimeStamps:START=10012004.18.23.43 STOP=10012004.18.26.02 SessionDesc:PT=0 PD=PCMU SR=8000 FD=20 FPP=2 PLC=3 SSUP=on CallID:1890463548@alice.uac.chicago.com LocalAddr: IP=11.1.1.150 PORT=5002 SSRC=r3k3k99weid RemoteAddr: IP=10.10.1.100 PORT=5000 SSRC=fjuekdn393k JitterBuffer: JBA=3 JBR=2 JBN=40 JBM=80 JBA=120 PacketLoss:NLR=5.0 JDR=2.0 BurstGapLoss:BLD=0 BD=0 GLD=2.0 GD=500 Delay:RTD=200 ESD=140 IAJ=2 Signal:SL=2 NL=-10 RERL=0 QualityEst:RLQ=90 RCQ=85 EXTRI=90 MOSLQ=3.4 MOSCQ=3.3 QoEEstAlg=G.107 DialogID:1890463548@alice.chicago.com;to-tag=8472761; from-tag=9123dh31111

## 4.8. IANA Considerations

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

## 4.8.1. SIP Event Package Registration

Package name: vq-rtcpxr Type: package Contact: Alan Johnston <alan.johnston@mci.com>

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Published Specification: This document

4.8.2. 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: Alan Johnston <alan.johnston@mci.com>

Intended usage: COMMON

Author/Change controller: The IETF.

4.9. Security Considerations

RTCP reports can contain sensitive information since they can provide information about the nature and duration of a session established between two endpoints. As a result, any third party wishing to obtain this information should be properly authenticated and the information transferred securely.

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4.10. Updates since -08

- Change title, minor syntax cleanup.

4.11. Updates since -07

- Significant syntax change -- made to align with typical SIP syntax. Parameter lines now separate tag from value with colon and individual parameters separate tag from value with equals.

- Clean up of examples based on comments.

4.12. Updates since -06

- Changed package name from "svcqual" to "vq-rtcpxr" as the focus will be only voice quality reporting based on RTCP XR.

- Rewrite of verbage in document to clarify that the focus will be voice quality reporting and not general performance or service quality reporting.

- Removed section on "Why SNMP is not appropriate" as it was inaccurate and misleading.

- Removed usage of SDP lines for describing the basic session parameters. A reduced format designed specifically for this document is provided instead.

- General clean up of syntax based on comments from WG

- Removed general description of parameters, instead referring to RTCP XR for more details in most cases.

- Reworked message flow examples using new syntax.

## 4.13. Updates since -04

- Changed package name from "perfrpt" to "svcqual"

4.14. Updates since -03

- Removed discussion of alternative mechanisms
- Changed from NOTIFY transport to PUBLISH transport.
- Changed package name from "rtcp-xr" to "perfrpt"
- Corrected call flows.

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- Minor updates to message body format.

- Added IANA registration for perfrpt SIP Event Package and application/rtcp-xr MIME registration.

- Added more discussion for motivation and reasons why SNMP is not suitable

4.15. Contributors

The authors would like to thank Rajesh Kumar, Dave Oran and Tom Redman for their discussions.

- 5. References
- 5.1. Normative References
  - [1] Bradner, S., "Key words for use in RFCs to Indicate Requirement Levels", BCP 14, RFC 2119, March 1997.
- 5.2. Informational References
  - [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.
  - [6] Roach, A., "Session Initiation Protocol (SIP)-Specific Event Notification", RFC 3265, June 2002.
  - [7] Crocker, D., Ed. and P. Overell, "Augmented BNF for Syntax Specifications: ABNF", RFC 2234, November 1997.

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