Abstract

This document discusses the motivation and requirements for the delivery of service quality reports from SIP endpoints to non-participants in the session. A publication mechanism using a new
SIP events package is proposed as a solution. An event package "svcqual" and an application/svcqual MIME type is defined in this document along with some example call flows.

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1. Introduction

This document defines a new performance report SIP event package and a new MIME type for VoIP performance metrics. In addition, the document includes the requirements, motivation, and a discussion why SNMP alone is not sufficient for this problem space. Using this mechanism, a SIP user agent (UA) can publish performance reports, or a performance quality collector can subscribe to receive performance reports from a particular SIP UA. The application/svcqual MIME type defined in this document is based on the information contained in a RTCP-XR VOIP block and is highly optimized and useful for voice session quality. As VoIP is currently the most immediate need for the real time quality reports available with this package, it has been defined first. However, the event package is designed to be extensible – existing MIME types which carry performance reports could be used, or other application/media specific MIME types could be defined and used within this framework.

2. Requirements

REQ-1: An authorized third party should be able to receive selected performance reports on a near real time basis.

REQ-2: The VoIP application should not have to store large amounts of information.

REQ-3: The VoIP application must be able to authenticate the third party.

REQ-4: The performance report information must be able to be transferred securely.

REQ-6: The reports will include or be associated with dialog identifiers for correlation purposes.

3. Motivations for the Approach

Monitor the application:

While QoS monitoring in network elements using for example SNMP is quite common, only the VoIP applications in endpoints are close enough to the experience of individual users, on a call by call basis in endpoints as diverse as desktop SIP phones, PDAs, PCs or mobile phones. For this reason, the approach taken here is to monitor the voice quality at the application level and not in network elements.

Focus on voice specific impairments:
RTCP extensions support the monitoring of various impairments such as packet loss, delay and jitter, but more important, specific metrics are included to measure burst errors that produce voice clipping. Voice clipping is the most annoying impairment in VoIP due to network congestion in the media path between VoIP applications.

4. Why SNMP is Not Appropriate

Since this type of QoS monitoring seems related to management, SNMP could possibly be used to collect this type of data as well. SNMP is however primarily used for management of network devices as they relate to the infrastructure but is not typically used for management of applications on that infrastructure. The focus of SIP is applications and the performance management for those applications cannot rely on SNMP. SNMP may be used to manage some aspects of the physical device aspects of the SIP user agent.

Specifically, SNMP may be used to manage the SIP user agent - the phone, soft phone or gateway. However, the information available in RTCP summary reports is of less interest to the management of the UA and more of interest to the VoIP service provider. In many cases, separate entities will be involved. For example, an enterprise may manage their own SIP phones using SNMP, but a service provider provides SIP and gateway services. It is unlikely a service provider will have SNMP privileges and may not be able to manage NAT/firewall traversal, etc. For these reasons, SNMP is not a good fit for this "service level" management function.

5. SIP Events Approach

In this approach, a new SIP events package is defined. The intended methods to use for this event are PUBLISH and SUBSCRIBE/NOTIFY. A VoIP application will send performance data using either of these methods to an entity which will make the information available to other applications.

6. Use of PUBLISH Method

A VoIP application which supports this specification may send performance report information using the PUBLISH method. An application wishing to access this performance data maintains a State Agent for the svcqual event package. The Request-URI of the PUBLISH method is set to the address of the resource for the VoIP application. The PUBLISH method is sent to the normal default outbound proxy server of the VoIP application.

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). Furthermore the information that is presented by the svcqual 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.

The State Agent can use normal mechanisms for publication throttling or rejection of the information as described in the PUBLISH [8] specification.

7. Event Package Formal Definition

7.1 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:

"svcqual"

7.2 Event Package Parameters

No event package parameters are defined.

7.3 SUBSCRIBE Bodies

No SUBSCRIBE bodies are described by this specification.

7.4 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.

7.5 NOTIFY Bodies

There are two notify bodies: a general report and a threshold report. The general report is used for periodic, mid-call reporting and end of call reporting. The metrics provided by this event are intended to be generically defined to allow both cumulative based and short interval based reporting.

The threshold report is used when call quality degrades. The general report is also included in the alert report to provide all of the necessary diagnostic information.

This specification defines a new MIME type application/svcqual which is a text encoding of the RTCP-XR statistics, with the addition of a
few additional identifiers.

7.6 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 [7] used in SIP [2], and contains references to elements defined therein.

General Report Event:

\[
VQEvent = VoiceQualityAlert / (LocalMetrics [CLRF RemoteMetrics]) [DialogID]
\]

\[
LocalMetrics = ("LocalMetrics") HCOLON VoiceQualityMetrics
\]

\[
RemoteMetrics = ("RemoteMetrics") HCOLON VoiceQualityMetrics
\]

\[
VoiceQualityMetrics = ("VQMetrics") CLRF
\]

\[
TimeStampInfo CLRF
SessionDesc CLRF
CallID CLRF
Ssrc CRLF
JitterBuffer CRLF
PacketLoss CRLF
BurstLoss CRLF
GapLoss CRLF
Delay CRLF
Signal CRLF
Quality
\]

\[
TimeStampInfo = "Timestamps" EQUAL start-time SP stop-time
\]

\[
start-time = "start" COLON alphanum [*alphanum]
\]

\[
stop-time = "stop" COLON alphanum [*alphanum]
\]

\[
SessionDesc = ("SessionDesc") EQUAL cline mline aline
\]

\[
cline = "cline" COLON alphanum [*alphanum]
\]

\[
mline = "mline" COLON alphanum [*alphanum]
\]

\[
aline = "aline" COLON alphanum [*alphanum] [*aline]
\]

\[
CallID = ("CallID" / "i") COLON callid
\]

\[
callid = word [ "@" word ]
\]
Ssrc    = ("Ssrc") COLON alphanum [*alphanum]

JitterBuffer = ("JitterBuffer") EQUAL jb-type SP jb-rate
                SP jb-nom SP jb-max SP jb-abs-max
jb-type    = ("type") COLON ("adapt" / "non-adapt" / "unknown"
                / "not available" )
jb-rate    = ("rate") COLON 1*3DIGIT / notAvail
jb-max     = ("max") COLON 1*3DIGIT / notAvail
jb-nom     = ("nom") COLON 1*3DIGIT / notAvail
jb-absmax  = ("abmax") COLON 1*3DIGIT / notAvail

PacketLoss = ("PktLoss") EQUAL loss-rate SP disc-rate
loss-rate  = ("loss") 1*2DIGIT / notAvail
disc-rate  = ("discard") COLON 1*2DIGIT / notAvail

BurstLoss = ("BurstLoss") EQUAL density SP length
GapLoss   = ("GapLoss") EQUAL density SP length
density   = ("density") COLON 1*4DIGIT / notAvail
length    = ("length") COLON 1*4DIGIT / notAvail

Delay = ("Delay") EQUAL roundtrip SP endsys SP jitter
roundtrip = ("roundtrip") COLON 1*4DIGIT / notAvail
endsys    = ("endsystem") COLON 1*4DIGIT / notAvail
jitter    = ("jitter") COLON 1*3DIGIT / notAvail

Signal = ("Signal") EQ signal SP echo SP noise
signal    = ("signalLevel") COLON 1*2DIGIT / notAvail
echo      = ("echoReturnLoss") COLON 1*3DIGIT / notAvail
noise     = ("noiseLevel") COLON 1*2DIGIT / notAvail

Quality = ("QualityScores") EQ r-lq SP r-cg SP r-cq-ext SP m-lq SP m-cq
r-lq      = ("r-lq") COLON 1*3DIGIT / notAvail
r-cg      = ("r-cg") COLON 1*3DIGIT / notAvail
r-cq-ext  = ("r-cq-ext") COLON 1*3DIGIT / notAvail
m-lq      = ("mos-lq") COLON DIGIT [",." 1*2DIGIT] / notAvail
m-cq      = ("mos-cq") COLON DIGIT [",." 1*2DIGIT] / notAvail

DialogID = ("DialogID") COLON callid *(SEMI did-parm)
did-parm = to-tag / from-tag / generic-param
callid  = token
to-tag  = "to-tag" EQUAL token
from-tag = "from-tag" EQUAL token

VoiceQualityAlert = ("VQAlert") COLON ViolationMetric SP
Severity SP ViolationDirection
CRLF VoiceQualityMetrics

ViolationMetric = ("Type") COLON ("r-lq" / "r-cq" / "r-cq-ext" /
ViolationDirection = ("Dir") COLON ("local" / "remote")

Severity = ("Severity") COLON ("Warning" / "Critical" / "Clear")

notAvail = "not available" / "na"

7.7 Voice Quality Metric Definitions

Timestamp Information
This line will provide the start and stop time for the metric measurement interval. In some cases, this could be the entire call and in other cases could be only a very short duration when threshold-based or interval-based reporting are supporting.

Session Description
This line should provide media session information using the <xref target="RFC3550">Session Description Protocol (SDP)</xref> for the local endpoint. The format defined only requires the endpoint to send its "c=" line, "m=" line, and all related "a=" lines.

SSRC
This line provides the SSRC parameter that is transmitted in the RTP packets which should provide unique identification of the media stream.

Jitter Buffer Information Line
The parameters in this line provide information about the jitter buffer within the media endpoint.

Jitter Buffer Type
Indicator of the jitter buffer is adaptive or static. When the jitter buffer is adaptive, then its size is being dynamically adjusted to deal with varying levels of
jitter. When non-adaptive, the jitter buffer size is maintained at a fixed level.

Jitter Buffer Adaptation Rate
This represents the implementation specific adjustment rate of a jitter buffer in adaptive mode.

Jitter Buffer Nominal Delay
This is the current nominal jitter buffer delay in milliseconds, which corresponds to the nominal jitter buffer delay for packets that arrive exactly on time. This parameter MUST be provided for both fixed and adaptive jitter buffer implementations.

Jitter Buffer Maximum Delay
This is the current maximum jitter buffer delay in milliseconds which corresponds to the earliest arriving packet that would not be discarded. In simple queue implementations this may correspond to the nominal size. In adaptive jitter buffer implementations, this value may dynamically vary up to JB abs max (see below).

Jitter Buffer Absolute Maximum Delay
This is the absolute maximum delay in milliseconds that the adaptive jitter buffer can reach under worst case conditions. If this value exceeds 65535 milliseconds, then this field SHALL convey the value 65535. This parameter MUST be provided for adaptive jitter buffer implementations and its value MUST be set to JB maximum for fixed jitter buffer implementations.

Packet Loss Information Line
The parameters in this line are general packet loss metrics.

Packet Loss Ratio
The percentage of packets lost within the network during the time period captured by the report.

Packet Discard Rate
The percentage of packets discarded due to jitter within the network during the time period captured by the report.

Burst Loss Information Line
The parameters in this provide burst loss metrics.
Burst Density
The percentage of packets lost and discarded within a burst (high loss rate) period.

Burst Length (mS)
The mean length of a burst.

Gap Loss Information Line
The parameters in this provide random loss metrics.

Gap Density
The percentage of packets lost and discarded within a gap (low loss rate) period.

Gap Length (mS)
The mean length of a gap

Delay Information Line
The parameters in this provide delay metrics.

Round Trip Delay (mS)
The round trip delay between RTP interfaces

End System Round Trip Delay (mS)
The "round trip" delay between the RTP interface and the analog or trunk interface.

Jitter
Using definition from the RTCP, interarrival jitter is defined to be the mean deviation of the difference in packet spacing at the receiver compared to the sender for a pair of packets. This metric is provided in milliseconds.

Signal Information Line
The parameters in this provide analog impairment metrics.

Signal Level
The signal level in decibals during talkspurts.

Noise Level (dBm)
The signal level in decibals during silence periods.

Residual Echo Return Loss (dB)
The residual (uncancelled) echo level from the analog or
trunk interface.

Quality Scores Information Line
This line provides various quality scores for the session.

R - Listening Quality
Estimated listening call quality expressed in a score from 0 - 100, per ITU-T standard G.107. The calculation of this value should only involve impairments measured from the listening direction.

R - Conversational Quality
Estimated conversational call quality expressed in a score from 0 - 100, per ITU-T standard G.107.

R - Conversational Quality - External
Estimated external conversational transmission quality expressed as a score from 0 - 100, per ITU-T standard G.107. This value provides the score for the segment of the session that is carried over a network segment external to the RTP segment; for example a cellular network.

MOS-LQ
Estimated listening call quality expressed as a floating point number between 0.0 and 5.0.

MOS-CQ
Estimated conversational call quality expressed as a floating point number between 0.0 and 5.0.

7.8 Call Flows and Syntax Examples

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

7.8.1 End of Session Notification using PUBLISH

<table>
<thead>
<tr>
<th>Alice</th>
<th>Proxy/Registrar</th>
<th>Collector</th>
<th>Bob</th>
</tr>
</thead>
<tbody>
<tr>
<td>REGISTER Allow-Event:svcqual F1</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>&lt;--------------------------</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>200 OK F2</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>&lt;--------------------------</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>INVITE F3</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Figure 1. Summary report sent after session termination.

In the call flow depicted in Figure 1, the following message format 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: svcqual
Accept: application/sdp, message/sipfrag
Content-Type: application/svcqual

LocalMetrics:VQMetrics:
TimeStampInfo=start:10012004.18.23.43 stop:10012004.18.26.02
SessionDesc=cline:IN IP4 10.10.1.123 mline:audio 5002 RTP/AVP 0
   aline:rtpmap:0 PCMU/8000 aline:ptime=20
CallID=1890463548@alice.uac.chicago.com
Ssrc=fjuekdn393k
JitterBuffer=type:adapt rate:2 nom:40 max:80 abmax:120
PktLoss=loss:5 discard:2
BurstLoss=density:0 length:0
GapLoss=density:2 length:0
Delay=roundtrip:200 endsystem:140 jitter:2
Signal=signalLevel:2 echoReturnLoss:14 noiseLevel:10
QualityScores=r-lq:90 r-cq:80 r-cq-ext:92
   mos-lq:3.4 mos-cq:3.3

RemoteMetrics:VQMetrics:
TimeStampInfo=start:10012004.18.23.43 stop:10012004.18.26.02
SessionDesc=cline:IN IP4 11.1.1.150 mline:audio 5002 RTP/AVP 0
   aline:rtpmap:0 PCMU/8000 aline:ptime=20
CallID=1890463548@alice.uac.chicago.com
Ssrc=r3k3k99weid
JitterBuffer=type:adapt rate:2 nom:40 max:80 abmax:120
PktLoss=loss:2 discard:1
BurstLoss=density:0 length:0
GapLoss=density:2 length:0
Delay=roundtrip:100 endsystem:140 jitter:2
Signal=signalLevel:2 echoReturnLoss:0 noiseLevel:9
QualityScores=r-lq:88 r-cq:80 r-cq-ext:92
   mos-lq:3.4 mos-cq:3.3
DialogID:38419823470834;to-tag=8472761;from-tag=9123dh311

7.8.2 End of Session Notification using NOTIFY

Alice          Proxy/Registrar          Collector          Bob

<table>
<thead>
<tr>
<th>REGISTER Allow-Event:svcqual F1</th>
</tr>
</thead>
<tbody>
<tr>
<td>&lt;------------------------</td>
</tr>
<tr>
<td>200 OK F2</td>
</tr>
<tr>
<td>----------------------------</td>
</tr>
<tr>
<td>SUBSCRIBE Event:svcqual F3</td>
</tr>
<tr>
<td>&lt;------------------------</td>
</tr>
</tbody>
</table>
In the call flow depicted in Figure 2, 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: svcqual
Accept: application/sdp, message/sipfrag
Subscription-State: active;expires=3600
Content-Type: application/svcqual
Content-Length: ...

LocalMetrics: VQMetrics:
TimeStampInfo: start:10012004.18.23.43 stop:10012004.18.26.02
SessionDesc: cline:IN IP4 11.1.1.150 mline:audio 5002 RTP/AVP 0
     aline:rtpmap:0 PCMU/8000 aline:ptime=20
CallID=1890463548@alice.chicago.com
Ssrc=fjuekd393k
JitterBuffer=type:adapt rate:2 nom:40 max:80 abmax:120
PktLoss=loss:5 discard:2
BurstLoss=density:0 length:0
GapLoss=density:2 length:0
Delay=roundtrip:200 endsystem:140 jitter:2
Signal=signalLevel:2 echoReturnLoss:14 noiseLevel:10
QualityScores=r-lq:90 r-cq:80 r-cq-ext:92
   mos-lq:3.4 mos-cq:3.3
RemoteMetrics: VQMetrics:
SessionDesc: cline:IN IP4 11.1.1.150 mline:audio 5002 RTP/AVP 0
     aline:rtpmap:0 PCMU/8000 aline:ptime=20
CallID=1890463548@alice.uac.chicago.com
Ssrc=r3k3k99weidtd
JitterBuffer=type:adapt rate:2 nom:40 max:80 abmax:120
PktLoss=loss:2 discard:1
BurstLoss=density:0 length:0
GapLoss=density:2 length:0
Delay=roundtrip:100 endsystem:140 jitter:2
Signal=signalLevel:2 echoReturnLoss:0 noiseLevel:9
QualityScores=r-lq:88 r-cq:80 r-cq-ext:92
   mos-lq:3.3 mos-cq:3.3
DialogID:38419823470834;to-tag=8472761;from-tag=9123dh311

7.8.3 Mid Session Threshold Violation using PUBLISH

Alice          Proxy/Registrar          Collector          Bob

Figure 3. Summary report sent during session with threshold report.

In the call flow depicted in Figure 3, the following message format is sent in F16:

    PUBLISH sip:collector@chicago.example.com SIP/2.0
This alert indicates that the quality of the call in progress, as calculated using the Listening Quality transmission rating R, has degraded to an unacceptable level. For further troubleshooting of the problem, all metrics are populated, including the remote values which were obtained via RTCP XR in the endpoints.
7.8.4 Mid Session Threshold Violation using NOTIFY

<table>
<thead>
<tr>
<th>Alice</th>
<th>Proxy/Registrar</th>
<th>Collector</th>
<th>Bob</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>REGISTER Allow-Event:svcqual F1</td>
<td>200 OK F2</td>
<td></td>
<td></td>
</tr>
<tr>
<td>200 OK F3</td>
<td>SUBSCRIBE Event:svcqual F4</td>
<td></td>
<td></td>
</tr>
<tr>
<td>&lt;--------------------</td>
<td>-------------------&gt;</td>
<td>200 OK F5</td>
<td></td>
</tr>
<tr>
<td>SUBSCRIBE Event:svcqual F3</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>&lt;-------------------</td>
<td>200 OK F6</td>
<td></td>
<td></td>
</tr>
<tr>
<td>INVITE F7</td>
<td>INVITE F8</td>
<td></td>
<td></td>
</tr>
<tr>
<td>200 OK F9</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>&lt;--------------------</td>
<td>200 OK F10</td>
<td></td>
<td></td>
</tr>
<tr>
<td>INVITE F8</td>
<td>ACK F11</td>
<td></td>
<td></td>
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<tr>
<td></td>
<td>200 OK F11</td>
<td></td>
<td></td>
</tr>
<tr>
<td>INVITE F8</td>
<td>ACK F12</td>
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<td></td>
</tr>
<tr>
<td>RTP</td>
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<td></td>
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<tr>
<td>RTCP</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>&lt;=================================================================&gt;</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>NOTIFY Event:svcqual F17</td>
<td>NOTIFY Event:svcqual F18</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>200 OK F19</td>
<td></td>
<td></td>
</tr>
<tr>
<td>200 OK F20</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>NOTIFY Event:svcqual F18</td>
<td>200 OK F19</td>
<td></td>
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<tr>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>BYE F13</td>
<td></td>
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<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>200 OK F15</td>
</tr>
</tbody>
</table>
Figure 4. Summary report sent during session with threshold report.

In the call flow depicted in Figure 2, 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
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER,
SUBSCRIBE, NOTIFY
Event: svcqual
Accept: application/sdp, message/sipfrag
Content-Type: application/svcqual
Content-Length: ...

VQAlert: Type:r-lq Severity:Warning Dir:local
LocalMetrics:VQMetrics:
TimeStampInfo=start:10012004.19.01.04 stop:10012004.19.01.52
SessionDesc=cline:IN IP4 11.1.1.150 mline:audio 5002 RTP/AVP 0
    aline:rtpmap:0 PCMU/8000 aline:ptime=20
CallID=1890463548@alice.chicago.com
Ssrc=fjuekdn3
JitterBuffer=type:adapt rate:2 nom:40 max:80 abmax:120
PktLoss=loss:5 discard:2
BurstLoss=density:0 length:0
GapLoss=density:2 length:0
Delay=roundtrip:200 endsystem:140
Signal=signalLevel:2 echoReturnLoss:14 noiseLevel:1
QualityScores=r-lq:82 r-cq:80 mos-lq:3.4 mos-cq:3.3
Remote:VQMetrics:
SessionDesc=cline:IN IP4 11.1.1.150 mline:audio 5002 RTP/AVP 0
    aline:rtpmap:0 PCMU/8000 aline:ptime=20
CallID=1890463548@alice.uac.chicago.com
```
This alert indicates that the quality of the call in progress, as calculated using the Listening Quality transmission rating $R$, has degraded to an unacceptable level. For further troubleshooting of the problem, all metrics are populated, including the remote values which were obtained via RTCP XR [4] in the endpoints.

7.9 IANA Considerations

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

7.9.1 SIP Event Package Registration

Package name: svcqual

Type: package

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

Published Specification: This document
7.9.2 application/rtcp-xr MIME Registration

MIME media type name: application
MIME subtype name: svcqual
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.

7.10 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.
7.11 Updates since -04
- Changed package name from "perfrpt" to "svcqual"

7.12 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.
- 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

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

8 Informative References


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