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RTCP XR - IP Video Metrics Report Blocks
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Abstract

This document defines extensions to the RTCP XR extended report packet type blocks to support video over IP (VoIP) monitoring for IPTV and video conferencing endpoint reporting.

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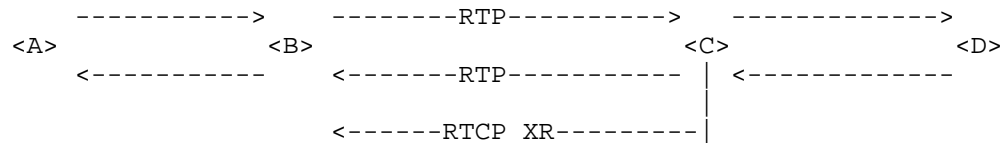
1. Introduction

This draft defines several new block types to augment those defined in RFC3611 for use in Quality of Service reporting for video over IP. The new block types defined in this draft are the IP Video Metrics Report Block, and the IP Video Metrics Configuration Block.

2. Definitions

2.1 Local and Remote IP Endpoints

A report block produced per this draft is normally produced by the endpoint of an RTP stream, and relates to the quality of the received RTP stream and impairments that may affect the RTP payload. The diagram below illustrates the potential end and mid points that may be involved in this process. Within the context of this example, endpoint "C" is the reporting endpoint and the RTCP XR report relates to the RTP stream from "B" to "C". The other points "A", "B" and "D" could also be generating RTCP XR reports.



With respect to RTCP XR report blocks generated by "C" in relation to the RTP stream from "B" to "C".

(i) The term "External" is used to relate to the network connected to the other side of "this" endpoint (i.e. to the connection from "C" to "D").

(ii) The Local IP Endpoint is "this" endpoint (i.e. "C")

(iii) The Remote IP Endpoint is the source for the RTP stream terminated at this endpoint, and for which packet/frame related metrics apply. (i.e. "B")

(iv) The Remote External Endpoint is the remote endpoint on the "external" side of this endpoint (i.e. "D").

(v) The endpoint on the external side of the Remote IP Endpoint (i.e. "A") does not have a specific term applied to it, however note that some metrics may apply to the "A" to "B" connection.

2.2 Cumulative and Interval Metrics

Cumulative metrics relate to the entire duration of the call to the point at which metrics are determined and reported, and are typically used to report call quality. Cumulative metrics generally result in a lower volume of data that may need to be stored, as each report supersedes earlier reports.

Interval metrics relate to the period since the last Interval report. Interval data may be easier to correlate with specific network events for which timing is known, and may also be used as a basis for threshold crossing alerts.

Note that interval metrics for the start and end of calls may be unreliable due to factors such as irregular interval length and the difficulty in knowing when packet transmission started and ended.

2.3 Bursts and Gaps

The terms Burst and Gap are used in a manner consistent with that of RTCP XR (RFC3611). A Gap is a period of time between Bursts such that any lost or discarded packets or frames are separated by some number of "good" packets or frames. A Burst is a period of time that fails the test for a Gap, and hence corresponds to a degraded quality period. The recommended value for Gmin in RFC3611 resulted in a Burst being a period of time during which the packet loss/discard rate exceeded 5%. As video is generally more sensitive to packet loss this report block uses a larger value for Gmin.

2.4 Numeric formats

This report block makes use of binary fractions. The terminology used is

S X:Y, where S indicates a signed representation,
X the number of bits prior to the decimal place and
Y the number of bits after the decimal place.

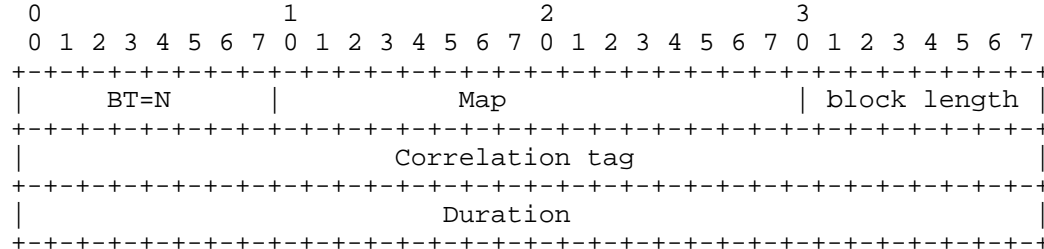
Hence 8:8 represents an unsigned number in the range 0.0039 to 255.996.

3 Video Metrics Report Block

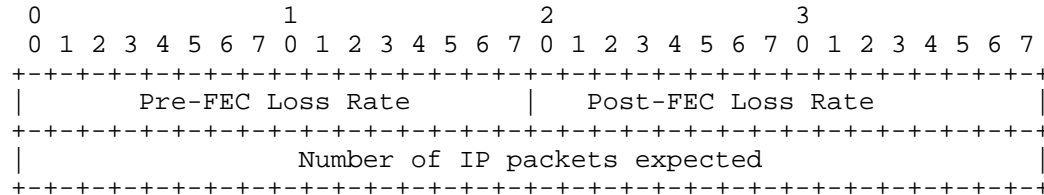
3.1 Block Description

This block comprises a header and a series of sub-blocks. The Map field in the header defines which sub-blocks are present.

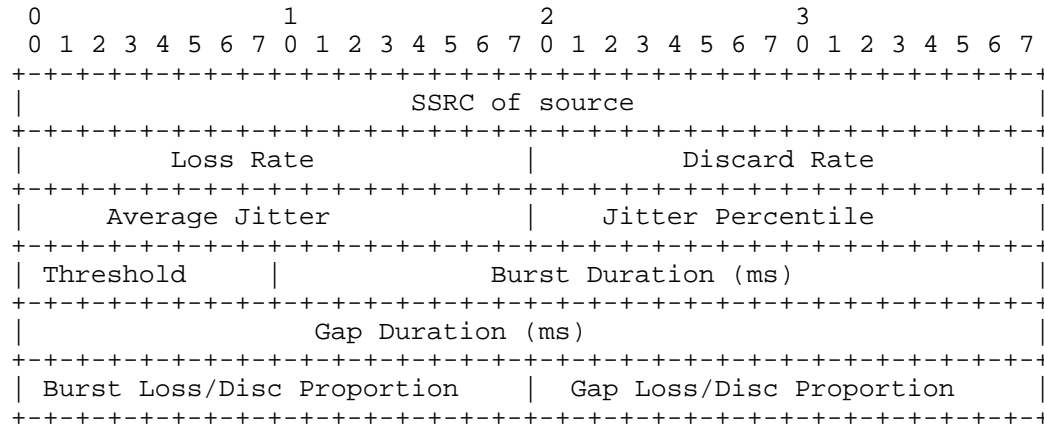
Header sub-block



IP Layer Loss Metrics sub-block



RTP Metrics sub-block



MPEG Transport Metrics sub-block

0								1								2								3							
0	1	2	3	4	5	6	7	0	1	2	3	4	5	6	7	0	1	2	3	4	5	6	7	0	1	2	3	4	5	6	7
Video Stream PID																Audio Stream PID															
Loss Rate																Discard Rate															
Average Jitter																Jitter Percentile															
Threshold								Burst Duration (ms)																							
Gap Duration (ms)																															
Burst Loss/Disc Percentage																Gap Loss/Disc Percentage															

Video/Audio Metrics sub-block

0								1								2								3							
0	1	2	3	4	5	6	7	0	1	2	3	4	5	6	7	0	1	2	3	4	5	6	7	0	1	2	3	4	5	6	7
Full Frame Loss Rate																Interpolated Frame Loss Rate															
VSTQ - Transmission Quality																VSPQ - Picture Quality															
VSAQ - Audio Quality																VSMQ - Multimedia Quality															
VSCQ - Control Quality																Reserved															
Video bit rate (bits/sec)																															
Audio bit rate (bits/sec)																															
A-V Delay (Network I/F)																A-V Delay (Video I/F)															
Round Trip Delay (media)																Round Trip Delay (control)															

Playout Buffer Metrics sub-block

0								1								2								3							
0	1	2	3	4	5	6	7	0	1	2	3	4	5	6	7	0	1	2	3	4	5	6	7	0	1	2	3	4	5	6	7
Playout Interrupt Count																Mean Playout Interrupt Size															
Playout buffer size																Mean buffer level															

3.2 Header

Implementations MUST send the Header block within each High Resolution Metrics report.

3.2.1 Block type

Nine Video Performance Reporting Metrics blocks are defined

```

mmm      = Video Metrics- Cumulative, Locally Generated
mmm+1    = Video Metrics- Cumulative, Relayed from Remote IP Endpoint
mmm+2    = Video Metrics- Cumulative. Relayed from Remote External
          Endpoint
mmm+3    = Video Metrics- Interval - Locally Generated
mmm+4    = Video Metrics- Interval - Relayed from Remote IP Endpoint
mmm+5    = Video Metrics- Interval - Relayed from Remote External
          Endpoint
mmm+6    = Video Metrics- Alert - Locally Generated
mmm+7    = Video Metrics- Alert - Relayed from Remote IP Endpoint
mmm+8    = Video Metrics- Alert - Relayed from Remote External
          Endpoint

```

The time interval associated with these report blocks is left to the implementation. Spacing of RTCP reports should be in accordance with RFC3550 however the specific timing of RTCP XR Video reports may be determined in response to an internally derived alert such as a threshold crossing.

3.2.2 Map field

An Map field indicates the optional sub-blocks present in this report. A 1 indicates that the sub-block is present, and a 0 that the block is absent. If present, the sub-blocks must be in the sequence defined in this document.

The bits have the following definitions:

```

0 RTP Metrics block
1 MPEG Transport Metrics block
2-15 Reserved, set to 0

```

3.2.3 Block Length

The block length indicates the length of this report in 32 bit words and includes the header and any extension octets.

3.2.5 Correlation tag

The correlation tag facilitates the correlation of this report block with other call or session related data or endpoint data.

3.3 IP Layer loss metrics sub-block

The IP Layer loss metrics sub-block MUST be present.

3.3.1 Pre-FEC Loss Rate

The proportion of IP packets lost before the effects of FEC, expressed as a binary fraction in 0:16 format.

3.3.2 Post-FEC Loss Rate

The effective proportion of IP packets after before the effects of FEC, expressed as a binary fraction in 0:16 format.

3.3.3 Number of IP Packets Expected

The number of IP packets that the receiving system estimates that it should have received.

3.4 RTP Metrics sub-block

If RTP transport is used, the RTP Metrics sub-block MUST be present and if present MUST be indicated in the Map field.

3.4.1 Source SSRC

The SSRC associated with the RTP stream to which this report block relates.

3.4.2 Loss Rate

The (post-FEC) proportion of RTP frames lost in the network.

3.4.3 Discard Rate

The proportion of RTP frames discarded due to late or early arrival.

3.4.4 Average Jitter

The average PPDV, MAPDV or Y.1540 jitter level in milliseconds

3.4.5 Jitter Percentile

The proportion of RTP frames arriving later than the jitter threshold

3.4.6 Threshold

The Gmin threshold associated with the definition of bursts and gaps.

3.4.7 Burst duration

The duration of bursts of lost and discarded RTP frames expressed in milliseconds.

3.4.8 Gap duration

The mean duration of gaps between bursts expressed in milliseconds.

3.4.9 Burst loss/discard proportion

The proportion of frames lost or discarded during burst periods expressed as a binary fraction.

3.4.10 Gap loss/discard proportion

The proportion of frames lost or discarded during burst periods expressed as a binary fraction.

3.5 MPEG Transport Metrics sub-block

The MPEG Transport Metrics sub-block MUST be present if MPEG Transport is used, and if present MUST be indicated in the Map field.

3.5.1 Video Stream Program ID

The Program ID (PID) associated with the video stream.

3.5.2 Audio Stream Program ID

The Program ID (PID) associated with the audio stream

3.5.3 Loss Rate

The (post-FEC) proportion of Transport Units lost in the network.

3.5.4 Discard Rate

The proportion of Transport Units discarded due to late or early arrival.

3.5.5 Average Jitter

The average PPDV, MAPDV or Y.1540 jitter level in milliseconds

3.5.6 Jitter Percentile

The proportion of Transport Units arriving later than the jitter threshold

3.5.7 Threshold

A threshold associated with the definition of bursts and gaps.

3.5.8 Burst duration

The duration of bursts of lost and discarded Transport Units expressed in milliseconds.

3.5.9 Gap duration

The mean duration of gaps between bursts expressed in milliseconds.

3.5.10 Burst loss/discard proportion

The proportion of Transport Units lost or discarded during burst periods expressed as a binary fraction.

3.5.11 Gap loss/discard proportion

The proportion of Transport Units lost or discarded during burst periods expressed as a binary fraction.

3.6 Video/Audio Metrics sub-block

The Video/Audio Metrics sub-block MUST be present.

3.6.1 Full frame loss rate

The proportion of full or intra-frame encoded video frames lost

3.6.2 Interpolated frame loss rate

The proportion of interpolated video frames lost

3.6.3 VSTQ - Video Service Transmission Quality

The video service transmission quality expressed as a score in the range 0.0 to 50.0. This is a codec independent measure of the ability of the bearer channel to support reliable video.

3.6.4 VSPQ - Video Service Picture Quality

The video service picture quality expressed as a score in the range 0.0 to 50.0. This is a codec dependant measure that is related to the subjective quality of the decoded video stream.

3.6.5 VSAQ - Video Service Audio Quality

The video service audio quality expressed as a score in the range 0.0 to 50.0. This is an audio codec dependant measure that is related to the subjective quality of the decoded audio stream(s).

3.6.6 VSMQ - Video Service Multimedia Quality

The video service multimedia quality expressed as a score in the range 0.0 to 50.0. This is a composite audio/video measure that is related to the overall subjective user experience and considers picture quality, audio quality and audio/video synchronization.

3.6.7 VSCQ - Video Service Control Plane Quality

The video service control plane (trick play) quality expressed as a score in the range 0.0 to 50.0. This is a measure that is related to the performance of the video stream control channel.

3.6.8 Reserved

3.6.9 Video bit rate

The short term average bit rate of the video codec.

3.6.10 Audio bit rate

The short term average bit rate of the audio codec

3.6.11 Audio-Video Delay (Network Interface)

The relative delay between audio and video measured before the decoder and expressed in milliseconds

3.6.12 Audio-Video Delay (Video Interface)

The relative delay between audio and video measured after the decoder and expressed in milliseconds

3.6.13 Round Trip Delay (Media)

The round trip delay for the media path, required only for interactive video sessions.

3.6.14 Round Trip Delay (Control)

The round trip delay for the video control (trick play) path.

3.7 Playout Buffer Metrics sub-block

The Playour Buffer Metrics sub-block MUST be present.

3.7.1 Playout Interruption Count

The number of interruptions in video playout that have occurred due to playout buffer starvation or excessive packet loss.

3.7.2 Mean Playout Interruption Size

The mean size of interruptions in playout, expressed in multiples of 100 milliseconds

3.7.3 Playout Buffer Size

The playout buffer size, expressed in multiples of 100 milliseconds

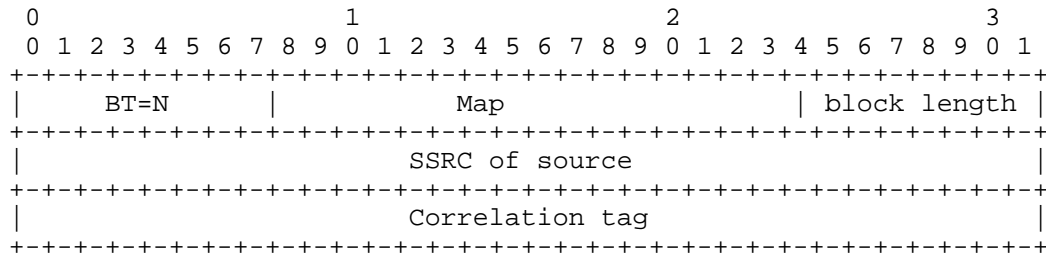
3.7.4 Mean Buffer Size

The average playout buffer size expressed in multiples of 100 milliseconds

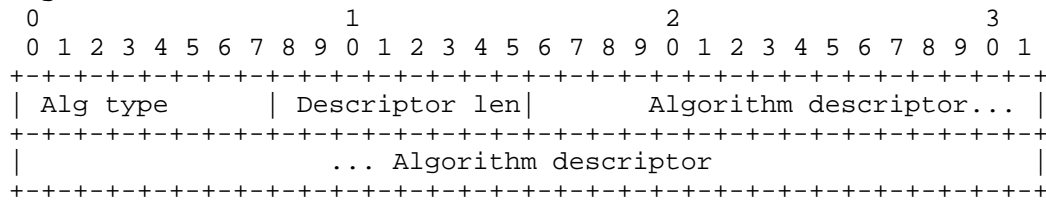
4. RTCP HR Configuration Block

This block type provides a flexible means to describe the algorithms used for call quality calculation and other data. This block need only be exchanged occasionally, for example sent once at the start of a call.

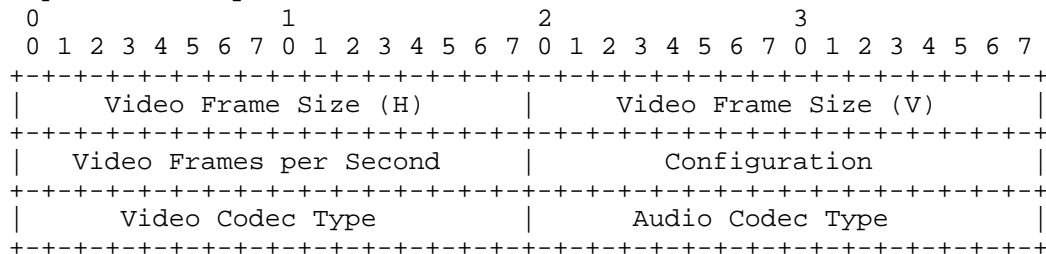
Header sub-block



Algorithm sub-block



Payload Descriptor sub-block



4.1 Header

Implementations MUST send the Header block within each Video Metrics Configuration report.

4.2.1 Block type

One Video Metrics Configuration block is defined

mmm+9 = Video Metric Configuration Block

The time interval associated with these report blocks is left to the implementation. Spacing of RTCP reports should be in accordance with RFC3550 however the specific timing of RTCP HR reports may be determined in response to an internally derived alert such as a threshold crossing.

4.2.2 Map field

An Map field indicates the optional sub-blocks present in this report. A 1 indicates that the sub-block is present, and a 0 that the block is absent. If present, the sub-blocks must be in the sequence defined in this document.

The bits have the following definitions:

- 0 Algorithm Descriptor 1
- 1 Algorithm Descriptor 2
- 2 Algorithm Descriptor 3
- 3 Algorithm Descriptor 4
- 4 Algorithm Descriptor 5
- 5 Algorithm Descriptor 6
- 6 Algorithm Descriptor 7
- 7 Algorithm Descriptor 8
- 8 Payload Descriptor
- 9 Vendor Specific Extension
- 10-15 Reserved, set to 0

4.2.3 Block Length

The block length indicates the length of this report in 32 bit words and includes the header and any extension octets.

4.2.4 SSRC

The SSRC of the stream to which this report relates.

4.2.5 Correlation tag

The correlation tag facilitates the correlation of this report block with other call or session related data or endpoint data.

4.3 Algorithm description

The Algorithm Description sub-block MAY be present however if present MUST be indicated in the MAP field

The Algorithm descriptor is a bit field which indicates which algorithm is being described. The bits are defined as:-

- Bit 0: MOS-LQ Algorithm
- Bit 1: MOS-CQ Algorithm
- Bit 2: R-LQ Algorithm
- Bit 3: R-CQ Algorithm
- Bit 4: Video Monitoring Algorithm
- Bit 5-7: Reserved and set to 0

The descriptor length gives the overall length of the descriptor in 32 bit words and includes the algorithm descriptor and length fields.

The algorithm descriptor is a text field that contains the description or name of the algorithm. If the algorithm name is shorter than the length of the field then the trailing octets must be set to 0x00.

For example, an implementation may report:

```
Algorithm descriptor = 0x0F   - R and MOS algorithms
Descriptor length = 3        - 3 words
Descriptor = "Alg X" 0x00    - description
```

4.4 Payload description field

The Extended Payload Description sub-block MAY be present however if present it MUST be indicated in the Header Map field. This sub-block provides a detailed description of the payload format used.

4.3.1 Video Frame Size - Horizontal

The encoded video frame size expressed in pixels

4.3.2 Video Frame Size - Vertical

The encoded video frame size expressed in pixels

4.3.3 Video Frames per Second

The number of video frames per second

4.3.4 Video configuration

Format TBD but e.g.:-

- HD
- Widescreen
- Audio channels (1, 2, 5.1..)

4.3.5 Video Codec Type

Format TBD but - MPEG1,2,4....

4.3.6 Audio Codec Type

Format TBD

5. Practical Applications

5.1 Overview

The objective of this section is to identify a number of cases in which there could potentially be some ambiguity in the application of the report blocks defined above or some exceptions to the defined operation of the metrics.

6. Summary

7. IANA Considerations

8. 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.

9. Contributors

10. Informative References

- [1] Bradner, S., "Key words for use in RFCs to Indicate Requirement Levels", BCP 14, RFC 2119, March 1997.
- [2] Schulzrinne, H., Casner, S., Frederick, R. and V. Jacobson, "RTP: A Transport Protocol for Real-Time Applications", STD 64, RFC 3550, July 2003.
- [3] Friedman, T., Caceres, R. and A. Clark, "RTP Control Protocol Extended Reports (RTCP XR)", RFC 3611, November 2003.

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