Video Over RTP Enhancements and Testing Methods

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Abstract

RTP, or real time protocol, is designed for internet transmission with minimal latency. Applications that require near millisecond latency will implement RTP to achieve such goals. Common applications for RTP include real time video streams, online conferencing applications, and voice over IP. To enable additional enhancements and modifications, RTP allows for extensions to the header in order to transmit extra data with the video and audio streams. This study explores the feasibility of various modifications of this extension header and how this impacts data transfer performance.

First we attempt to implement sending and receiving Video QOS metrics inside of the RTP packet rather than through the RTP control stream. Second, we add a role based video streaming to ensure video is directed only to authorized users. In addition, we design a testing scheme to measure latency and other QOS metrics in order to compare base protocol designs against other modifications.

RTP Packet Header

Bit	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24	25	26	27	28	29	30	31
0	V		P	X	CC				M	Payload Type							Sequence Number															
32	Timestamp																															
64	SSRC identifier																															
96	CSRC identifier(s)																															
...	Profile-specific Extension Header Identifier																Extension Header Length															
...	Extension Data																															

RTP Packet Header Layout

V (Version): Indicates the verision of the protocol
P (Padding): Tells the reciever that padding is enabled.
X (Extension): Tells the reciver that extension modifications to the header are enabled.
CC (CSRC Count): Stores the number of CSRC identifiers
M (Marker): Used to indicate to the reciver this is special packet
Payload Type: Indicates the type of data in the packet
Sequence Number: The packet number. Used to correct out of order transmission and detect packet loss
Timestamp: Used for synchronization of different streams
SSRC identifier: Determines who has sent the packet
CSRC identifier(s): Lists who has contributed data to the packet
Profile Specific Extension Header Identifier: Extension specific headers
Extension Header Length: Length of the extension data, including header
Extension Data: Extra profile specifc data

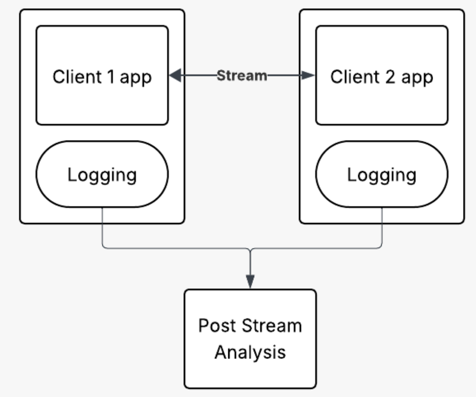
Video and Audio Transmission Over RTP

Generally, multiplexing video and audio RTP streams over the same session is not recommended. If one stream changes protocols, it becomes impossible to determine which one had changed. It also provides the ability to particpate in each seperatly. As a result, video and audio streams are sent seperately and require reconstruction.

Streams may be sampled at different rates, and require synchronization. This is done via a common clock, typically via NTP. By combining the data from the NTP from both streams, a common mapping can be found for synchronization

Testing Metrics

To test the efficacy of RTP transmissions under different network conditions, we measure these network quality metrics. Each has a unique effect on the quality of an RTP stream. To Eliminate overhead of analyzing results during the stream, data will instead be collected and logged to be analyzed later.



Transmission Delay

Transmission delay measures the time needed for the sender to push all bits onto the network.

$$t = \frac{L}{R}$$

where L is size of packet in bits, and R is bandwidth, measured in bits per second

Transmission delay has a critical role in real time systems, as it propogrates itself as higher latency.

IP Packet Delay Variation (IPDV)

IPDV is the can be measured as the root mean square of packet variation samples.

A sample, k, measures the average of the absolute difference in latency between a successive n packets. Where R_i and S_i are the arrival times and sending time of the packet respectively

$$IPDV(k) = \frac{\sum_{i=n*(k-1)+2}^{k*n+1} |(R_i - R_{i-1}) - (S_i - S_{i-1})|}{n}$$

Applying the absolute value, rather than squaring, makes the sample average resistant to a small number of outliers, which are largely unnoticeable to the user.

The root mean square of the samples can then be calculated as:

$$IPDV_{RMS}(l) = \sqrt{\frac{\sum_{k=N*(l-1)+1}^{l*N} IPDV^2(K)}{N}}$$

Root mean squares provides a measure of variation as a mangnitude, unlike a standard deviation, which measures variation as a distance from the mean.

IPDV is noticeable in congested networks. Real time apps compete with other network services for bandwidth. This manifests itself as high packet variation, resulting in a stream that buffers often and offers a lower quality of experience for the user.

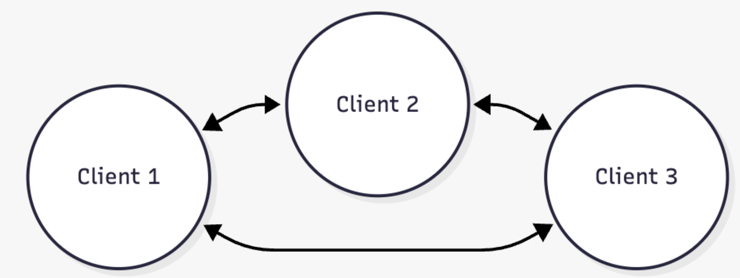
Packet Loss

Packet loss is the percetage of packets that are successfully received by a client

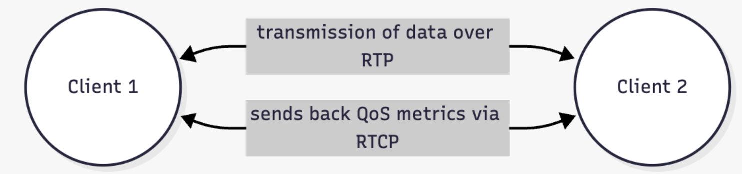
$$PL = \frac{\text{Packets Sent} - \text{Packets Received}}{\text{Packets Sent}} * 100\ \%$$

High packet loss causes RTP streams to be choppy and unresponsive. Poor network conditions and failed routing are the main causes of packet loss

Sessions



Extensions



In a traditional RTP setup, data and QoS tranmssions are seperated into different transmissions on seperate ports

With the extension header, it is possible to transmit both types of data in the header. This adds complexity to RTP in encoding and decoding, but has the benefit of reducing the overhead when seperate protocols are used.

References

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