End to End VoIP stats

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HPE Edge;

End to End VoIP stats

Abstract

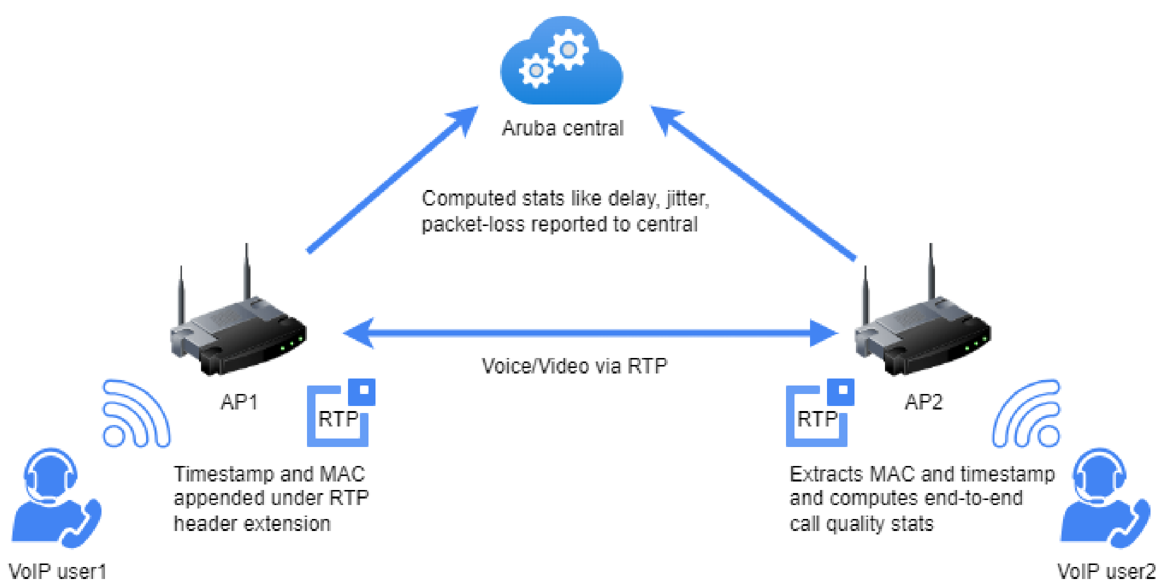
UCC or Unified Collaboration and Communication tools are increasingly being used in this day and age of remote working for work and collaborate with folks from various remote locations. With the future looking at hybrid work place and with the increase in adoption of applications such as Teams, Zoom, reliability of the network and testing it for latency / delay and jitter and localizing network issues becomes a paramount importance to a network administrator. With the proliferation of WLANs, there has been a prominent leap in supporting real-time applications such as Voice over Internet Protocol (VoIP) over WLANs.

Real-time Transport Protocol (RTP) is the widely adopted protocol for providing end-to-end network transport functionality suitable for transmitting real-time data (such as voice, video). This paper proposes a solution where Aruba APs add timestamps to RTP header extensions for the computation of bidirectional end-to-end call quality stats of every client running a VoIP call in the network.

Problem statement

Modern campus networks are evolving towards AI based intelligent systems that are driven by data from the network. VoIP related issues are by far, the toughest issues to debug especially given that they are bandwidth hungry applications and the RTP payloads are encrypted. Some VoIP service providers report Mean Opinion Score (MOS) as a measure for the call quality. However, when a call reports poor MOS score, service providers may not be able to determine the cause without visibility into the network when the call was made. This draft proposes a solution where AP includes timestamps in RTP header extensions and the AP at the receiving endpoint extracts this timestamp and computes the VoIP quality with the required visibility in the network to identify and eliminate issues that affects the network performance. Additionally, this solution would also report call participants' details to present a dashboard view of call details on a per-client basis.

Our solution



Currently, UCC feature on the APs provides the capabilities of classifying media (voice/video) flows, prioritizing the RTP flows and monitoring call quality stats (delay, jitter, packet-loss) on the wireless direction (downstream) only for audio calls. These stats that are reported by monitoring RTP traffic are between AP and VoIP client and not end-to-end. The proposed solution provides more precise end-to-end measurements of these stats by adding timestamps under RTP header extensions (a custom header only meant for data exchanged between APs) for every RTP packet that has been received by the AP. The AP connected to the receiver VoIP endpoint shall extract this timestamp and note the timestamp when the packet was transmitted to the receiver device. The time difference between these timestamps provides delay and the variation in delay yields jitter. End-to-end packet-loss can also be computed by monitoring sequence numbers pertaining to a given Synchronization Source Identifier (SSRC). Wallclock time of all the APs would be synchronized with the same NTP server (Network Time Protocol) for uniformity in timestamp calculation. These stats would be sent periodically to Aruba Central where the stats shall be aggregated by maintaining a database to provide live call data in real time and maintain a dashboard that describes the quality of the call. Additional value add to this would be including MAC address of the VoIP client under another custom RTP header extension which would enable identification of participants of the call. This information could be collated at Aruba Central to publish the call details on a per-client basis.

With campus networks moving towards AI based intelligent systems, stats like these would serve as a crucial feedback for automated systems to identify VoIP issues in real-time and take corrective actions or notify as alerts for providing deeper insights with a utility like Aruba NetInsight.

Evidence the solution works

We have experimented with a Teams call where AP firewall was programmed to add timestamp as a part of RTP header extension and observed that the timestamps are exchanged successfully between the APs to which both the endpoints are connected.

Competitive approaches

VoIP service providers like Teams report Mean Opinion Score (MOS) as a measure for the call quality. However, when a call reports poor MOS score, service providers may not be able to determine the cause without visibility into the network when the call was made. Also obtaining stats from Teams requires API calls like Graph API which has its associated cost.

Reply Cloud is a VoIP monitoring service provider who provides cloud based VoIP monitoring solution with a hardware device for monitoring and analyzing call quality from within the network but it is only capable of providing stats of SIP calls. This solution provides the capabilities of monitoring enterprise grade apps such as Teams, zoom etc that too without any additional hardware. Additionally, this feature clubbed with UCC would provide deeper insights with voice/video distinction.

Current status

Currently, as a proof of concept, we have experimented with a Teams and this could be extended to applications such as Skype, SIP and zoom.

Next steps

We could enhance the solution provide visually appealing stats to depict the story of the call quality of every single call (both live and historical calls). For an AI managed network, this solution would provide a crucial feedback for automated systems to identify VoIP issues in real-time and take corrective actions. Poor VoIP instances could also be notified as alerts for providing deeper insights to a network admin along with a utility like Aruba NetInsight. With UCC classification, we could also remotely capture pcaps selectively for VoIP flows, downloadable from central to debug VoIP packet flow. The inclusion of MAC address under a custom header extension would enable both the endpoint APs to identify the participants of the call and this data can be collated at Aruba Central to present a dashboard of end-to-end stats of every individual devices involved in every single VoIP call.

References

VoIP Quality Testing l Reply Cloud

A General Mechanism for RTP Header Extensions [RFC-8285]

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