FEATURE BRIEF: MEAN OPINION SCORE

Unified communications (UC) are the lifeblood of any enterprise. UC relies on Voice over Internet Protocol (VoIP) as a foundation to build on. VoIP quality improvements have usually focused on how to encode and route voice traffic over IP networks. Alternatively, Codecs have focused on trading off compression with available bandwidth and recovering from packet losses to improve call quality.

Quality of Service (QoS) for VoIP differentiates traffic based on IP source addresses by giving higher priority to voice packets to improve quality. However, the quality of VoIP calls remains a major problem due to network impairments such as delay, jitter, and packet loss. And when packets are sent over the Internet in a software-defined wide-area network (SD-WAN) deployment, the problem of VoIP quality becomes more acute as the variations in network impairments can be large.

The 128T Session Smart™ Router addresses this issue by measuring delay, jitter, and packet loss between routers in the 128T network using Bidirectional Forwarding Detection (BFD). Mean Opinion Score, or MOS, is then used to classify traffic flows in the network as these impairments occur. This capability allows managed service providers to provide their customers with superior UC and contact center services, and it allows enterprises to provide their employees with unparalleled services that enable them to improve business operations.

Network Impairments Defined

Latency

Network latency does not affect the quality of the delivered audio, but rather the interaction between the two end users. At 100ms (milliseconds) of latency, the users start talking on top of each other, and at 300 ms, the conversation becomes impossible to follow.

Jitter

High jitter results in choppy voice or temporary glitches. VoIP devices implement jitter buffering algorithms to compensate for packets that arrive at high timing variations, and packets can even get dropped when they arrive with excessive delay.

Packet Loss

Voice packets always use User Datagram Protocol (UDP). Packets may not arrive at the destination, or get discarded if they arrive delayed or contain errors. This results in missing audio information at the destination.

What is MOS?

MOS is a commonly used measure for video, audio, and audiovisual quality evaluation. ITU-T has defined several ways of referring to MOS in Recommendation P.800.1, and has made it possible to estimate MOS by using packet loss, delay, and jitter for VoIP packets.

Latency, jitter, and packet loss cannot be eliminated from real world networks. However, network paths can be chosen based on these performance metrics to ensure superior VoIP quality. The industry has adopted MOS as the universal metric to measure and classify the conversation quality that happens over a network. As the name suggests, it is based on the opinion of the user and ranges from 1.0 to 5.0 with the following classifications:

MOS	Quality	Impairment
5	Excellent	Imperceptible
4	Good	Perceptible but not annoying
3	Fair	Slightly annoying
2	Poor	Annoying
1	Bad	Very Annoying

Typically, the highest MOS that can be achieved is 4.5 for the G.711 codec. The cutoff MOS score for calls that can be tolerated is around 2.5. Ideally, the MOS score is calculated by asking the participants to put a score to the conversation. Another way to estimate the call quality is based on the network's latency, jitter, and packet loss.

How to Calculate MOS

The most popular method to calculate MOS based on network performance metrics is based on the E-model, which calculates the rating factor, R, which then is used to derive the MOS value. An R value larger than 93.2 gets the maximum MOS value. Depending on latency, jitter, and packet loss, points are deduced from the 93.2 to get an estimate of the MOS value. Details of this calculation and how to arrive at MOS score can be found in Recommendation G.107.

The following table shows sample MOS scores calculated using the above formula:

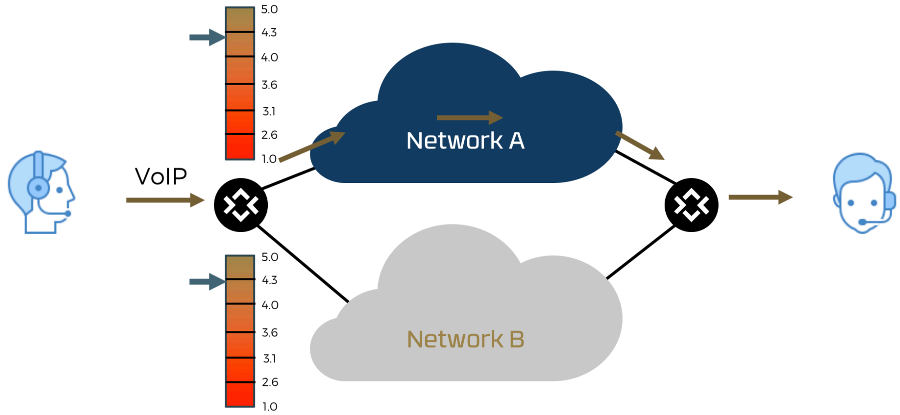
Delay (ms)	Jitter (ms)	Loss (%)	MOS
100	10	5	3.92
200	50	5	3.19
100	10	10	3.35
100	10	20	2.06

Once MOS is calculated for different paths, the voice streams can utilize paths based on MOS associated with the service. The session can migrate to an alternate path when MOS for a given path changes.

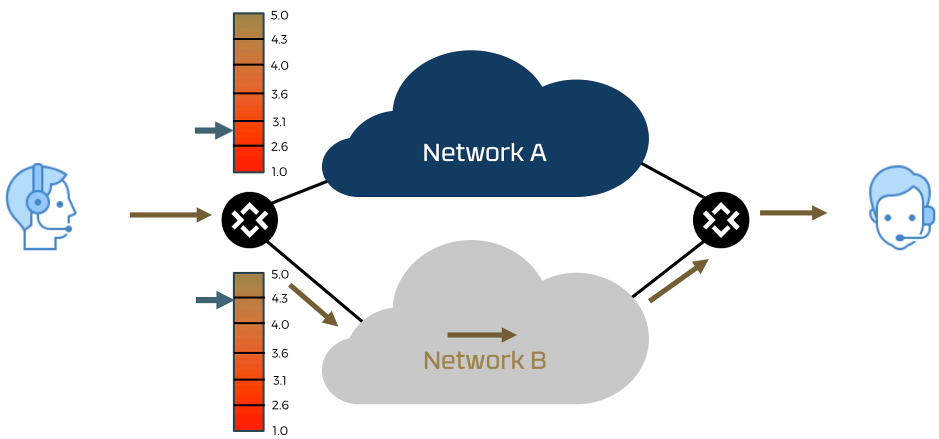
MOS on 128T Session Smart™ Routers

128T Session Smart™ Routers measure delay, jitter, and packet loss between themselves using Bidirectional Forwarding Detection (BFD). This data is used for selecting paths for different flows when multiple paths are available between the 128T Session Smart Routers.

The 128T Session Smart Router calculates MOS based on the BFD data. The network administrator can associate a threshold MOS with a session, and paths for that session will be chosen to ensure that the path meets the minimum defined MOS. If there is a MOS associated with a session type, then it supersedes any associated delay, jitter, and packet loss thresholds.



The 128T Session Smart Router will continue to monitor the session and the link performance. If MOS associated with the link falls below the MOS threshold associated with the session on that link, then the router switches the session to an alternate path that meets the MOS required by that session.



The administrator can define if the session should continue to stay on the alternate path or switch back to the original path when the MOS associated with the original path improves.

This ensures that VoIP streams are always on a path that meets the required MOS and provides a superior end user experience.

Summary

The 128T Session Smart™ Router uses enhanced BFD to proactively monitor network impairments of different network paths, making it possible to estimate a MOS score associated with these paths based on the performance values. This can then be used to choose specific paths in the network based on a MOS value associated with a VoIP stream. Based on performance criteria, the VoIP stream can be switched to other paths based on the MOS value to guarantee that the stream takes a path which offers the required performance.

This ability to calculate MOS values associated with different network paths and choose paths to meet specific MOS values for VoIP streams provides customers and managed service providers with the ability to offer higher quality for unified communications/contact center services to their end users.

Case Study

ConvergeOne is a leading IT services provider of collaboration and technology solutions to large and medium enterprises. ConvergeOne has over 11,000 enterprise customers including 73% of Fortune 100 companies. ConvergeOne Secure Connect is a subscription-based, fully-managed SD-WAN that connects customers' networks to ConvergeOne Cloud Experience (C1CX) collaboration solutions. The solution is powered by 128T Session Smart™ Routers.

One of ConvergeOne's customers is a large financial services company that utilizes the 128T Session Smart Routers to connect to C1CX for unified communications (UC)/contact center(CC) services. The financial services company launched an internal banking software update which resulted in a short term 4x volume increase for CC traffic. During this time, the 128T Session Smart Routers provided a temporary capacity increase by routing MPLS overflow traffic over quality-monitored encrypted Internet connections. By monitoring MOS over different pathways, the 128T Session Smart Routers provided intelligent call routing that ensured a superior end user experience. There was an unanticipated failure in the MPLS network. The 128T Session Smart Routers detected the degradation in MOS scores and successfully migrated all the calls to the Internet without a single call drop.

More details of the deployment can be found at a joint webinar hosted by 128 Technology and ConvergeOne.