CODEC-BASED QOS ANALYSIS OF VOIP OVER CONVENTIONAL NETWORK AND SDN

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ABSTRACT

Due to the advancement of technology in recent years, there has been a significant increase in the usage of bandwidth-consuming real-time applications such as VoIP, web-streaming. To cater to these ever increasing modern networking demands, Software-Defined Networking (SDN) is considered as the main structure for the future networking. We have analyzed and compared the QoS performance over conventional network and SDN, using Riverbed Modeler and Mininet Emulator, respectively, by considering eight different VoIP call scenarios based on the combination of voice codecs and call-type. The results for a single VoIP call scenarios showed 6 to 13 percent improvement in the throughput performance. In the case of concurrent VoIP calls, SDN can effectively handle congestion due to the efficient bandwidth utilization and the presence of a proactive centralized controller, resulting in improved QoS performance of VoIP in terms of throughput, jitter, packet loss and Mean Opinion Score (MOS).

Keywords: VoIP, SDN, QoS, Mininet emulator, Voice codecs

INTRODUCTION

The computer networks have evolved into a highly complex entity that are difficult to manage and struggle to scale to the needs of some of today's requirements. These conventional technologies fall short of the size and complexity of many modern networking deployments despite their impressive historical record [3]. There is a need to accelerate the innovations in networking to cater to increasing demands of the modern network applications and services. Software-Defined Networking (SDN) is a relatively new approach in the networking domain developed to address and overcome the weaknesses of the conventional networks. SDN has changed the way of network designing and management.

Over the past few years, there has been an exponential increase in the use of real-time applications and multimedia streaming over the Internet. These applications are delay-sensitive and need high bandwidth, hence requiring to be transmitted with higher priority over the networks. SDN plays an important role in traffic prioritization based on the Quality of Service (QoS) and service level agreements. Voice over Internet Protocol, or commonly known as VoIP is a typical interactive real-time application that needs to be provisioned instantly over the networks due to its strict QoS requirements, which includes jitter to be less than 30 ms, delay to be no more than 150 ms and under 1% packet loss.

In this paper, we have analyzed and compared the QoS performance of VoIP with reference to eight different scenarios, four each on the conventional network and the SDN. These scenarios are based on two voice codecs, G.711 and G.729A, as well as a single VoIP call scenario and multiple concurrent VoIP calls scenario. We have implemented the conventional network topology in “Riverbed Modeler,” and the similar SDN topology with the help of “Ryu” as the remote SDN controller using “Mininet Emulator.”

The remainder of the paper is structured as follows: First, we briefly describe the related work and technical concepts, which include the fundamentals of Software-Defined Networking, Voice over IP and QoS parameters. The experimental setup for both, conventional network and SDN, is outlined further. After this, we present the simulation results and analysis of our research work. Finally, the paper is concluded with the last section.

RELATED WORK AND TECHNICAL BACKGROUND

A. Related Work:

Researchers in [1] conducted a comparison of conventional networks based on routers that use the OSPF routing protocol and Software-Defined Networks that use the OpenDayLight controller and Open vSwitches. They analyzed and compared network topologies based on the number of routers or Open vSwitches used, and the various payload conditions applied to them. The authors of [2] proposed an SD-WAN solution to ensure business quality VoIP communication on a home network. The experiments involved the actual setup of Cisco SD-WAN enabled edge routers, and the results of jitter, latency, packet loss and MOS were compared with the conventional broadband, SD-WAN single link, and SD-WAN dual link. The authors of [4] presented a performance evaluation of SDN-based wireless network for real-time applications like VoIP, file transfer and video streaming. They used Mininet Wi-Fi the emulator and Ryu as the SDN controller, and compared the SDN performance based on the impact of various background traffic.

B. Software-Defined Networking (SDN):

In this digital era, faster networks with least downtime are high priority. SDN architecture is introduced as the core framework of the future networking. Separation of control plane and data plane in SDN alters the conventional network architecture. The provision of a centralized controller and various applications allows programmability in the network.

In general, there are three main layers of this architecture namely Data plane, Control plane and Application plane. Primarily consisting of the end devices, the data plane is responsible for forwarding and filtering services. However, in SDN, these devices in the data plane have no decision capabilities, as the programmable SDN controller at the control plane takes care of the underlying low-level complexities. Being centrally located, the controller is capable of making optimal decisions due to its ability of seeing the entire network. The application plane takes care of network services like firewalls, routing and load balancing. These services run above the controller and decide how to best supervise the packet forwarding in the network.

The Application Programming Interfaces (APIs) are included in the SDN architecture to communicate between its planes. The network management traffic between application plane and control plane is handled by the northbound API. The southbound API is aimed for the controller to communicate with data plane devices and is intended to impose controller strategic decisions on the devices. This is where the OpenFlow protocol comes into play. The East-West API is responsible for communication between controllers within the control plane. It keeps a track of controllers' health with the help of status messages.

The decoupling of planes in SDN enables logically centralized controller to provide real-time automation of operation, provisioning, monitoring and troubleshooting, resulting in higher uptime and, as a result, the improved performance.

C. Voice over IP and Voice Codecs:

VoIP is a typical real-time application whose payload data is required to be processed instantaneously, with minimal delay. The protocols used in the VoIP communication typically include Real-time Transport Protocol (RTP), Session Initiation Protocol (SIP) and User Datagram Protocol (UDP). It is very important to properly apply queuing algorithms, QoS model and traffic priority for all the traffic in accordance with the networking environment.

A codec is a mechanism used in IP telephony to maximize bandwidth utilization by compressing the payload of the call and decompressing it at the call destination. G.711 is a voice codec used in modern digital telephone network that is known for providing the best voice quality at 64 kbps bit rate. The G.729A codec with a bit rate of 8 kbps, has a low bandwidth requirement, however, provides a good voice quality. Generally, higher the bit rate, better is the voice quality.

D. Quality of Service (QoS):

Congestion occurs in a network when data traffic from multiple links is aggregated onto a single link for transmission, or when link speeds are mismatched, or when a link fails. A proper QoS provisioning and queuing algorithms help in traffic prioritization in order to optimize the network performance and alleviate the congestion. The QoS provisions are based on the Differentiated Services Code Point (DSCP), a predefined 8-bit field in the packet header.

The overall network performance is evaluated based on the QoS parameters such as throughput, jitter, packet loss, Mean Opinion Score (MOS). Throughput is the rate at which packets are successfully delivered from one point on the network to another. The loss of packets along the way reduces throughput and eventually, the QoS. The variation in the delay of received packets is referred to as jitter. Packets are sent in an evenly spaced continuous stream from the source. However, the delay between each packet can vary due to network congestion or improper queuing. To support real-time and interactive traffic, jitter must be controlled and minimized. Packet loss is the total number of packets lost or dropped during transmission, which can be caused by congestion, excessive delay, or by an error in the packet header. MOS value represents the user's perception of the QoS offered by the network on a scale of 1 (bad) to 5 (excellent).

EXPERIMENTAL SETUP

The same topology design is used in both conventional and SDN networks, with the only difference being the devices used. It's basically a tree topology with two hosts each connected to two switches, and those switches connected to another switch. A router is used in a conventional network, whereas SDN uses an OpenFlow protocol-based Open vSwitch with a controller overhead. We simulated the following four scenarios using "Riverbed Modeler" for conventional network and "Mininet Emulator" for SDN topology,

1. G.711 Single Call,
2. G.729A Single Call,
3. G.711 Concurrent Calls,
4. G.729A Concurrent Calls.

A. Riverbed Modeler:

Riverbed Modeler is a discrete event simulator that is used to model and analyze networks. Along with the network topology, we must configure VoIP as an application and define its associated flows on our network using Application Configuration and Definition Configuration, as shown in fig. 1 and fig. 2. Encoder Scheme, ToS or DSCP, Repetition Pattern, Operation Mode are all defined here.

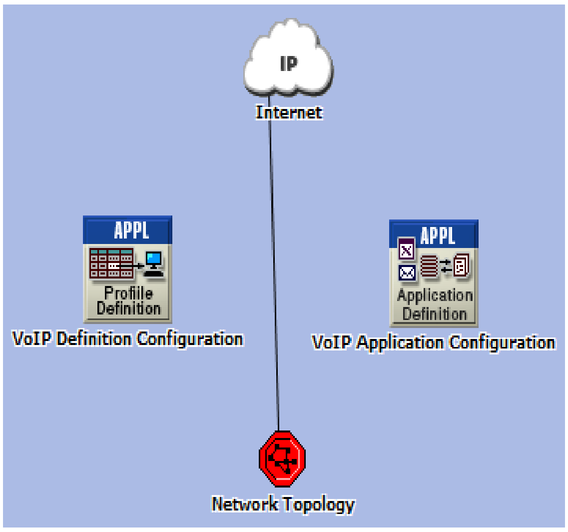

Figure 1: Riverbed Modeler Setup

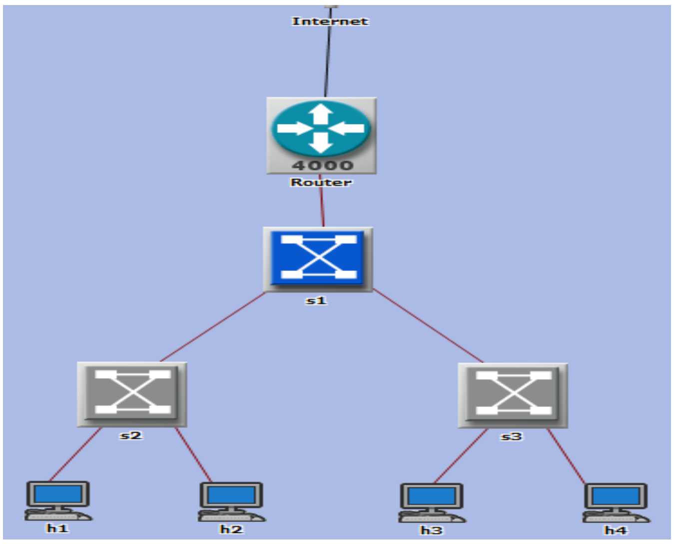

Figure 2: Conventional Network Topology in Riverbed Modeler

B. Mininet Emulator:

Mininet Emulator, which runs on the Linux platform, creates virtual hosts, switches, controllers and links based on software network. The Open vSwitch in the SDN topology replaces the router's function, enabling the network to communicate automatically. We interfaced "Ryu" as a remote SDN controller on a localhost which uses Python for scripting. As shown in the fig. 3, we have used a similar network topology in SDN. Further, we define VoIP flows using Mininet's Command Line Interface (CLI).

The total codec bandwidth in Mininet is calculated by considering the codec sample size and the header lengths of protocols used at all the TCP/IP layers. It is 126400 bits for G.711 and 70400 bits for G.729A. To prioritize voice packets, the Type of Service (ToS) for VoIP is set to Expedited Forwarding (EF), which is denoted by DSCP decimal 184.

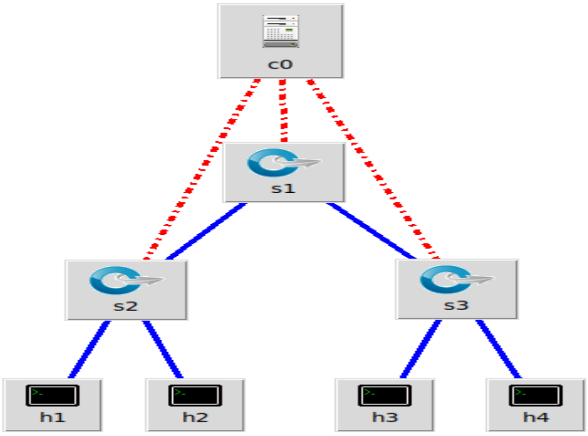

Figure 3: SDN Topology with a Controller Overhead

SIMULATION RESULTS AND ANALYSIS

A. Single VoIP Call Scenarios:

Using the network topology in fig. 2 and fig. 3, we simulated the 60 seconds VoIP communication between 2 individual hosts in all the scenarios and noted the readings at an interval of 1 second. By setting the operation mode to "Serial" for a single VoIP call scenario, fig. 4 shows the average throughputs on a conventional network using discrete event simulations in the Riverbed Modeler. It can be seen that average throughput is close to 116800 bits for G.711 and approx. 60800 bits for G.729A.

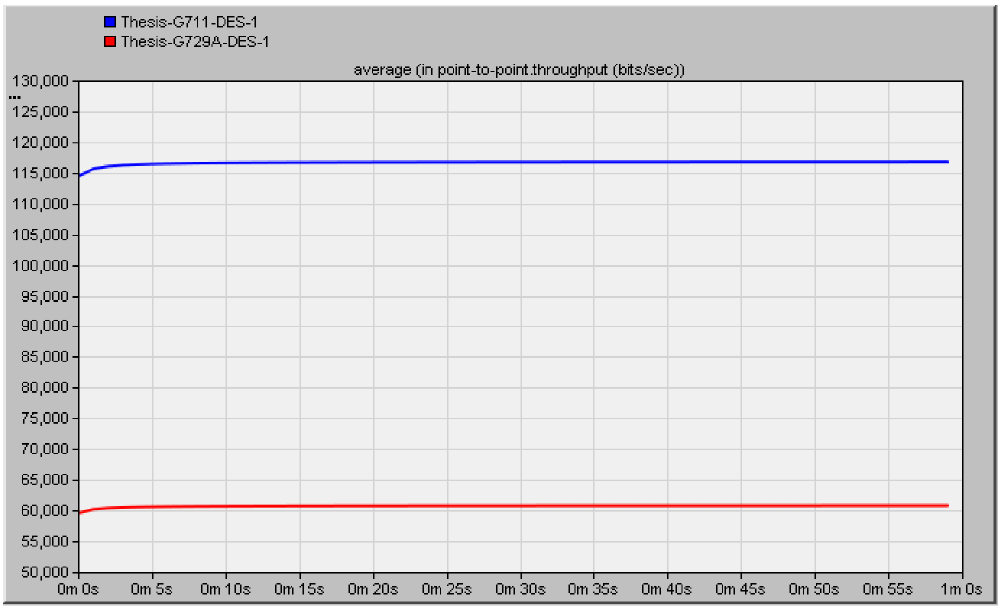

Figure 4: Single VoIP Call Average Throughput on Conventional Network

The voice payload size was calculated using the obtained throughput and packet count, and it was found to be 160 bytes in case of G.711 and 80 bytes for G.729A. By applying the same settings for SDN using Mininet CLI, the result for average throughput is shown in fig. 5.

According to the results in fig. 4 and fig. 5, VoIP over SDN is more efficient in terms of bandwidth utilization than the conventional network, with the average throughputs in SDN being 124000 bits for G.711 and 68880 bits for G.729A. This is approx. 6% and 13% higher for G.711 and G.729A, respectively, as compared to the conventional network.

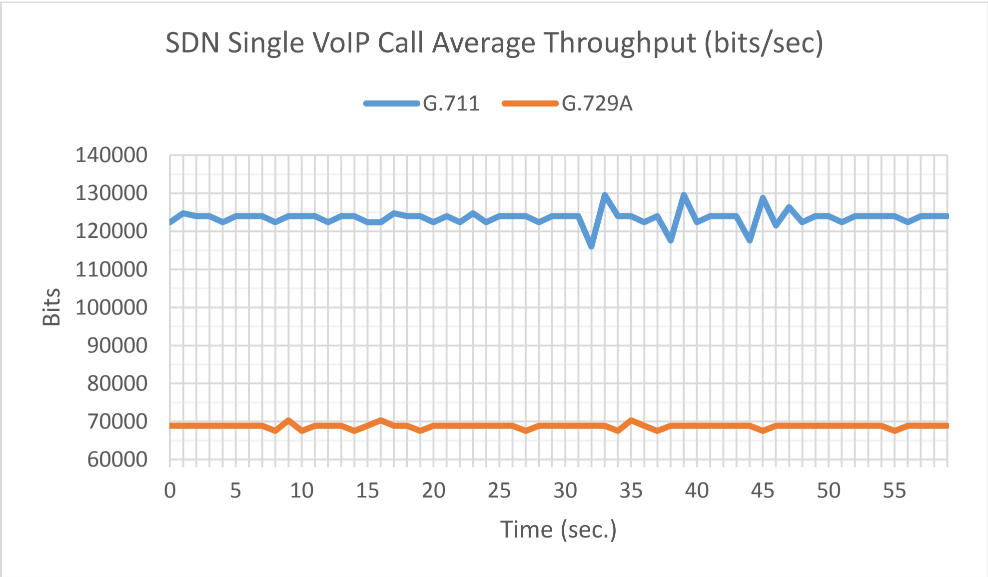

Figure 5: SDN Single VoIP Call Average Throughput

For both conventional network as well as SDN, the single call jitter values in milliseconds were found to be infinitesimally small, practically 0 in both the codecs. Similarly, there was no packet loss observed for these scenarios. This is because both the networks are capable enough to carry the prioritized voice payload generated by a single VoIP call.

From fig. 6, it can be seen that the average MOS value for a single VoIP call in conventional network is 4.36 for G.711 and 4.02 for G.729A. Similarly, MOS was calculated for a single VoIP call over SDN and found to be slightly higher than that of the conventional network, 4.4 for G.711 and 4.1 for G.729A.

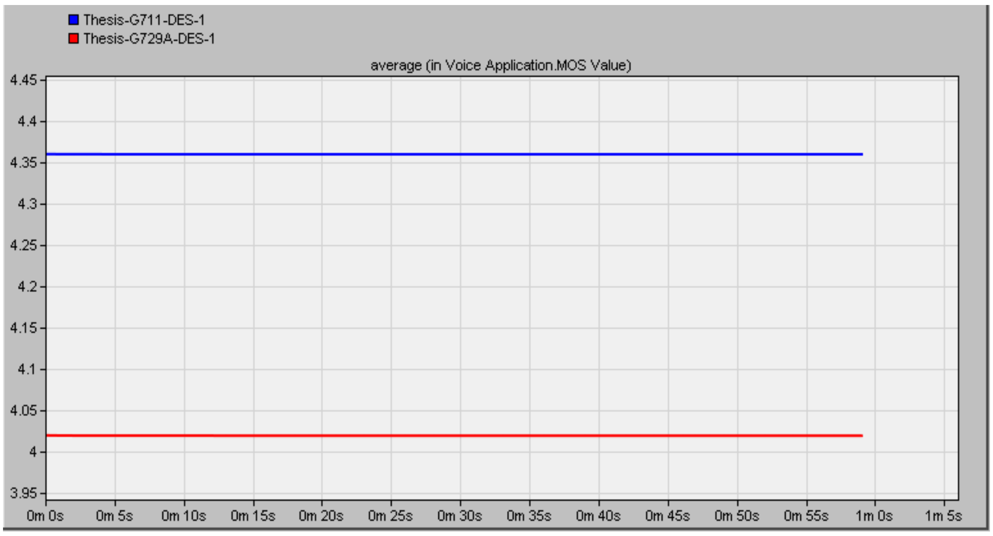

Figure 6: Single VoIP Call MOS Value on Conventional Network

B. Concurrent VoIP Calls Scenarios:

In case of concurrent calls, we simulated 33 simultaneous VoIP calls between hosts using different port numbers in order to overwhelm the network resources with prioritized voice traffic.

Fig. 7 depicts the average jitter for concurrent VoIP call scenarios in both conventional network and SDN. It is observed that the jitter performance in all the scenarios remains within acceptable limits (<30 ms) for the entire simulation time, with the exception of G.711 on conventional network. In this scenario, jitter performance degrades exponentially after 50th second. Since G.711 requires more bandwidth as compared to G.729A, it consumes the network resources faster. Because of this, there is a sharp increase in the queuing delay resulting in high jitter. However, this problem doesn't occur in SDN as it uses the available bandwidth more efficiently.

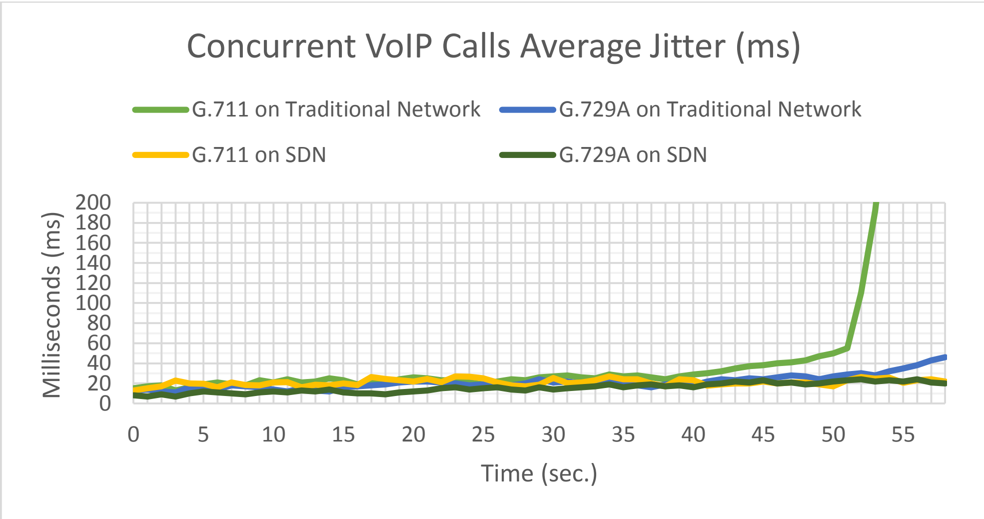

Figure 7: Concurrent VoIP Calls Average Jitter

In concurrent calls scenarios over SDN, the packet loss occurs only at the 1st second, since the Ryu SDN controller is in the learning phase. Being the brain of SDN, the controller adds new entries to its flow table so that express forwarding of packets can occur and the queue is not overwhelmed. At 1st second, G.711 observed 1.58% packet loss, whereas it's 4.08% in case of G.729A. No packets were lost after 1st second, bringing down the overall packet loss to just 0.027% and 0.07% for G.711 and G.729A, respectively.

From fig. 8, it is observed that the average MOS value for concurrent VoIP calls in conventional network remains constant at 4.02 for G.729A. However, in the case of G.711, MOS goes on decreasing exponentially from 4.36 after 50th second. Similarly, MOS was calculated for concurrent VoIP calls over SDN. Due to the controller's learning phase at the 1st second, MOS value is lower initially. However, it increases and remains constant after the 1st second. MOS for G.711 is 4.0 at the 1st second and 4.4 thereafter, whereas MOS for G.729A is 3.4 at the 1st second and 4.1 thereafter.

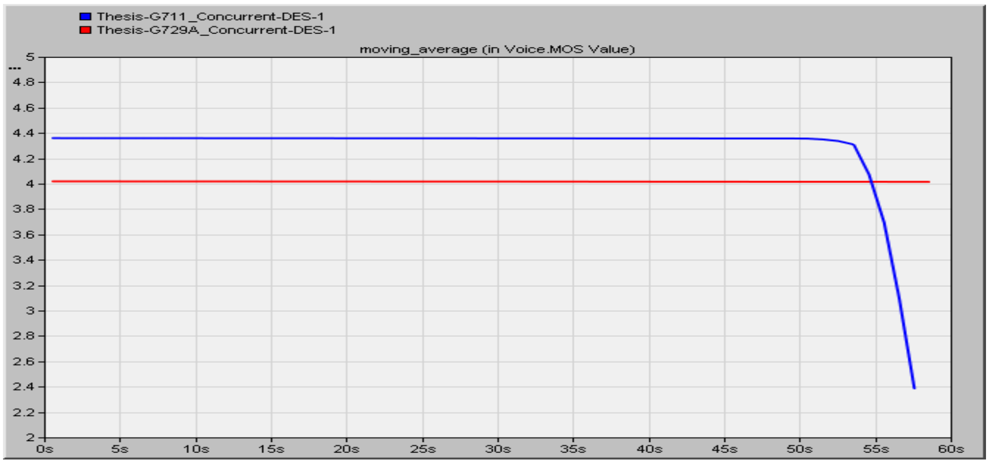

Figure 8: Concurrent VoIP Calls MOS Value on Conventional Network

CONCLUSION

SDN is a relatively new approach that improves the overall network performance by proactively designing, configuring and managing the network and corresponding applications. In this paper, we analyzed and compared the QoS performance of VoIP as a real-time application over conventional network and a similar SDN-based topology. Due to SDN's efficient bandwidth utilization, overall throughput was observed to be higher and average jitter remained within acceptable limits, as opposed to the conventional network. Furthermore, in concurrent VoIP calls scenarios, SDN effectively dealt with the congestion because of the separation of data plane and control plane, which gave the SDN controller a centralized view of the entire network architecture. Since the SDN controller was learning about the entire network, a few packets were lost at the 1st second, which accounted for all of the packets lost in the entire simulation. Not a single packet was lost thereafter. G.711 codec over SDN provided the best voice quality, which can be confirmed from the MOS value. All of these benefits of SDN resulted in the improved QoS performance of the VoIP application, when compared to the conventional network.

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