OpenFlow based VoIP QoE monitoring in Enterprise SDN

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***Abstract*—The global view enabled by a Software Defined Network (SDN) architecture allows us to observe the contribution of each congested router to the overall Mean Opinion Score (MOS) degradation of a VoIP service. Thus, when a number of routers in the path may add to the MOS degradation, the SDN controller can quantify the contribution from each router. This work presents the implementation of a network monitoring system for OpenFlow protocol aware enterprise VoIP networks. The presented solution enables network administrators and developers to access on demand information about intermediate packet loss and MOS for real-time applications. The main contribution of this work is an out-of-the-box architecture to enable users to find the exact location of quality degradation with a small operational cost.**

I. Introduction

Enterprise Voice over IP (VoIP) solutions are commonly used by large corporations and have increased in popularity among small and medium businesses in recent years particularly through the availability of cloud-based VoIP services. Within an enterprise scenario, the management and configuration of the network and the traffic is often done by an IT department with little or no expertise in the particular requirements of VoIP services. These departments are moving towards the DevOps paradigm, where service development is integrated into the operations functions of the enterprise. It is therefore interesting to explore the use of tools for VoIP Quality of Experience (QoE) monitoring to be used by such departments.

SDN provides functions to implement fine-grained network management and control, and is considered to be a key element to implement Quality of Service (QoS) and network optimization algorithms. While recent SDN protocol proposals, such as OpenFlow [1] or NetConf [2] introduce programming interfaces to enable controllers to execute complex traffic engineering, these are not currently used to implement QoE monitoring for real-time traffic.

The transfer of real-time data over the Internet and communication channels in heterogeneous packet enterprise networks is subject to errors of various types - typically the most severe impact comes from router or link congestion resulting in packet loss. The impact of transmission errors on the user perception of real-time services can be investigated by measuring certain parameters of the VoIP stream (passive monitoring) or by probing the network with a packet train (active monitoring). Active measurements should be avoided where possible due to the overhead incurred.

Given the fact that OpenFlow controllers are capable of querying network switches or routers for flow-, port-, queue-, table-, meter- and group-statistics, we propose a QoE monitoring solution to reveal the negative effects of each congested router on the overall quality. This is expressed as a MOS, which is going to be valuable in implementing QoS functions for voice, audio and video. The aim of this work is to monitor the performance of a group of calls and implement the functionality in the SDN controller. Thus, a global view of the call quality degradation can be used to calculate not only the overall MOS for a collection of calls, but reflect on the impact of each intermediate link, to locate the source of the quality degradation.

The main contributions of the paper are:

- Methods to identify VoIP quality degradation on each router or switch in enterprise VoIP environment, and per link monitoring for VoIP in non-invasive style
- Lightweight SDN monitoring solution with low communication overhead
- SDN architecture capable to integrate with real-time log analysis platforms to enable anomaly detection based on automated analysis of logs produced by SDN components.

II. Related Works

This section explores the limitations of existing commercial and open-source solutions currently used for traffic monitoring in SDN, such as OpenNetMon [3], SQprobe [4] and NetFlow [5].

OpenNetMon [3] is capable of measuring the latency by injecting probe packets, however it does not feature traffic matching functions and hence cannot isolate real-time traffic. Furthermore, it only returns overall metric, by installing monitoring flows, creating additional overhead.

SQprobe [4] uses passive and active monitoring combined. For passive monitoring the architecture includes agents embedded into endpoints, distributed probes and other mid-stream devices. For active testing, distributed software test agents generate voice, video, and network diagnostic traffic and report quality/performance metrics back to a central reporting and configuration interface. This solution results in an end-to-end

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view, which is limited for accurate troubleshooting of quality degradation problems.

NetFlow [5] is a widely used and recognised scalable network management tool. Passive measurement imposes low overhead to the network, however it requires full access to network devices, which raises security and privacy issues. Netflow typically has 5 minute granularity, and although NetFlow produces accurate long-term statistics, it is too coarse to be used in proactive network reconfiguration strategies.

Another approach [6] has been used to determine VoIP QoS at intermediate nodes in a SDN network. Each OpenFlow switch in the network must be iMOS-enabled in order to compute the metric, which is useful when localised reconfiguration or prioritization is required, but does not provide the global view that is afforded by the SDN controller.

III. System Design

A. Quality of Experience

Parametric model approaches such as the E-model [7] or PESQ [8] are well suited to monitoring live VoIP sessions [9]. These approaches measure the speech quality by monitoring the flow of voice packets without access to speech signals. The E-model implements a mathematical model that combines all the impairment factors that affect the voice quality in a single metric called the R value that can then be mapped to the MOS score. R values are in the range of 0 to 100, where R = 0 represents the worst quality and R = 100 represents the best quality. In the original form defined by International Telecommunication Union (ITU) R value is calculated as: $R = R0 - Is - Id - Ie + A \quad (1)$ where $R0$ is the Signal to Noise ratio (S/N) at 0 dBr point, Is represents the speech voice impairments, Id is the impairments occurred due to the delay, Ie is the impairment due to the equipment (e.g. codec and packet loss) and A is the advantage factor (e.g. A = 0 for wireline).

In enterprise network scenario, Equation (1) can be reduced to: $R = 93.4 - Ie(codec, loss) \quad (2)$ since delay impairment in enterprise networks is much less than 100 ms, hence it has no impact on the overall R calculation value. Enterprise network loss mainly happens due to congestion on one or more routers output buffer overflow due to link capacity limits, which is the scenario emulated in our experimental network.

B. OpenFlow Metric

The monitoring task is accomplished by the SDN controller which is connected to all the switches via a secure channel, and its functions are enhanced by a monitoring module that collects real-time flow statistics and calculates end-to-end packet loss and per link packet loss.

In the signaling between the controller and the switches, OpenFlow $Read - State$ messages are used by the controller to collect various information from the switch, such as current configuration, statistics and capabilities. Our solution proposes the use of those messages to gain information about network traffic.

In order for the SDN controller to configure and manage the network, it needs to be able to retrieve accurate information about the state of the network, in particular its topology. As a result topology discovery functions are critical for each controller. Once all active switches are discovered by the controller, the controller starts to query for aggregate statistics every P seconds.

The proposed controller module uses the packet count from each of the OpenFlow enabled devices in the network in order to calculate link packet loss rates. Specifically, the one way packet loss rate for a link is calculated by taking the aggregate flow packet count on the source and destination at time t and subtracting these from the aggregate flow packet counts at time $t + P$. The difference in the packets processed at each node, therefore yields the packet loss rate which is typically due to a buffer overflow at the sender's outbound interface.

C. Real-time Log Analysis Engine Integration

Proposed solution is integrated with real-time log analysis engine in order to create custom tags for spotting important events. The log analysis platform of choice in current testing environment is Logentries by Rapid7 [10]. The proposed system integrates with Logentries real-time log analysis platform, not only for the benefit of real-time alerts and notifications, but also it acts as a secured cloud database that stores historical MOS and packet loss data for further analysis.

IV. Test scenario

A typical application of VoIP is for a large corporation to use its existing Wide Area Network (WAN) infrastructure for data traffic to carry voice calls between its headquarters and its branch offices, possibly including a distributed call centre. Such WANs are often not well-provisioned to serve voice traffic, even after a transition to SDN architecture and are usually the weakest links [11]. The aim of these experiment is to identify those degraded links that may arise as network traffic scales up.

In order to meet emulation requirements, the network functions including traffic end points, routers and links are emulated in MiniNet [12]. The SDN functions are implemented using an open-source POX controller.

A. Packet Loss Modeling

The network impairment that has the greatest effect on QoE in real-time applications, such as voice or video is packet loss [13]. A previous work [14] in the area of packet loss modeling in IP networks used a Markov process approach for traces of traffic and packet loss processes. The results show that simple Markov models are appropriate to capture the observed loss pattern. In this work we apply 2-state Gilbert-Elliot model, extensively used for defining error patterns in transmission channels. Both states may generate errors as independent events at a state dependent error rate $1 - k$ in the good and $1 - h$ in the bad state, respectively.

In these tests, multiple VoIP calls are initiated between Host 1 and Host 10, with the VoIP traffic generated using

Linphone [15], an open-source VoIP software. The links used are depicted in Figure 1. In order to emulate a realistic network traffic scenario, background traffic was injected using iPerf. It is a bidirectional TCP traffic sent between Host 1 and Host 10 at rate 100 packets/sec 40bytes/packet. This traffic was injected mainly to present the capabilities of proposed monitoring system to isolate VoIP traffic statistics.

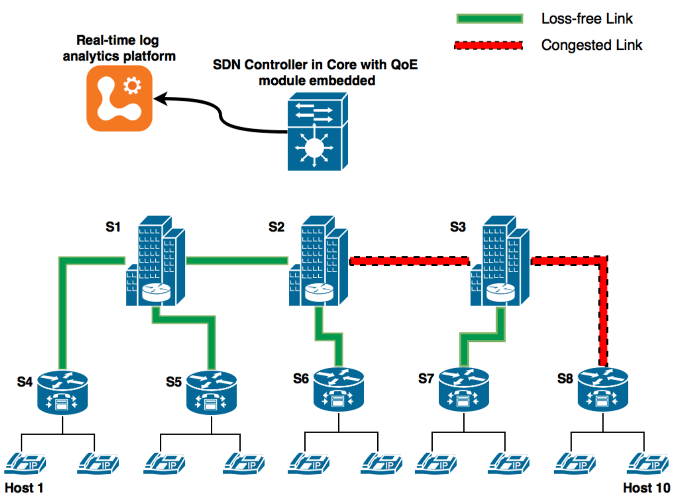

Fig. 1. Links S4–S1–S2–S3–S8 are considered to be the communication path, where S3–S8 and S2–3 are congested for testing purposes

The congestion on links S3–S8, S2–S3 is introduced to emulate buffer overflow on the routers S3 and S2 using constant loss model and bursty loss modeling technique described above. Both tests are executed using following simulation steps:

1) Simulation starts with 0% loss on all routers
2) 2% loss is introduced on S3 router
3) 1% loss is introduced on S2 router
4) Cool-down period with 0% loss on all routers

V. Results

We investigate the ability for the proposed SDN controller-based monitoring technique to report intermediate and end-to-end MOS for a given set of calls. The proposed solution achieves that by calculating accurate loss rates and using the simplified E-model detailed in Section III-A.

A. Constant Loss

The intermediate MOS is calculated using the simplified E-model with previously measured loss values. The graph in Figure 2 presents the intermediate MOS aggregated for all the calls that were monitored as well as end-to-end MOS, where end-to-end MOS does not include host-to-switch links that are beyond SDN architecture visibility. What has not been achieved prior to this work, is the functionality of viewing quality degradation contribution on each intermediate link. Referring again to the results presented in Figure 2, we can observe excellent MOS score on links S4–S1 and S1–S2 for the duration of the experiment, due to no loss on those links. Nevertheless, we observe that a small degradation in quality on more than one of the links may have significant impact on the overall call quality.

Finally, to evaluate the correctness of MOS calculation we plot the call quality reported from Linphone on the graph in Figure 2. Packet Loss Concealment (PLC) is implemented in Linphone, as a countermeasure to the negative effect of loss on overall call quality, hence the presented Linphone rating is slightly higher, yet MOS score degradation and improvement is observed at the same time intervals throughout the total duration of the simulation.

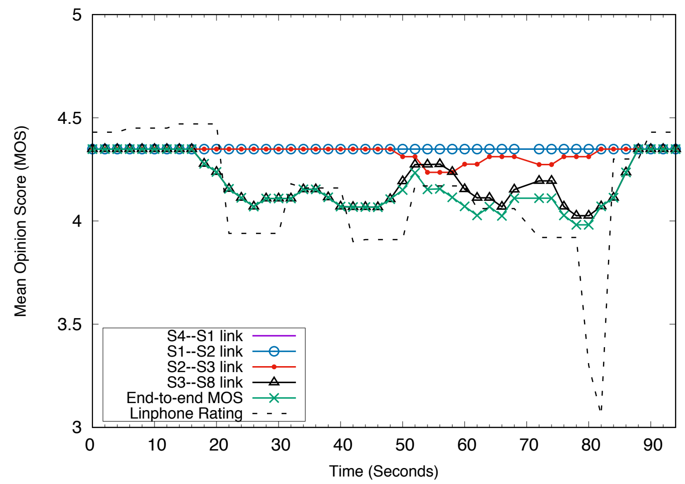

Fig. 2. Intermediate MOS calculation for constant 2% loss on S3 router, followed by 1% loss on S2 router using the proposed solution

B. Bursty Loss

In order to simulate a more realistic packet loss scenario, we use the bursty packet loss modeling approach described in Section IV-A.

The intermediate MOS calculation in this bursty loss scenario is depicted in Figure 3. Looking at these results from DevOps perspective we can observe time intervals, where MOS drops down to 3.7, and the links where the quality degradation occurs are immediately exposed.

C. Intermediate Call Quality Degradation

MOS based quality assessment is carried out for a much longer call duration, where results are depicted in the graph in Figure 4. Loss rates introduced this time are 2% on link S2–S3 and 1% on S3–S8, which makes those two links two of the biggest contributors to MOS degradation in this test scenario.

The results in Figure 4 can be visually exposed to a DevOps person, for example similar to Figure 5, where MOS distributions for each link in the network are displayed along with the network topology. The health of individual network links can be evaluated by referring to pie charts [16], [17] assigned to each link. The aim of experiments is to show that even small loss can be measured and mapped onto MOS scale, but more importantly, the contribution of each congested link to the total MOS score can be exposed in this solution.

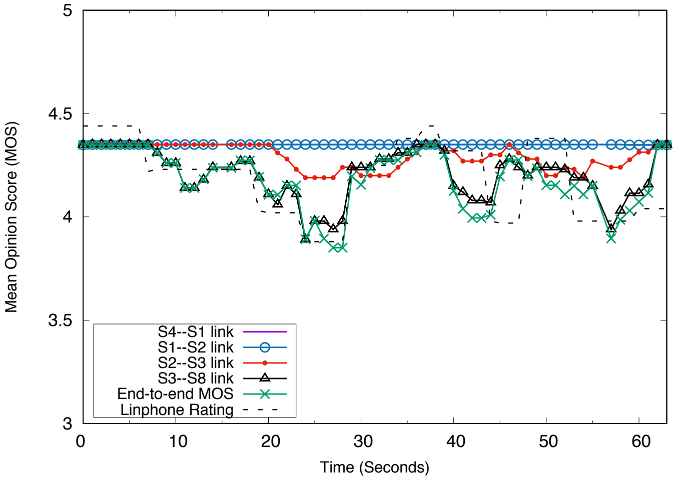

Fig. 3. Intermediate MOS calculation for bursty loss on S3 router, followed by bursty loss on S2 router using the proposed solution

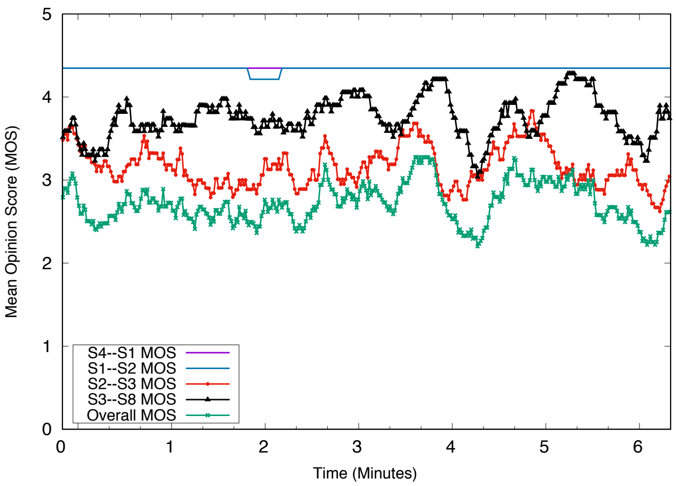

Fig. 4. MOS evaluation for a 6 minute call. Network links S2–S3 and S3–S8 contribute to overall low call quality

VI. Conclusion

The proposed monitoring system delivers out-of-the-box methods to identify quality degradation on each link in enterprise VoIP environments. The most significant contribution is the ability to perform per-link monitoring for VoIP in non-invasive style. We show results that validate the correctness of the packet loss measurements with other existing solutions, such as iPerf and Linphone. This work uses the global network view enabled by the SDN, so if the degradation of quality has multiple contributions, network administrators can see in real-time, where those links are located. The proposed solution outperforms other end-to-end solutions, because it provides the user with the view at each intermediate link in the communication path.

Acknowledgment

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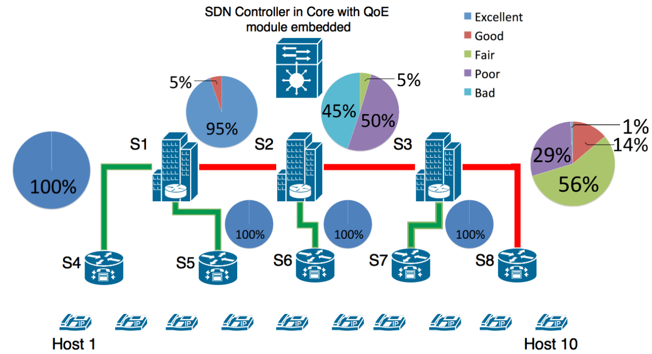

Fig. 5. MOS based quality assessment for every link in the network based on MOS calculation results presented in Figure 10

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