OpenSIP for Enterprise Networks

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Abstract: Nowadays, communication industries are developing for a large enterprise. One issue is known that the load balance is used in communication technology using Internet protocol (VoIP technology). In this paper, we discuss the implementation of a communication system VoIP technology running with OpenSip. The simulated results show that the application of OpenSip is able to build a unified communications network, satisfy requirements and adapt to many enterprises.

1. Introduction

Enterprises often have branch offices that are exchanging information and using VoIP systems to provide internal telephone services to departments in their company such as Asterisk technology. However, because of the growth in the number of employees, the demand for voice communication is enormous, while Asterisk PBX is experiencing hardware problems. Therefore, it makes the system wasted communication resources. On the other hand, OpenSIP to upgrade the SIP network framework with emerging technologies, such as SDN and NFV. SDN provides effective routing and resource management by decoupling the data and control planes along with a software-based centralized control. Moreover, NFV assists SDN by virtualizing various network devices and functions. To achieve load balancing amongst SIP proxies using SDN technology, the author [2] moves the SIP proxy functionalities from the data to the control plane. In doing so, the data plane is eradicated from the SIP proxy equipment and the complications involved in management. Finally, the author [2] find to virtualize the SIP proxies in such a way that SDN manages the virtual proxy infrastructure by employing the Network Function Virtualization (NFV) concept [2]. To implement a communications system VoIP technology applications with OpenSip, this paper proposed a model with communication system based on internet protocol. An open source system support network infrastructure and more powerful. It meets the requirements of the company's data processing capabilities (load balancing). The objective of this paper is to build a VoIP system used in enterprise networks. We use OpenSip which is known as an open source built perform the work of a SIP server. This system can handle thousands of calls with lower cost, high reliability, unified management, routing as well as efficient load balancing.

2. System Model

Our models were built VoIP system which achieve the following requirements: i) stable running on available equipment, ii) ensure bandwidth for the entire network. To oblige the communications in internal company, we can use an internal switchboard which is fully able to meet the demands of the business such as Asterisk ... But to provide better routing and managing a large number of users. There are main majors is uses, such as providing: voice traffic, routing and efficient load balancing. We choose OpenSip is an open source built perform the work of a SIP server can handle thousands of calls with high reliability. SIP messages are processed quickly and customized according to the user's desired along with the integration of the module. The Figure 1 specifies the network model with current system. It includes two main sites A and B. My model focuses on site A. The infrastructure of the VoIP network uses the available company's network and hardware. This makes the cost increasing slightly. SIP protocol is chosen for the system, because this is a standard protocol of the IETF. VoIP server is shared server with the available server of the company, which was set up a virtual server running the Linux operating system. VoIP Control Software is open source software, OpenSip, has similar features as a switchboard, installed on the available server at the company. Thus, a completed SIP Network Server system integrated in available server as shown in Figure 2.

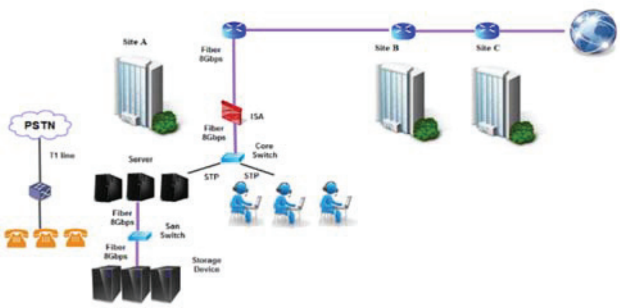

Figure 1: The core network diagram.

VoIP system has been integrated into a current LAN system. All the computers and devices with VoIP network address belongs to192.168.1.x to exploit this system. We initialized VoIP telephones and mobile phones to communicate between departments. This advantage in our system is: users can use the mobile devices or softphone which have send text messages, video call each other and accompanied by images.

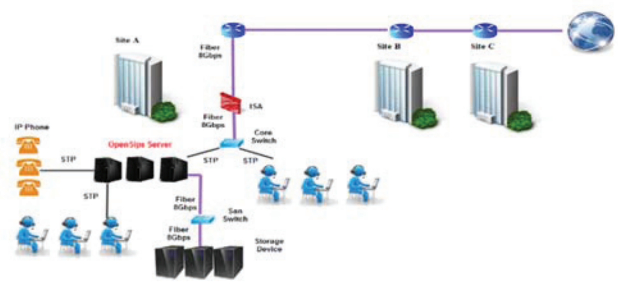

Figure 2: Network diagram with OpenSip.

This service enables the inside network to call each other as well as the previous VoIP Systems of the company. Plus, the new VoIP System is optimized for the using of the company. It also increases the effective of network bandwidth for the phone call over network.

The Session Initiation Protocol (SIP) is a communications protocol for signalling and controlling multimedia communication sessions in applications of Internet for voice and video calls, in private IP systems, as well as in instant messaging over Internet Protocol networks. SIP works in concert with these protocols by enabling Internet endpoints (called user agents) to discover one another and to agree on a characterization of a session they would like to share. For locating prospective session participants, and for other functions, SIP enables the creation of an infrastructure of network hosts (called proxy servers) to which user agents can send registrations, invitations to sessions, and other requests. SIP is an agile, general-purpose tool for creating, modifying, and terminating sessions that works independently of underlying transport protocols and without dependency on the type of session that is being established (RFC 3261).

3. Simulation results

To solve the OpenSip system in operation network model of company we launched two major models (test case) as follows in scenario 1, the user only makes calls to each other and registers account to the OpenSip server. After account is enabled, the client will conduct another test call via the company's server OpenSip. The default bandwidth using to connection is totally 16Kbps (15Kbps for Codec and 1Kbps for Signalling). The bandwidth is used for the VoIP system in company is about 10Mbps. We see that the highest bandwidth is 8000Kbps in Figure 3. During the call, bandwidth always is at an average of 1700Kbps. And there are only 2 points of bandwidth increased high suddenly. The reasons are that the registration of many users to the server at the same time and the network is not optimized for VoIP. We observe with more than 100 connections, the total bandwidth occupied around 1700Kbps. That is satisfy the requirement.

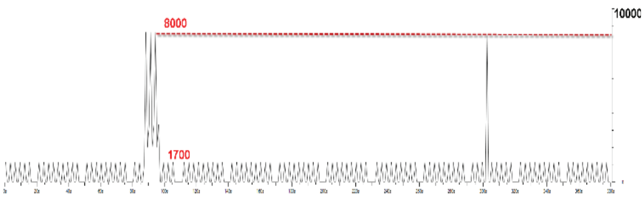

Figure 3: Signal when making calls with voice

In scenario 2, we design a hybrid network including phone, PC, laptop, mobile device simultaneously. In this model, the VoIP network will simultaneously run multiple services Such as: calls, video call, message to create hundreds of connections. The we checked the performance of the network and load capacity of OpenSip.

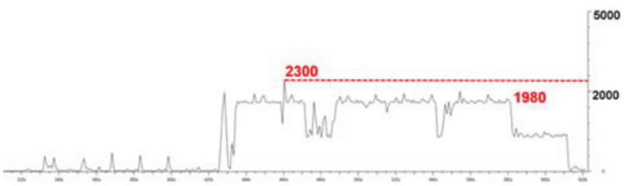

Figure 4: Bandwidth was used between mobiles

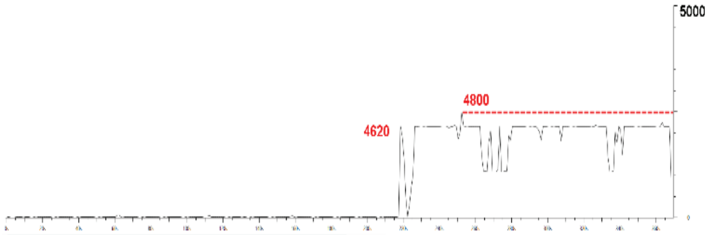

Figure 5: Bandwidth was used between VoIP phone (video call)

We run with approximately 30 pairs and 70 pairs of devices. Consequently, once only one type of equipment is used, there is high stability.

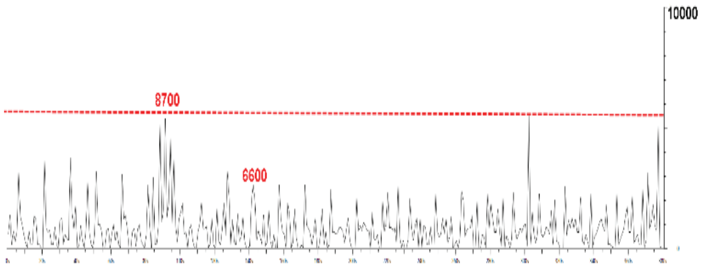

Figure 6: Bandwidth was used between mobiles and other devices (video call)

I simulated with approximately 100 pairs of variable devices. However, it does not affect call quality and bandwidth is guaranteed.

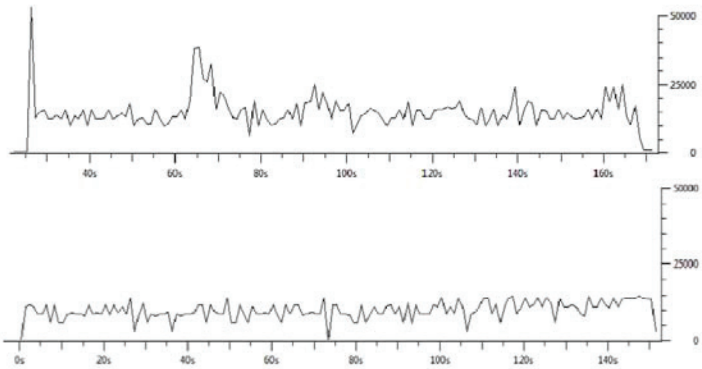

Figure 7: Compare bandwidth was used the propose model.

The figure shows that there is the ability to stabilize when the number of calls is much higher. However, this delay is smaller in the proposed model than in the case using the current infrastructure, which demonstrates the efficiency of our proposed model

The Figure 3,4,5,6,7 demonstrate optimization of OpenSip satisfy company requirements and voice call services. However, the video-call service is not full supported because of low bandwidth and unoptimized for video-call service.

4. Conclusion

In this paper, we discuss the technology OpenSip, employing to the current networks. This system can provide internal calls Services for VoIP networks such as: lower cost, high reliability, unified management, routing as well as efficient load balancing. The simulation results show that all OpenSip implementations achieve a high throughput, low response time, and resource efficiency.

5. Acknowledgement

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6. References:

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