Performance Analysis of Session Initiation Protocol on Emulation Network using NIST NET

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Abstract - Protocol testing for various real world applications can be analyzed using several methods, i.e. measurement, simulation and emulation. The first method from several method is live network testing. The second method is network simulation (simulation using Network Simulator) and the last method is emulation method (Emulab Testbed, DETER Testbed, and NIST Net). Emulation Method in network is the method which combines real elements of network application that has been deployed. For example an end host and protocol implementation, with synthetic elements, simulated or abstracted, such as network link, intermediate node, and background traffic.

This paper shows the result of our experiment and testing of a Voice over Internet Protocol (VoIP) application using Session Initiation Protocol (SIP). VoIP testing is conducted using an open source network emulation software tool called NIST Net. VoIP which use SIP can be deployed in a real network. The network parameters used in our experiments are bandwidth and drop packets. The result shows the bandwidth and jitter characteristic during the experiment. It can be summarized that the quality of VoIP depends on bandwidth, jitter, and drop packet.

Key words – SIP, NIST NET, VoIP, Emulation.

1. Introduction

The development of technology in computer network with all related applications encouraged network computer scientist to increase the reliability of existeng computer network that had already exist, in order to optimize the network performance.

Therefore specific method is needed to build and analyze the performance of an application deployed in a particular network. Subsequently, the results of the experiment can be analyzed and used to improve the performance of the existing computer network to support the various applications being deployed.

There are three methods which can be used by scientists to test an application in specific network computer, i.e live testing network, simulation network method, and emulation network method.

Each method has some strengths and weaknesses which complement each other. The implementation and deployment of those methods depend on resources, supporting infrastructures, and requirements.

Network simulation method is a method which provides the condition in which network element that can be controlled. Network condition can be repeatable for several experiments in the network. With simulation, it can be easier to control and make protocol in several abstract levels. On the other hand, emulation method can not be made repeatable in other experiments, as it is conducted in a real time situation.

2. Basic Theory

Presently, in a network testing we can perform three methods namely measurement, simulation and emulation. Table 1 compares the three methods. Modelnet is an emulation method that uses open source software which runs in the UNIX system; in this case we use FreeBSD or Debian. Meanwhile, Emulab is the testbed system which is established and maintained by the University of Utah. Using this testbed user only had an account and subsequently can use the provided network resources which have already provided in the testbed to do some experiments.

Emulab integrates simulation, emulation, and live network experiment into general framework which provide integrated abstraction, services, and another utilities from relevant environment, such as node and path allocation and also naming and mapping all of that into specific domain mechanism. Meanwhile DETER Testbed has special condition and working procedure similar with Emulab, because this DETER Testbed project is related to Emulab [1].

3. NIST Net

NIST Net is an open source software developed by National Institute for Science and Technology Network (NIST Net). NIST Net is a software which enable emulation method and run in LINUX operating system environment [2].
The working procedure of NIST Net can be made analogous to a one loop network with NIST Net inside the box as depicted in Figure 1. NIST Net can emulate traffic which becomes IP packet that come and go through NIST Net. NIST Net can make some changes into passing packets. It can also generate some impacts in to the network characteristic.

NIST Net enables the user to make some changes in general parameters in network, such as delay packet, bandwidth limitation, drop packet, and duplication packet. If the parameter has already been chosen, then NIST Net will marked the IP packet that going through, drop the packet or filter it. Everything depends on what parameter that has been changed by user [3, 4, 5, 6, 7].

4. Methodology and System Design

In this experiment we conducted two scenarios, which are point-to-point and multipoint connections. The topology of the system for point-to-point and multipoint is depicted in Figure 2 and Figure 3.

In our experiment we run SIPP at client 1 and client 2 (for point-point). For multipoint connection we added one more client, namely client 3. The experiment results represent the characteristic of the performance of VoIP application and signaling system using SIP in this system [8, 9, 10, 11, 12, 13].

5. Performance Analysis of VoIP in Point-to-Point Connections

The results of our experiments are depicted in the graphic results which show the jitter and bandwidth characteristic. Our experiments can be divided into two conditions, i.e. point-to-point and multipoint. For each experiment we change the parameters, i.e. bandwidth and packet drop [14, 15, 16].

- Bandwidth for Point-to-point connection.
Result of bandwidth characteristic versus Jitter experiment in client 1 can be found in Figure 4.

Result of bandwidth characteristic versus Jitter (1st data) experiment in client 2 can be found in Figure 5.
Result of bandwidth characteristic versus Jitter experiment in client 2 (2nd until 5th data) can be found in Figure 6.

- Drop packet for Point-to-point connection.
  Result of Drop Packet characteristic versus Jitter (1st, 2nd, 3rd, and 5th data) experiment in client 1 can be found in Figure 7.

- Drop packet for Point-to-point connection.
  Result of Drop Packet characteristic versus Jitter (4th data) experiment in client 1 can be found in Figure 8.

Result of Drop Packet characteristic versus Jitter experiment in client 2 can be found in Figure 9.

6. Performance Analysis of VoIP in Multipoint Connections

- Bandwidth for Multipoint connection.
  Result of bandwidth characteristic versus Jitter experiment in client 1 (1st, 2nd, 3rd, and 4th data) can be found in Figure 10.

Result of bandwidth characteristic versus Jitter experiment in client 1 (5th data) can be found in Figure 11.
Result of bandwidth characteristic versus Jitter experiment in client 2 is depicted in Figure 12.

Result of bandwidth characteristic versus Jitter (1\textsuperscript{st}, 2\textsuperscript{nd}, 3\textsuperscript{rd}, and 4\textsuperscript{th} data) experiment in client 3 can be found in Figure 13.

Figure 14 shows the result of bandwidth characteristic versus Jitter (5\textsuperscript{th} data) experiment in client 3.

- Drop packet for Multipoint connection.

Result of Drop Packet characteristic versus Jitter experiment in client 1 can be found in Figure 15.

Result of Drop Packet characteristic versus Jitter experiment in client 2 can be found in Figure 16.

Result of Drop Packet characteristic versus Jitter experiment in client 3 can be seen in Figure 17.

The following list shows the summary of our experiments:

- Bandwidth is one of the most significant parameter which affected the quality of VoIP. These have been shown in Figure 4, 5, 6, 10, 11, 12, 13, and 14.
- Bandwidth decreased will raise the number of delayed packet. Jitter will also be increased and the quality of VoIP will be poorer.
- The characteristic of bandwidth versus jitter and the quality of VoIP is the same in both point-to-point and in multipoint connections. These have been shown in Figure 4, 5, 6, 10, 11, 12, 13, and 14.
In varying the number of drop packet and no bandwidth limitation; the quality of VoIP depends on the number existing drop packet in a real time transmission. These have been shown in Figure 7, 8, 9, 15,16, and 17.

Jitter depends mostly on the available bandwidth during the transmission. The percentage of drop packet during the transmission is the most significant parameter in this condition.

In some cases, the shows a very large jitter result, preventing the establishment of the VoIP session. This is caused by the SIP server which is not yet ready to start a new session of signaling transmission for a new VoIP call. It can also be the result of the collision and congestion of packets on the transmission line. These have been shown in Figure 5, 8, 11, 12, 13, and 14.

The drop packet in transmission line which can be controlled by NIST Net is a random drop packet, therefore the jitter result will be varied with random quantity. These have been shown in Figure 7, 9, 15, 16 and 17.

7. Conclusion

From our experiments on SIP using Nist NET, we can summarise as follows:

VoIP applications depend on the existing and available bandwidth in the transmission line.

In both point-to-point and multipoint connections, jitter will increase with the decreased of bandwidth in the transmission line. The VoIP quality would be better when the bandwidth increased. The quality of the voice mostly depends on the quantity of drop packet in transmission line (when the drop packet is varied).

The minimum bandwidth required to run VoIP application using SIP is more than 3500 bytes/sec.

Jitters which exceed 10 ms can make more impact to the quality of VoIP.

REFERENCES


