Analysis of Voice Over IP System using SIP Protocol

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Abstract— This research is intended to study a method to analyze SIP Protocol network operations of a telephone system for organizations. This aims to build up VoIP technology and operation principle backgrounds as well as be able to use as an practical application that enhance network's component values in the organizations for efficiency optimization use.

Organizations use telephone's technologies as an application to lower internal telephone call cost and other factors cost. The results show that one can increase both channel and format of communications. For the new communications, these offer fast responses in addition with lower operation investment cost.

Keywords-VoIP; Asterisk; SIP Protocol; Open Source; Network Analysis

I. INTRODUCTION

Currently, communication and telephone systems are one of the key factors used for companies and organizations and are focused in the term of how to reducing the cost for both local and oversea calls. Many organizations then have considered this as one of the main issues and brought up VoIP's technology to lower telephone calls and faxing cost up to 90% or more which depend upon basic organization's structures connected to the communication systems.

Applying VoIP's technology to telephone systems of these organizations is to achieve benefits and lowing organization costs. This is done by using principles and low components cost those results in more values of existing network components to maximize benefits than the typical one. As before, Router and Switch have been used to receive and send data only. As VoIP is introduced, sound signals can be added as the other option. In practical, VoIP's technology can work out for organizations and companies in managing telephone numbers as required.

II. GOALS

1) To learn operations of telephone systems through SIP Protocol network.

2) To develop telephone systems through SIP Protocol network.

3) To analyze operations of telephone system through SIP Protocol network.

III. RELEVANT THEORIES

A. Voice over Internet Protocol Technology (VOIP)

VoIP (Voice over IP) is a new technology for calling through an internet network with lower calling cost both local or oversea which is obviously the top benefit. This can be achieved because by calling through an internet, there is no connection through a telephone line needed or no extra service cost required. Calling through an internet uses the same processes as sending data through as internet as shown in Fig. 1.

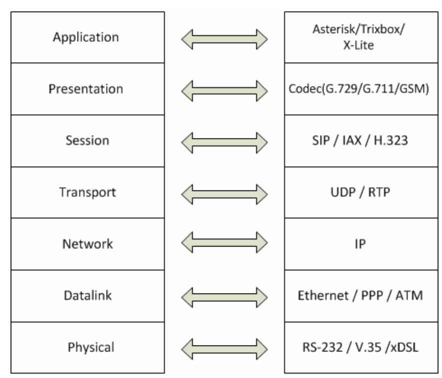

Figure. 1: Comparing OSI Model with VoIP Operation

B. Session Initiation Protocol (SIP)

Session Initiation Protocol (SIP) is a standard protocol of Internet Engineering Task Force (IETF) to start user session. Interactive related to multimedia set such as video, sound, chat, games and virtual reality SIP is the same as HTTP or SMTP because the operation in Application layer of communication models, Open Systems Interconnection(OSI) by Application layer is a level of responses to ensure the possibility of communications. SIP can initiate multimedia session or Internet telephone call and modify or terminate. Also, this protocol can invite participants to be uncast or multicast session that are not necessary concerning the initiation. Because SIP supports name matching and redirection services, this makes it possible for users to start and to receive communications and services from all locations. And for a network specifying any users, SIP will be a protocol that responding requests related to requests from a sub network and responded from the main network. Users identified by SIP URL, any requests can be transferred through transferring protocols such as UDP, SCTP or TCP. This SIP decides to terminate the system applied to a session of communications and parameters mean and calls required participants into communications. As all are insured, SIP will initiate all parameters at the end of communications and control transferring and terminate protocol components. SIP can be divided into two components: User Agents or UA and SIP Server.

IV. Equipments and Tools for the Research

A. *FreeBSD Operation System*

FreeBSD (Free Berkeley Software Distribution) is a high level operation system that has capabilities working on variety of architectures, for example X86(Intel), DEC Alpha, Sun Spars and PPC (PowerPC), originally from BSD Unix of university of California, Berkeley. FreeBSD has many outstanding features including ability to manage high level network, high efficiency, system safety and compatible with different computer architectures while other operation systems cannot offer. FreeBSD can be used to construct Internet and Intranet server as shown in Fig.2.

Figure. 2: FreeBSD Operation System

B. *Asterisk*

Asterisk is a Free Open source that supports VoIP by acting as a terminal of telephone exchange for VoIP system and also supports PSTN system from managing internal numbers used to Dial plan in the organizations. Asterisk is a tool used to support connections between a telephone line and computers with telephone receiving/sending ports. Asterisk is complete IP-PBX telephone system software that works with many operation systems such as Linux, Max OSX, Opens, FreeBSD and Sun Solaris and has prepared high quality operating functions of private branch exchange (PBX) in it. Asterisk supports VoIP (Voice over IP) with many protocols such as SIP, H323, LAX, MGCP, and SCCP (Cisco Skinny) that support standard tools and require low cost hardware as shown in Fig. 3.

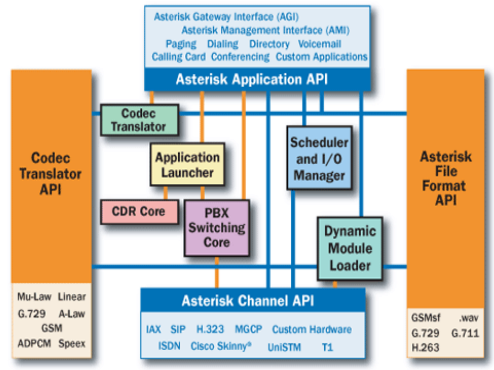

Figure. 3: Asterisk Architecture

C. *Wire shark*

Wire shark or ethereal is a network protocol analyzer program that analyzes network systems. Normally, if we use a computer to open general websites or to connect to other computers, no protocols will be shown. If protocols are required to show to learn how to create a system or Hack (not recommend), protocols detection program is become necessary. The operation of wire shark is done by detecting messages sent between protocols and we will call this as Packet sniffer.

V. Test Procedures

A. *Analyze and Design Systems*

1) Components that work together in the system are divided into two parts as following

a) Hardware

- Exchange branch send telephone signals to Server IP-PBX or FXS Server IP-PBX.
- Switch makes a computer connected to available ports to be able sending data at the same time.
- Router is used to define or select a route to send the next package. Inside this Router, there is information about managing a route for any packages.
- Two computers.

b) Software

- FreeBSD operation system Version 6.2 manages system resources.
- Asterisk Version 1.4 acting as exchange branch is a program to control all VoIP telephone system operations. This program can even add, remove, modify, and define the authorities of system operations.

- X-Lite acts as an IP Phone and be able to installed into a computer and used as IP Phone right away.
- Wire shark analyzes Network systems and works on both UNIX and Windows operation systems as shown in Fig.4.

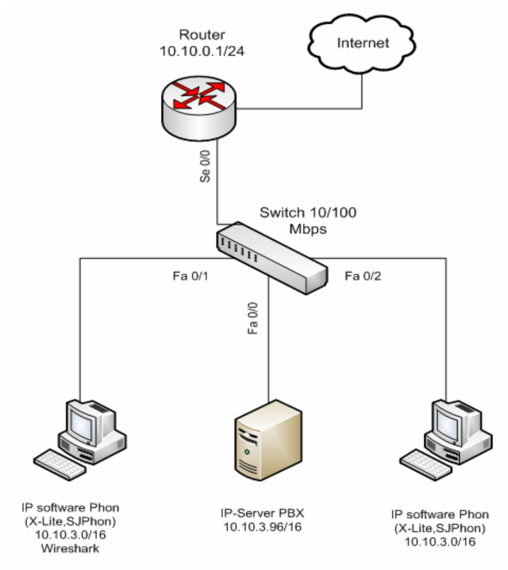

Figure. 4: Telephone System Network Diagram

B. *Working Steps*

Start working on a Asterisk telephone system can be done the same as running another operation system, even FreeBSD operation system is also in the similar manners. The command used is #asterisk-r, then Asterisk telephone system operating window will pop up in CLI (Command Line Interface) mode as shown in Fig. 5.

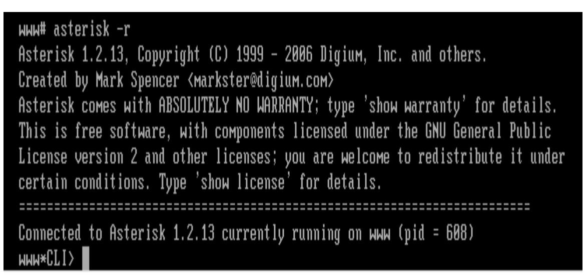

Figure. 5: Asterisk Telephone System Operation

C. *Telephone Call Record Monitoring*

Asterisk Call Detail Record or Asterisk CDR for short is a feature used to record calls detail in the form of file format .cave and can be recalled as in Fig. 6.

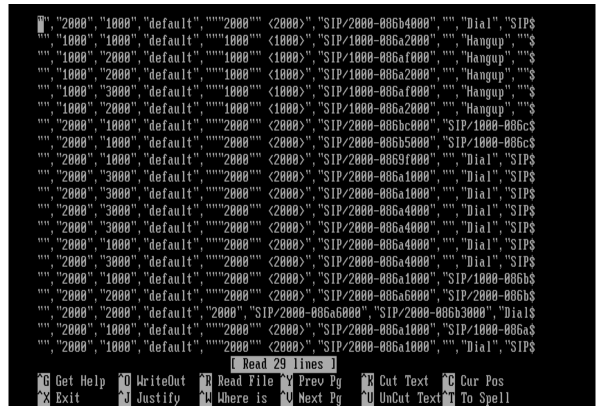

Figure. 6: Calls Record Details.

D. *Wire Shark Package Detection*

Wire shark detects records received and sent through port 21 of STP that operates from terminal to destination through IP address 10.10.3.96 connected to destination at IP address 10.10.3.42 as shown in Fig. 7.

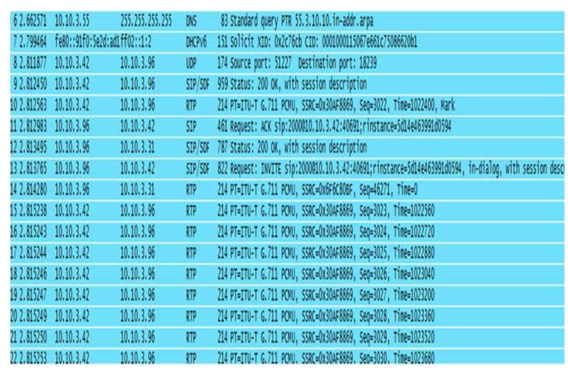

Figure. 7: Wire Shark Package Detection

VI. CONCLUSION

This research is provided to show system operations and system structures. The work was tested under defined scopes and found that the system can operate with efficiencies and work for the real applications in addition with lower organization costs under all defined project scopes mentioned. Also, the system has all defined system requirement features and deep understands to develop systems. Currently, VoIP telephone system is gaining interest because its stability, its response to users and system developer with efficiencies of system operation detection using Wire shark program to verify how systems connected, what IP's are going through, what IP's used to connect and calls made as desire or not.

VII. SUGGESTIONS

- Server Asterisk may continue be developed to be able to fax results in eliminate extra cost.
- Be able to develop in the form of Video Conference, makes it more convenient and lower trips cost.
- Be able to apply for educations such as Video Conference learning, etc.

ACKNOWLEDGMENT

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