Asterisk Open Source to Implement Voice over Internet Protocol

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Summary
Voice over IP (VolP) protocol is used to carry voice signal or the IP network. This allows us to use IP Telephone instead of the dedicated voice transmission telephone lines. This paper is focused in the Implementation of Voice over Internet Protocol (VolP) using open source (Asterisk). There are some characteristics for system in this paper such as types of protocol used to establish the connection. The platform for this system design is by using Linux as an Operating System (Debian). The open source software is used to implement the proposed solution.
The problem is necessary for the IP telephony stream to be converted by a gateway to another format, either for interoperation with a different IP based multimedia scheme or because you are placing a call to the traditional Public Telephone Network (PSTN). The overall technology requirements of an IP telephony solution can therefore be split into four categories: signaling .encoding, transport and gateway control.
The objective of this paper is to setup a VolP server by using Asterisk open source which is implemented on a Local Area Network (LAN) and also the main characteristics of VoIP will be explained, to make more use of the internet line rather than its usage for only surfing and chatting.
Key words:
Debian, VoIP, Linux Operating System, H.323 protocol and Sisson Initiation protocol (SIP).

1. Introduction

The "VolP Implementation" was selected because of its potential to be commercialized in the real world. The aim of this paper is to setup VolP server using open source (Asterisk) on the Local Area Network (LAN). The focus will be on Session Initiation Protocol (SIP) during transmitting voice to other party on the network. Platform for this paper is in Linux Operating System (Debian).

To setup a server that provides VolP using Asterisk we must determine the advantages of using VoIP, check the differences of Asterisk over the type of VolP server, and study the differences between H.323 protocol and Sisson Initiation protocol (SIP).
Early VolP service relied on advertising sponsorship to subsidize costs, rather than by charging customers for calls. The gradual introduction of broad band Ethernet service allowed for greater all clarity and reduced latency, although calls were still often marred by static or difficulty making connections between the Internet and Public Telephone Networks (PSTN). However, startup VolP companies were able to offer free calling service to customers from special locations. The results are the following:
1. Setup a server that provides VoIP using Asterisk.
2. Determine the advantages of using VoIP.
3. Check the difference of Asterisk over other type of VoIP server.
4. Study the differences between H.323 protocol and Session Initiation protocol (SIP).

2. Asterisk Open Source

Asterisk is an open source free software implementation of a telephone Private Branch Exchange (PBX) originally created by Mark Spencer of Digium .Like any PBX, it allows a number of attached telephones to make calls to one another, and to connect to other telephone services including Public Switched Telephone Network (PSTN). Its name comes from the asterisk symbol.***** which in Unix (and Unix-like operating systems such as Linux) and DOS environments represent as wild card, matching any sequence of characters in a filename.The basic Asterisk software includes many features available in the proprieties PBX systems: (voice mail, conference phone menu and automatic call distribution).

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3. Types of protocol

There are many types of protocol used in Voice over Internet Protocol (VoIP) implementation such as Session Initiation Protocol (SIP) and H.323 as the following:

A. Session Initiation Protocol

The Session Initiation Protocol (SIP) is a signaling protocol used for establishing sessions in an IP network. A session could be a simple two-way telephone call or it could be a collaborative multi-media conference session. It means that a host of innovative services become possible, such as enriched e-commerce, web page lick-to-dial. Instant Massage with buddy lists and IP Centrex services.

Session Initiation Protocol (SIP) provides

The necessary protocol mechanism to support the following basic functions:

l. **Name translation and user location**: Determination of the end system to be used for communication.

2.**Feature negotiation**: Allows station users involved in a call to agree on the features supported recognizing that not all features are available to all station users.

3.**Call participation management**: During a call, a station user cans conference other Station users into the call or can call connections to conference parties; station users can also be transfer red or placed on hold.

4. **Call features changes**: A station user should be able to change the call Characteristics during the course of the call: new feature may be enable based on call requirements or new conference station users.

The two major components in a SIP network are (Agents and Network Servers. A User Agent Client (UAC) initiates SIP request, and a User Agent Server (UAS) receives SIP request and return responses on user behalf. A Registration Server receives updates regarding the current user location, and a next-hop server, which has more information regarding called party location. A Redirect Server receive request. Determines next-hop Server, and returns an address to client.

The SlP requirement message consists of three elements as the following:

- Request line
- Header
- Message body

The SlP response messages consist of three elements as the following:

- State line
- Header
- Message body

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B. H.313 Protocol.

This International Telecommunication Union (ITU) protocol was originally designed to provide an IP transport mechanism for video conferencing .It has become the standard in IP-based video-conferencing equipment, and it briefly enjoyed fame as a VolP protocol as well. While there is much heated debate over whether SIP or H.323 (or IAX) will dominate the VolP protocol world, in Asterisk, H.323 has largely been deprecated in favor of IAX and SIP. H.323 has not enjoyed much success among users and enterprises, although it might still be the most widely used VolP protocol among carriers.

4. Implementation

The implementation of the system design should setting up the Asterisk server where there are four main things to be done, see Figure 1:

1-The operating system that compatible with Asterisk (i.e. Linux: Debian) in this paper.
2-Download and install all the packages needed as platform for the Asterisk to run.
3- Install the Asterisk and configure it.
4-Install softphone in clients and try to connect to the Asterisk server to see how it works.

A. <u>System Design</u>

In this paper the design on a local area network (LAN) for the VoIP server using Asterisk open source is shown in Figure 2. This design includes only one Asterisk server and two clients with X-Lite softphone connected to the server .The open source software have been used starting from the installation for the server until the installation for the clients.

4.2 <u>Steps to Implement System Design:</u>

There is sequence of steps should be followed before implementation system design as follows:

1- Lunix Operating System (Debian) has been chosen in this paper as the main platform for the Asterisk server. After the setting up the operating system also includes the download the package that’s important in this implementation.

2- The Asterisk open source software have been downloaded and installed in the server such as Libpri and Zaptel.
3- After compiling all the files needed for the server to run, test has been done to make sure the server can run smoothly without any problem.
4- Then, two clients have been chosen as the clients to the Asterisk server. For the clients X-Lite softphone is used because this software supports the Session Initiation Protocol (SIP), see Figure 3.
5- The configurations for the clients are made in the server itself such as the clients Id, context, and password. This will ensure that the server can be used by a specific client t hat has been registered in the server .The configuration also will choose what type of protocol that can be used to established connection between the clients and the Asterisk server.
6- Type of protocol that can be used established connection between the clients and the Asterisk server. See Figure 4.
7-The connection between clients and the server are satisfied using the SIP protocol.

5. Results

This paper provides all the results obtained according to implementation which is "Voice over IP “ (VolP).

1. Used the open source software (Asterisk) because it is more secure due to this VolP system uses server configuration itself.
2. To transmit data from client to the server or other client, there are many types of protocol that can be used such as SIP, H.323, and IAX. This paper used the SIP protocol.
3. The server will reject the connection with clients when the clients use different protocol.
4. This design includes only one Asterisks server and two clients installed with X-Lite softphone connected to the server.
5. The performance analysis of Asterisk VoIP Server with Static and Real-Time Database Call Processing depends on privet branch exchange for Business purpose.
6. Based on actual experiments, conclusion is drawn that Static dial plans are more suitable for both machines used in experiments as compared to real-time dial plan because it serves maximum calls/sec during experiments as a function of CPLJ speed.

6. Discussion

The sip.conffile is for the clients' configuration. This file shows the ID for the users that can connect to the server. It also indicates the password context and kind of protocol to connect to the server.

This file shows the configuration of the clients that can use the server. This file indicates the types of protocol that the clients used the address to bind and port to use (like port 5060) and the extension for calls made by the clients.

7. Conclusion

From the implementation of the proposed solution the effect of the performance results, its leads to the following conclusion:

1. The Voice over IP is used to communicate with less cost than the normal Telephone line.
2. The open source VolP server (Asterisk) also has some advantages such as it is controlled by the server itself.
3. The Asterisk server ensured the security as it will establish connection for the clients that have been assigned in the server.
4. The Asterisk server can be used for many types of protocol such as Session Initiation Protocol (SIP), H.323, and also IAX, where in this paper only focused in Session Initiation Protocol (SIP).

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Fig 1: VoIP Connection

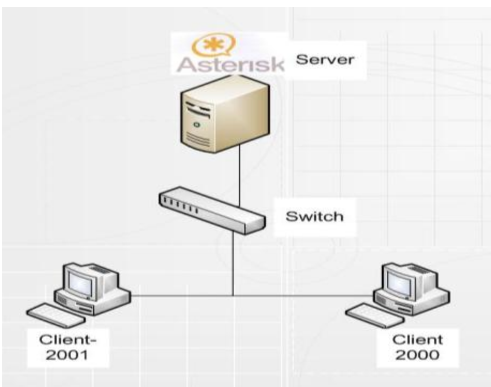

Fig 2: Asterisk Server Design in LAN

Fig 3 :X-Line Configuration

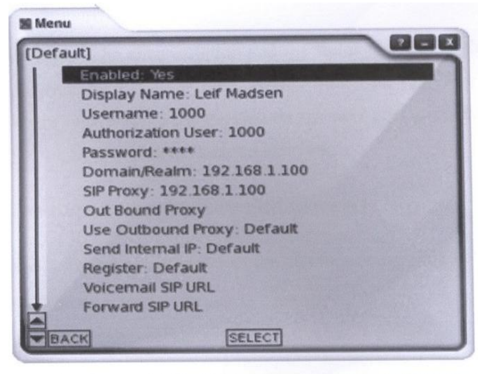

Fig 4 : X- Lite User Configuration