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Research Article

Design and Develop a Telephony System for Establishing Calls between Two PBX Servers Using IAX2 Protocol

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Abstract

Telephony system is the most common way for modern communication. People can communicate with each other's through various ways. Most common existing telephony system are PSTN, ISDN which have lot of drawback such as consume more bandwidth, unreliability, insecure communication media, complex to maintenance. The project design and develop a Telephony system for transferring voice calls within a wired or wireless network. This system works within home, office, simple organization, a campus area to communicate with each other within a wired or wireless zone. In this system communication can takes place between users to users. This project integrates voice with data network and transfer voice over data network more efficiently. There are many existing protocols for transferring voice over data network but none of these are fully perfect. So, the task uses such protocol which satisfies maximum criteria of all existing protocols. For configuring trunk and channels between two servers IAX2 is used. Developed System able to transfer calls from one terminal device to another terminal device within the same or different PBX which consume less bandwidth and quality of calls are better than any other protocols.

Introduction

Telephony system is the most important part of communication. In Modern communication circuit-switched network is common way for person to person communication which is referred to as Public Switched Telephone Networks (PSTN). But communication cost is high, unreliable, maintenance is complex, call routing is complex in circuit-switched network. To reduce these problems, a new trend that is beginning to emerge in recent years is to provide telephony service over IP networks, known as IP telephony. Advantage of this network is cost savings, especially for corporations with large data networks. This system is easy to configure, easy to implement within campus, in a business office and other sectors with no cost. The main advantages of IP telephony is easily integrate with existing system like PSTN, GSM network. These systems are not only transfer calls but also send email, chat with others, establish video calls etc. In telephony system server to server communication or server to user communication there are different types trunking protocol such SIP, IAX2, RDP, H.323 etc. In this project IAX2 protocol is used.

The objective of this project is to design and develop a telephony system within a local LAN or Wireless network. This system maintains communication within an organization through telephony system which reduces communication cost dramatically. It provides communication within a local network.

- Develop a telephony system within a LAN or wireless network
- Configuration of two PBX server
- Connecting two PBX using IAX2 trunking protocol
- Maintaining communication of an organization within LAN or wireless network
- Reduce communication cost within an organization
- Users can use both soft phone and telephone

System Review

There are different trunking protocols such as H.323, SIP, and SDP etc. Comparing SIP with protocols we can say that SIP protocols best of them, but it has some limitation. Few years ago, IAX protocol is invented and then it is one of the best over SIP. Different types of Telecommunication Company such as Sussex [1], NetTelco [2] use IAX2 protocol for their communication.

Bandwidth consumption comparison of SIP and IAX [3,4] (Table 1)

Table 1: BW of some protocols with codec.

Protocol	Codec	BW Theoretically (Kbps)	BW Real (Kbps)
SIP	G.711	5232	5131
SIP	GSM	2184	2087
IAX	G.711	1986.4	2166
IAX	GSM	462.4	552

From Table 1 it is seen that

a. SIP & G.711 codec produce very good quality of voice, but it consumes highest bandwidth.

b. IAX & GSM codec consume lowest bandwidth, but it produced high traffic.

c. IAX & G.711 codec requires high bandwidth and ideal for power traffic level is relatively high.

d. SIP & GSM codec is ideal for plans that do not support IAX.

Analyzing the table data, we can say that IAX protocol is good call transferring.

So IAX2 is used and develop this system with the following steps-

i. Environment Setup

ii. Connecting Asterisk PBX [5] server with IAX2 trunking protocol and establish call session in real-time network.

iii. Check the system and protocol with others for development.

Methodology

First of all, we need to setup open source Linux Operating then install Asterisk PBX, Free PBX web GUI on it. Calculate IP for PBX and terminal devices.

- Configuring IAX2 Trunks
- Configuring the Outbound Routes
- Create channels for end users and bind them with PBX
- Test Call generate and service testing

System Model

In this system Asterisk PBX, Different types of terminal devices, IAX2 tunneling protocol are used (Figure 1).

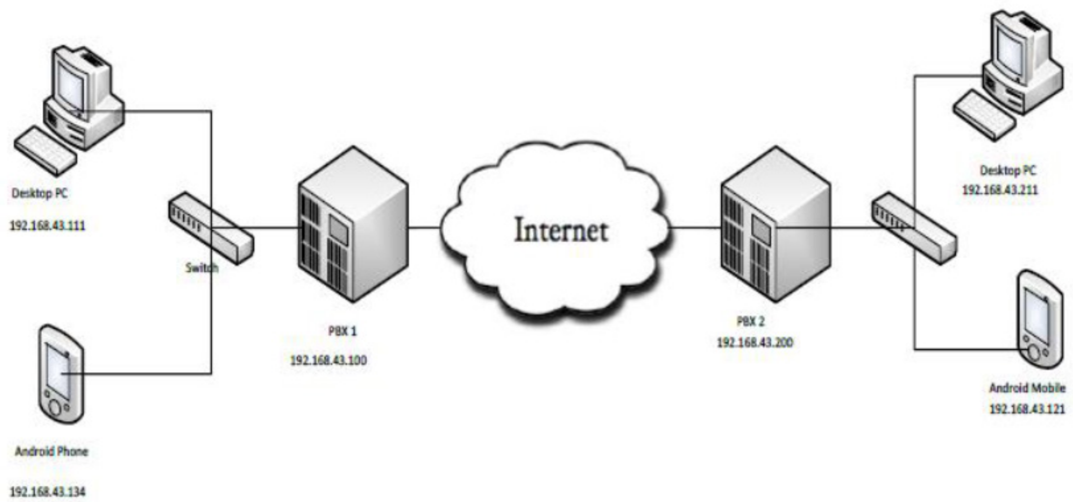

Figure 1: Scenario of the system.

Asterisk architecture

Asterisk consists of five base components [6]:

- Dynamic Module Loader - When Asterisk was first started, the Dynamic Module Loader loads and initializes each of the drivers which provide channel drivers, file formats, call detail record backends, codecs, applications and more, linking them with the appropriate internal APIs.
- PBX Switching - The essence of Asterisk is a Private Branch Exchange Switching system, connecting calls together between various users and automated tasks. The Switching Core transparently connects callers arriving on various hardware and software interfaces.
- Application Launcher - launches applications which perform services for users, such as voicemail, file playback, and directory listing (Figure 2).

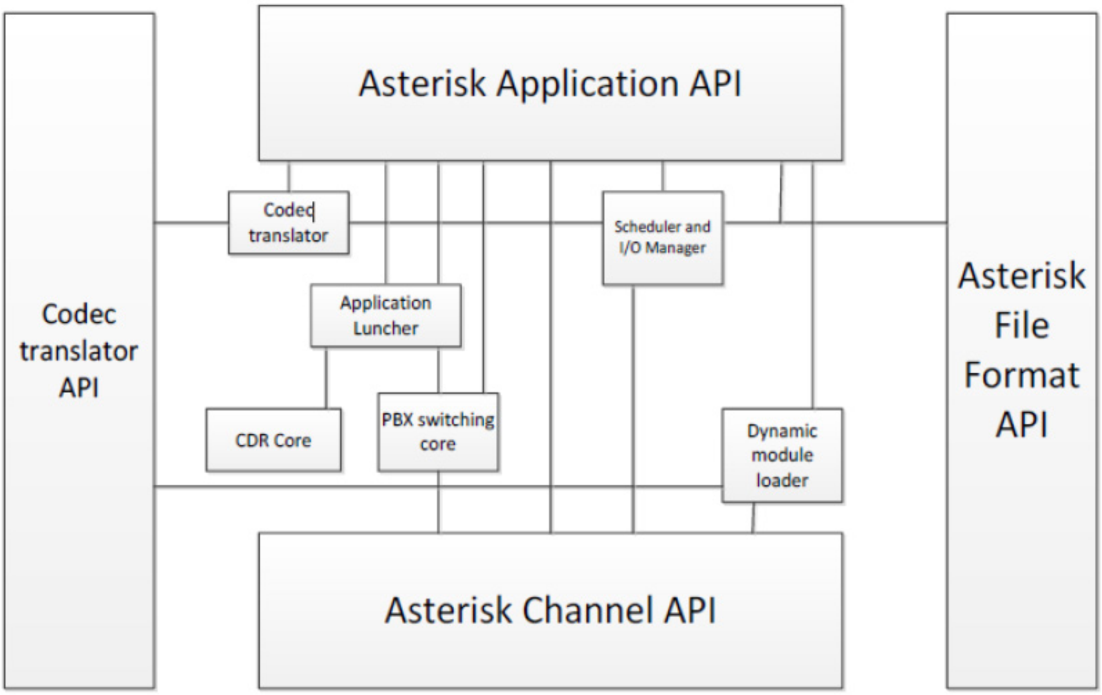

Figure 2: Asterisk architecture.

- Codec Translator - uses codec modules for the encoding and decoding of various audio compression formats used in the telephony industry. A number of codecs are available to suit diverse needs and arrive at the best balance between audio quality and bandwidth usage.
- Scheduler and I/O Manager - handles low-level task scheduling and system management for optimal performance under all load conditions.

There are many types of trunking protocols. Among them we differentiate SIP and IAX2 [7] (Table 2)

Table 2: Difference between IAX2 and SIP.

IAX2	SIP
Consume less bandwidth	Consume more bandwidth
Carry greater number of concurrent phone calls	Carry less number of concurrent phone calls
Compress commands and codes into the smallest size as possible	SIP makes commands and codes into plain text
More secure	Less secure
Single port 4569 (no NAT required)	uses port 5060 and then 10000-50000 for RTP streams

Implementation

Step 1:

For setup Asterisk we need to run Linux Server and Centos is such kind of server. So, first setup centos 6.4 server from DVD or ISO (Figure 3).

Step 2:

After completing the installation, the login screen appears that means CentOS setup is finished. Now it is ready to provide root and centos as login username and password. Then it appears terminal and writes the command for installing Asterisk and FreePBX GUI [7-9]. After install Asterisk and Free PBX GUI, it is ready to use (Figure 4).

Figure 3: Booting centos from installation media.

Figure 4: Login screen of centos.

Step 3:

Normally server has default configuration with DHCP, and dynamic addresses are change frequently. So, it needs to change with static IP. For implementation IP with network 192.168.43.0 and subnet mask 255.255.255.0 is used.

IP addresses of each device (Table 3)

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Table 3: Assigning IP address.

Device	IP address
PBX1	192.168.43.100
PBX2	192.168.43.200
Desktop PC 1	192.168.43.111
Desktop PC 2	192.168.43.211
Android Phone 1	192.168.43.134
Android Phone 2	192.168.43.121

Step 4:

For configuring PBX first, it is needed to access the server. This can be described in two ways. One is by SSL by terminal of another PC and another is by accessing the GUI which is previously installed into it. This GUI runs on web server reside on it. For graphically access first find the IP address and put it on the remote browser in the same network. Then we access the Graphical login page (Figure 5).

Figure 5: Login page of free PBX.

Step 5:

Accessing the GUI from browser it asks username and password. After inserting the username and password "FreePBX System Status" page will appear. We start our main configuration from here (Figure 6) [10].

Step 6:

For integrate terminal device need to channel which transfer calls between end device and PBX. So, configure the extension file with user number and password including channel protocol (Figure 7).

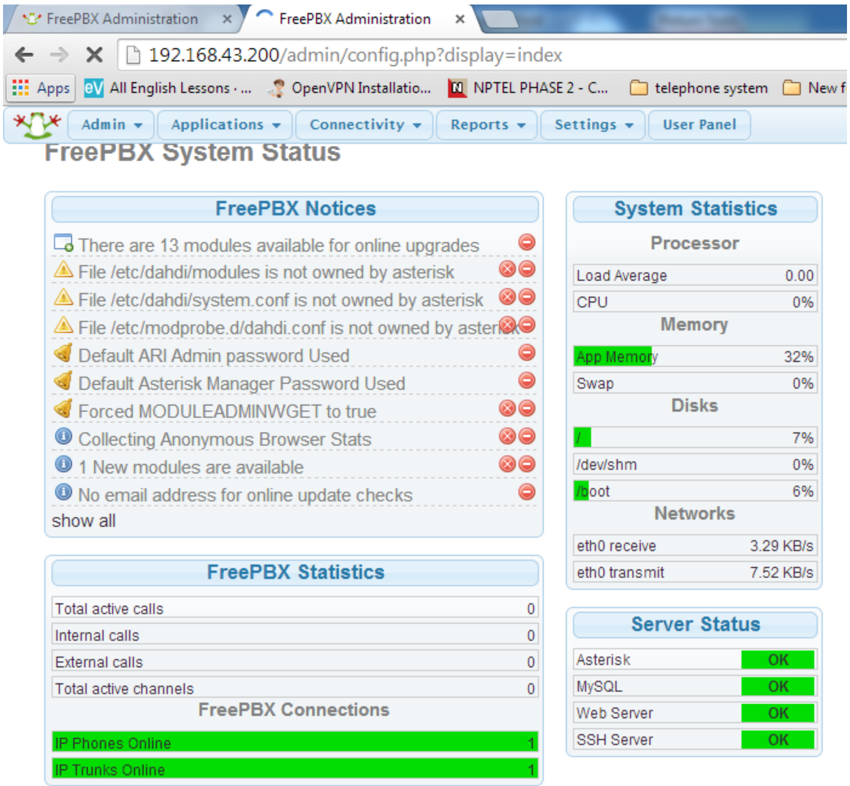

Figure 6: FreePBX system status.

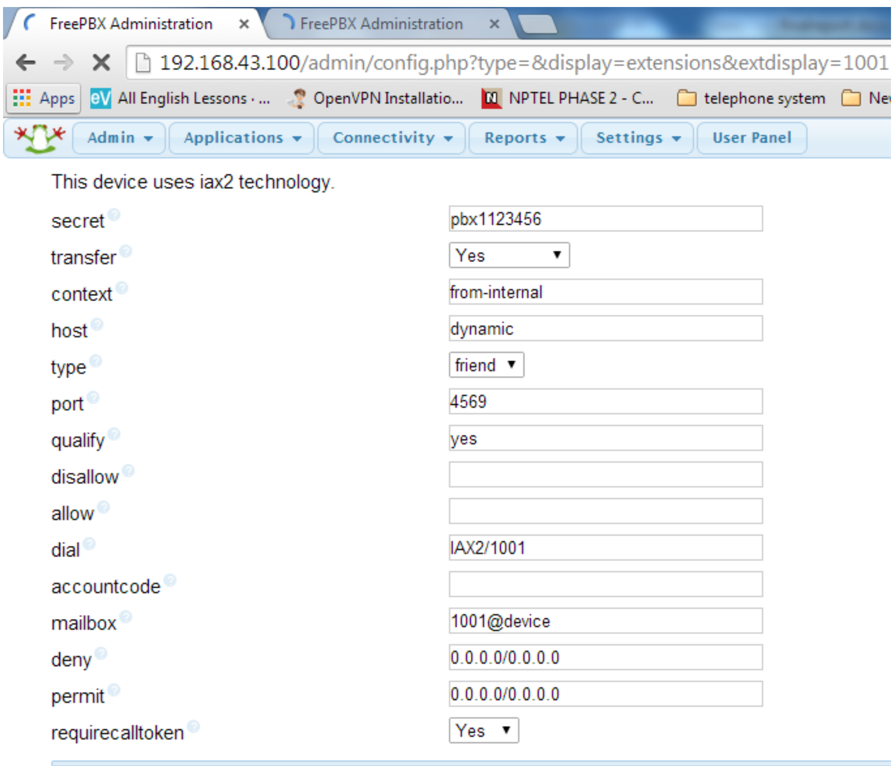

Figure 7: Extension of PBX 1.

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Step 7:

For transfer call from one server to another server it is necessary to configure trunk with appropriate transfer rule. If the dialed number matches with the dial manipulation rule the trunk transfer the call, otherwise not (Figure 8).

Step 8:

Need to configure handset for connecting to PBX 1. For this it is necessary to provide *account name, host, username, password* etc. After providing the correct information click *save button* and it automatically connects the PBX server (Figure 9).

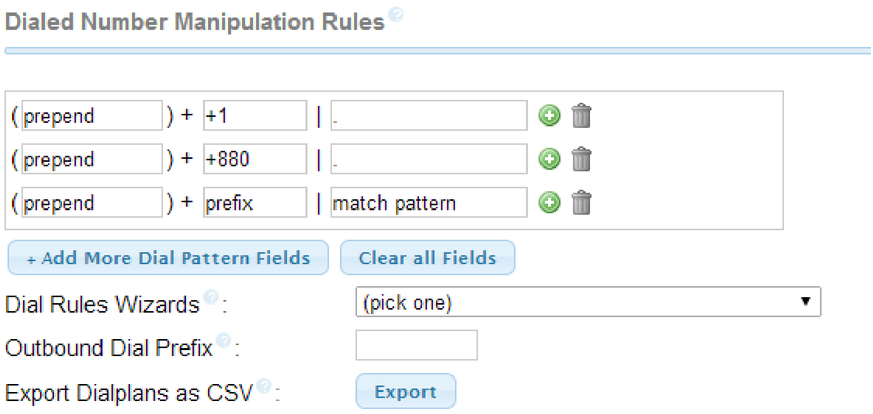

Figure 8: Dial Plan setting.

Figure 9: Handset configuration.

Conclusion

Implementing the protocols, the project found that IAX2 is better than SIP for trunking between two PBX servers. Comparisons are shown in Table 3 between IAX2 and SIP [11]. This system serves better than the system which is configured by SIP protocol used in various existing system. If the server resides different network, it fails to communicate [12].

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Future Work

- We design and develop a new trunking protocol which reduces more bandwidth.
- Develop a pre-configured bootable OS which automatic connect to the radius server using MAC address verification.
- Implementation of MAC addresses authentication and verification.

Acknowledgement

None.

Conflict of Interest

No conflict of interest.

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