



ISSN Print: 2394-7500  
ISSN Online: 2394-5869  
Impact Factor: 5.2  
IJAR 2016; 2(1): 183-186  
www.allresearchjournal.com  
Received: 15-11-2015  
Accepted: 18-12-2015

**Trupti Datarkar**  
Dept. Electronics Engineering  
BDCOE, Sewagram.

**Bobade NP**  
Assistant Professor  
B.DCOE, Sewagram.

**Dr. Gaikwad MA**  
Professor,  
B.DCOE, Sewagram.

## Review on performance analysis of VoIP over the LAN network

**Trupti Datarkar, Bobade NP, Dr. Gaikwad MA**

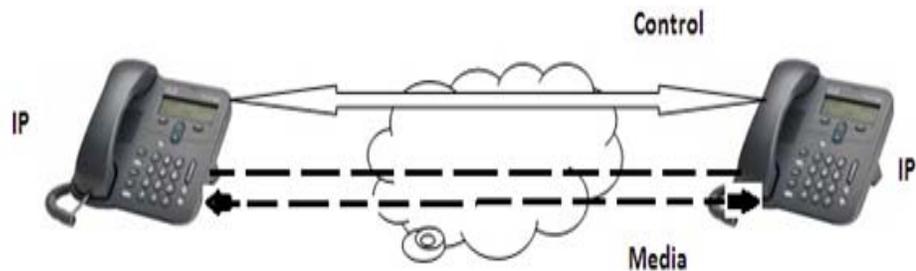
### Abstract

This paper intends to gift some vital theoretical and sensible results that we tend to feature throughout setting up a VoIP (Voice over Internet Protocol) server with the well-known open supply VoIP server Asterisk. For a totally functional voice exchange we tend to need to set up a server based on Asterisk, connecting shoppers to the server with the help of soft/hard phones then comes the configuration aspects of the soft phones with the server. Here in our implementation we tend to have connected the shoppers to the server with the assistance of SIP protocols.

**Keywords:** VoIP, Asterisk, CentOS, SIP.

### 1. Introduction

The term VoIP stands for vocalization web Protocol. VoIP originated in middle 90's, once hobbyists began to note the potential of effort voice data packets over the online instead of communication through normal communication systems. the concept is to use the online as a communication network with some extra capabilities. VoIP converts the voice signal from a phone into a digital signal, sends it through the online, so converts it back at the other finish.



**Fig:** Setting up of a VoIP call

### 2. Ease of Use

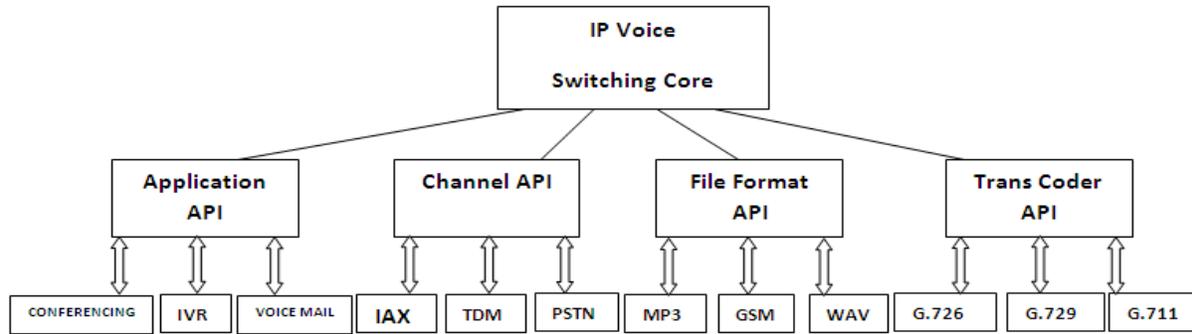
#### Asterisk as a voice exchange

Asterisk will voice science in several protocols, and may operate with the majority standards-based telecom instrumentality exploitation comparatively cheap hardware like for PCI cards [3]. Asterisk in truth creates a PBX that rivals the practicality of ancient phone based mostly systems. the advantages related to associate degree Asterisk based mostly voice exchange can be summarized as:

- Low implementation cost
- Working on TCP/IP protocol
- PBX with enhanced features
- Low Maintenance required
- Convergence of Voice, Video, Data on a single connection
- Easy to add or remove additional extensions.

**Correspondence**  
**Trupti Datarkar**  
Dept. Electronics Engineering  
BDCOE, Sewagram.

## Architecture of Asterisk Based PBX



### 3. Literature survey

In <sup>[1]</sup> Andre du Toit “Private PBX networks-Cost effective communication solutions” in IEEE, 1992. Being the regulative body of artifact market, the commodities market Commission has not taken any steps to make awareness among the public since easement when over a decade of introduction of a artifact by-product market in Republic of India, FMC has taken some constructive steps to open themselves to the general public with reference to regulation, system of operation, players of the market, FMC’s structure Chart and major initiations taken by the govt. and commission. This paper is arrange to take the initiatives taken by the central government and and FMC to the common public and higher understanding of the operation of FMC towards artifact by-product Market in Republic of India when easement.

In <sup>[2]</sup> Guo Fang Mao, Alex Talevski, Elizabeth Chang, “Voice over Internet Protocol on mobile devices” in ICIS 2007.

Voice over net Protocol (VoIP) may be a thanks to do a conversation over an information network. VoIP product promise converged telecommunications and information services that square measure chapel; a lot of versatile and supply smart voice quality as compared to ancient offerings. though VoIP is wide used, VoIP on mobile devices remains in its infancy. Currently, there square measure variety of VoIP solutions for mobile phones. Howell; VoIP solutions developed victimisation Java two Platform small Edition (J2ME) don’t seem to be offered. Java based mostly solutions square measure wide compatible with several devices. during this papal; robust focus has been granted to cross-device compatibility through the utilization of the wide supported J2ME framework. The implementation details of VoIP consumer victimisation J2ME square measure illustrated.

In <sup>[3]</sup> Md. Zaidul Alam, Saugata Bose, Md. Mhafuzur Rahman, Mohamma Abdullah Al-Mumin, “Small office PBX using Voice over IP” in ICACT 12-14 FEB,2007

This paper highlights the look and implementation aspects of a VoIP based mostly asterisk voice exchange, developing a completely useful voice exchange needs to line up a server supported Asterisk, connecting shoppers to it server with the assistance of sappy phones and so configuring the soft phones with the server. Here in our implementation we’ve got connected the shoppers to the server with the assistance of IAX protocols. the primary a part of the paper contains some introductory ideas regarding VoIP, followed by asterisk’s internal design. within the third a part of the paper we tend to discuss regarding the code nd protocols employed by the packet shift based mostly PBX networks and at last

we tend to brush up regarding the look and implementation aspects.

In <sup>[4]</sup> Ryosuke Yamamoto, Fumikazu Iseki, Moo Wan Kim, “Validation of Voip system for University Network” in ICACT, Feb 2008. This paper propose AN example of FMC supported IMS. Then the define of the VoIP system that has been by experimentation developed for the university network supported the projected FMC is represented

In <sup>[5]</sup> Asterisk.org, "Features and Architecture of Asterisk PBX", <http://www.asterisk.org/features>, accessed in March, 2006. This paper highlights the planning and implementation aspects of a VoIP primarily based Asterisk voice exchange, Developing a completely useful voice exchange needs to line up a server supported Asterisk, connecting purchasers to it server with the assistance of sappy phones and so configuring the softphones with the server. Here in our implementation we’ve got connected the purchasers to the server with the assistance of IAX protocols.

The first a part of the paper contains some introductory ideas concerning VoIP, followed by Asterisk’s internal design. within the third a part of the paper we have a tendency to discuss concerning the codecs and protocols utilized by the packet shift primarily based PBX networks and eventually we have a tendency to brush up concerning the planning and implementation aspects.

### 4. Proposed approach framework and design

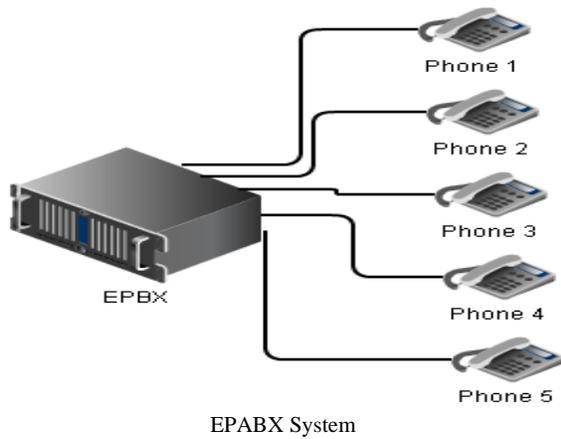
#### 4.1. Problem Statement

The Problem square measure physical science non-public Branch Exchange system the value of wiring for the extensions were enhanced furthermore because it is unable to extend the extensions and not versatile for users.

EPABX stands for Electronic Private Automatic Branch Exchange and it is a phone line sharing device. The device lets you connect the extensions. It works like a mini telephone exchange. These days EPABXs come with a huge arsenal of features like recording, auto call answer, conference calling, call metering, multifunction integration Etc.

Disadvantages of EPABX:

- Limited number of extensions
- No caller ID service
- No Voicemail facility when user is offline
- Complex wiring system
- Less security
- Less flexibility for user
- More electrical power dependency
- No expanding facility

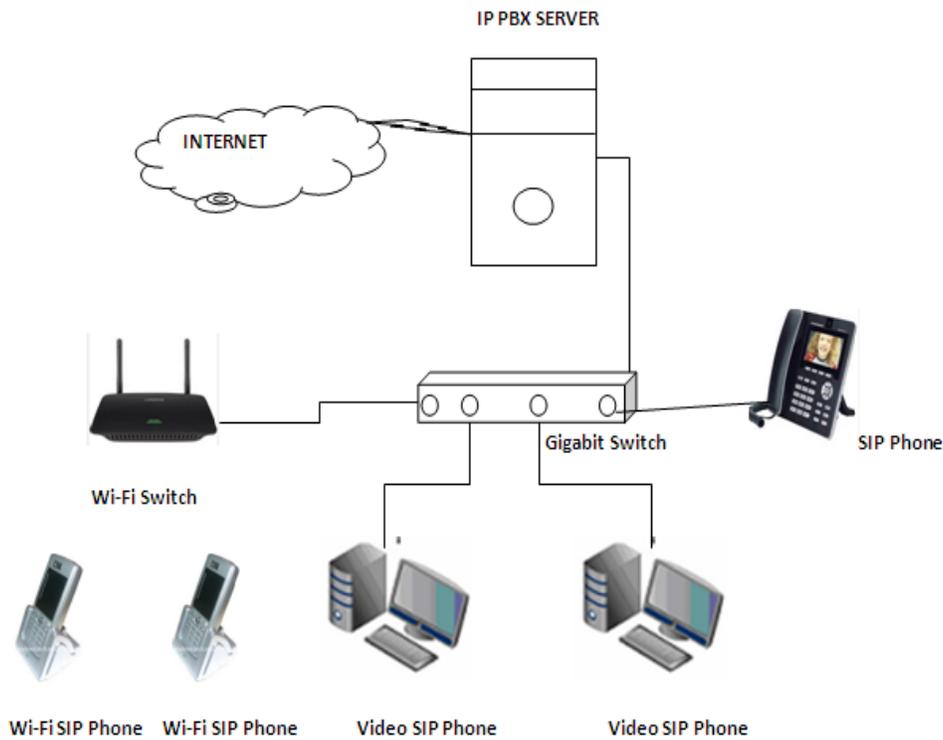


**4.2. Proposed System**

An scientific discipline PBX handles voice signals below net protocol, conveyance advantages for pc telephone integration (CTI). Associate in Nursing IP-PBX will exist as physical hardware, or will do its functions nearly, performing arts the

call-routing activities of the normal PBX or key system as a computer code. The virtual version is additionally known as a \"Soft PBX\".

An scientific discipline PBX takes the place of the PBX you'll have already got for your company's PSTN calls. Like its PSTN full cousin, Associate in Nursing scientific discipline PBX (for non-public branch exchange) is Associate in Nursing electronic patch board that receives, routes, holds, forwards to voice mail, or otherwise manipulates calls that arrive over the web, instead of via the PSTN. it's going to be totally automatic or have a personality's secretary World Health Organization routes incoming calls from a main scientific discipline number to internal scientific discipline numbers or extensions. wherever a PSTN PBX will connect several incoming and internal phone lines through a collection of mechanical or electronic switches, Associate in Nursing scientific discipline PBX are going to be automatically easier, usually either package that resides on a server or atiny low, freelance server that connects together with your existing knowledge network.



**Fig:** Network Architecture of IPPBX

An informatics PBX provides additional economical shift and a additional skilled "look" than if everybody during a business has their own separate informatics association and account. It permits phone calls to be forwarded at intervals the corporate, and for internal voice-mail and conferencing capabilities which may otherwise need to be outsourced. associate degree informatics PBX is additionally rather more versatile than a PSTN PBX, permitting associate degree basically infinite range of extensions and voice-mail boxes, and desktop management via an online browser instead of at a collection of PSTN switches. they'll conjointly change the recording of incoming and outgoing conversations (subject to legal considerations). IP PBXs, each as software package

and as physical devices, area unit comparatively inexpensive [5].

**4.3. Design**

Asterisk is with success designed for optimum flexibility. Distinctive Apis square measure outlined around a central PBX core system. This advanced core handles the interior interconnection of the PBX, clearly abstracted from the precise protocols, codes, and hardware interfaces from the telecom applications. this permits Asterisk to use any appropriate hardware and technology obtainable currently or within the future to perform its essential functions, connecting hardware and applications. The Asterisk core handles this stuff internally.

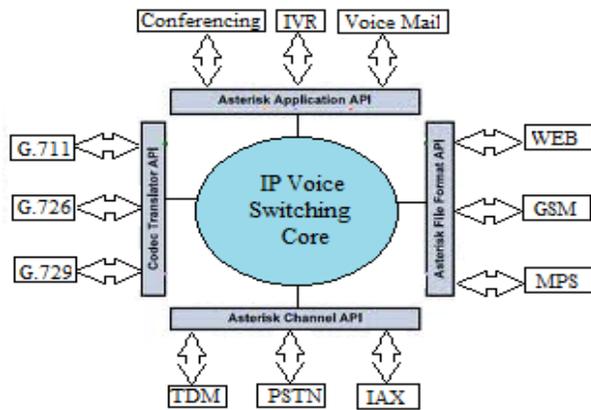


Fig: System Designing

#### 4.4. Performance measurement

For this project the performance parameter will be Bandwidth, Jitter and latency. These parameters will be monitored by using Wireshark software which run on Linux server

#### 5. Conclusion

Here we propose to replace the EPABX system with IPPBX system. This paper provides data concerning the IPPBX and its communication. The irresponsibility of the system is totally depends upon the network. If the network fails the full communication fails. conjointly to speak with outside the organization the net facility ought to be out there and therefore, the net charges are applicable otherwise for the interior communication the line of work are freed from charge. This non-public branch exchange conjointly provides voicemail facility. Once users is offline or busy the line of work user is ready to send the voicemail for the referred to as user. The researchers are attempting to implement the system on wireless local area network further as for the wireless local area network enabled movable therefore because the users might communicate with one another exploitation mobile phones and can analyze the performance of the system by exploitation some parameters like information measure of the network, result of variety of users, irresponsibility and adaptability. By comparing the two systems it is conclude that the IPPBX system is having advance features than EPABX system such as IVR, WiFi Intercom on smart phone, Call monitoring and more as well s researcher try to reduce the cost of IPPBX system by implementing Asterisk as a software for intercommunication via IP address.

#### 6. References

1. Andre du Toit. Private PBX networks-Cost effective communication solutions in IEEE, 1992.
2. Guo Fang Mao. Alex Talevski, Elizabeth Chang, Voice over Internet Protocol on mobile devices in ICIS, 2007.
3. Md. Zaidul Alam, Saugata Bose, Md. Mhafuzur Rahman, Mohammad Abdullah Al-Mumin. Small office PBX using Voice over IP in ICACT I2-14 FEB, 2007.
4. Ryosuke Yamamoto, Fumikazu Iseki, Moo Wan Kim. Validation of Voip system for University Network in ICACT. Asterisk.org, Features and Architecture of Asterisk PBX, 2008.  
<http://www.asterisk.org/features>, accessed in March, 2006.

5. Taemoor Abbasi, Shekhar Prasad, Nabil Seddigh, Ioannis Lambadaris. A comparative study of the SIP & IAX voice protocols in CCECE/CCGEI, Saskatoon, 2005.
6. Anand Gorti. A fault tolerant VoIP implementation based on open standards in EDCC in 2006.
7. Andre du Toit. Private PBX networks-Cost effective communication solutions in IEEE, 1992.
8. Guo Fang Mao, Alex Talevski, Elizabeth Chang. Voice over Internet Protocol on mobile devices in ICIS, 2007.
9. Dynamic Codec Selection Algorithm for VoIP-ICDT, 2011.
10. Md. Zaidul Alam, Saugata Bose, Md. Mhafuzur Rahman, Mohammad Abdullah Al-Mumin. Small office PBX using Voice over IP in ICACT I2, 2007.
11. Ryosuke Yamamoto, Fumikazu Iseki, Moo Wan Kim. Validation of Voip system for University Network in ICACT, 2008.
12. Asterisk. Org, Features and Architecture of Asterisk PBX, <http://www.asterisk.org/features>, accessed in March, 2006. Ip-Pbx Module Requirements, 2014.
13. Anand Gorti. A fault tolerant VoIP implementation based on open standards in EDCC in 2006.
14. Exploiting VoIP. Telephony in IP-Pbx Solution Ajasa AAA and Shoewu O. PJST, 2012.
15. Voice over IP. Mobile Telephony Using WIFI Rahul C. Vaidya, Prof. S.S. Kulkarni 2012; 1:2229-5518.