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Alternative of Traditional PBX using VoIP for Voice/Video call, Voice Mail, Instant Messaging

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Abstract

The work proposed in this paper added features for placing the voice and video calls using IP phones (soft phones) hence increasing the mobility of the users. The model is successful in carrying out voice and video calls on PC's connected with both wired LAN and wireless LAN using Soft Phones and android supported hand held connected with the wireless network. Every user is provided with his own extension number, the communication devices can make voice and video call, voice mail, Instant messaging and Interactive voice response, that can be used to connect within organization. We had used Ozeki Server for this project; Ozeki is an open source software platform which uses PBX (Private Branch Exchange) as the kernel to build unified communications system. It can choose the combination of different communication components to achieve customized solutions. This project implemented the functions of VOIP (Voice over Internet Protocol) like voice call, video call, chat and voice mail.

Keywords: *IP phones, Voice/Video call, PBX (Private Branch Exchange)*

INTRODUCTION

The Internet is a global system of computer networks interconnected using the standard Internet Protocol Suite (TCP / IP) to serve many users around the world. It is a network of private business, community, academic, and international or local governments are connected to each other. Such solution, among employees can communicate with the Internet that has been provided by the office.

Customers can communicate with VoIP over IP networks [1]; so as to reduce communication costs can offset the cost of the customer. Customers can also directly contact the VoIP number of employees, so that the aims and objectives can be delivered direct customer. There will be no any need for extra infrastructure for the telephone network and don't need to install the costly equipment's; we just need ATA telephone adapter to provide this service.

When you call someone then it routs to the prescribe person through the Internet and if the person is not responding to the analogue phone, the call automatically routed to the smart phone or any other phone which must be android supported.

The purpose is to improve the mobility in the network as well as on a single line performing multiple functions.

Each users can communicate wherever they are, even though the area is very difficult for the cable network, provided that the area can access the Internet. It is also very Cheap, because the cost of a call is not affected from a distance or close the communication was made, but from out of the use of Internet access, No one can ignore its Mobility factor, by using WLAN network, so that users can communicate in mobile conditions.

Almost every organization use PBX, but this project is more cost efficient with many extra embedded feature in a low cost that is why every organization, hospitals, educational sectors or NGO will be demanding product like this project. Every organization which need PBX inside it is necessary to installed this project rather to installed the old concept old PBX.

LITERATURE SURVEY

VoIP protocol is used in VoIP transport so that voice data can be sent properly, SIP protocol is used , the following explanation of the SIP. SIP protocol is supported by some protocols, such as RSVP to make a reservation on the network, RTP and RTCP media for transmitting and know the quality of service, as well as media SDP to describe the session [4]. There are two types of network servers, namely: Proxy server is a server that receives the request, processes it, and forwards the request it receives to the next hop server after changing some headers in the request message [3]. The configuration will require a form of gateway interfaces that connect VoIP networks to the Internet network.

Computer network: Two computers can be said to be connected if they exchange data / information, a variety of owned resource, such as files, printers, storage media (hard disk, CD-room, and flash disk). Data in the form of text, audio, and video media moving through wires or wirelessly enabling computer users in computer networks to exchange files / data, print on the same printer and using hardware / software that is connected in a network together. In computer networking system known connections between computers, namely:

- Peer to Peer
- Client - Server

VoIP (Voice over Internet Protocol)

Voice over Internet Protocol known as IP Phones. In general, VoIP is defined as a system that uses the Internet to transmit voice data packets from one place to other using IP protocol intermediaries. In fact, VoIP is more focused on using the Internet compared with traditional phone infrastructure built earlier[2].

VoIP systems employ session control and signaling protocols to control the signaling, set-up, and tear-down of calls. They transport audio streams over IP networks using special media delivery protocols that encode voice, audio, video with audio codec's and video codec's as Digital audio by streaming media.

VoIP Protocols

VoIP protocol is used in VoIP transport so that voice data can be sent properly.

SIP (Session Initiation Protocol)

SIP is a protocol multimedia issued by the group incorporated in Multiparty Session Control (MMUSIC) within the organization Internet Engineering Task Force (IETF) as documented in a Request For Command document (RFC)[6].SIP is a protocol that is at the application layer that defines the initial, modification, and termination of a multimedia communication session. Multimedia communications sessions include relationship, distance learning, and other applications.

SIP protocol is supported by some protocols, such as RSVP to make a reservation on the network, RTP and RTCP media for transmitting and know the quality of service, as well as media SDP to describe the session.[7] By default, SIP uses UDP protocol, but in some cases may also use TCP as the transport protocol.

OBJECTIVE

- To design IP based PBX network architectures
- To customize efficient and effective Soft Switches
- To implement real time and non-real time applications
- Design Global network by joining small IP based PBX
- Integrate Analog Phone and configure these phone
- Installation of server and troubleshoot all problems
- Registered SIP account and give all features of system

METHODOLOGY

Preparation Phase

Preparation needs to do is prepare the VoIP network in general; we are going to implement the project in college while using all resources of our college.

Bandwidth: Bandwidth given to PC VoIP server is 1MB. With the number of VoIP users were 6 pieces, and use codec PCMU. So generally get computations bandwidth used by 6 x 64KB = 384 Kb. So with 1MB bandwidth is adequate.

Network architecture: The network architecture is shown here. The Fig 1 showing the interconnection of the hardware components between different devices.

Soft phone: Soft phone used in this project has call forward features required by the organizations. In other soft phones call forward facility existed, but they require advance registration process, and are less user friendly.

Connection: The medium used for the VoIP user for connection to the server is the Internet. So the user can connect to VoIP server via the Internet wherever they may be. Connection of VoIP is shown in Fig. 2

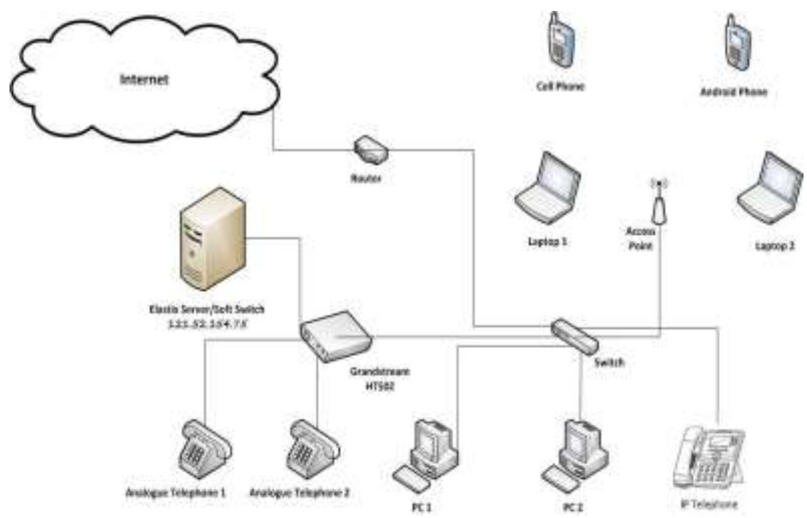

Fig. 1 VoIP Network built drawings [7]

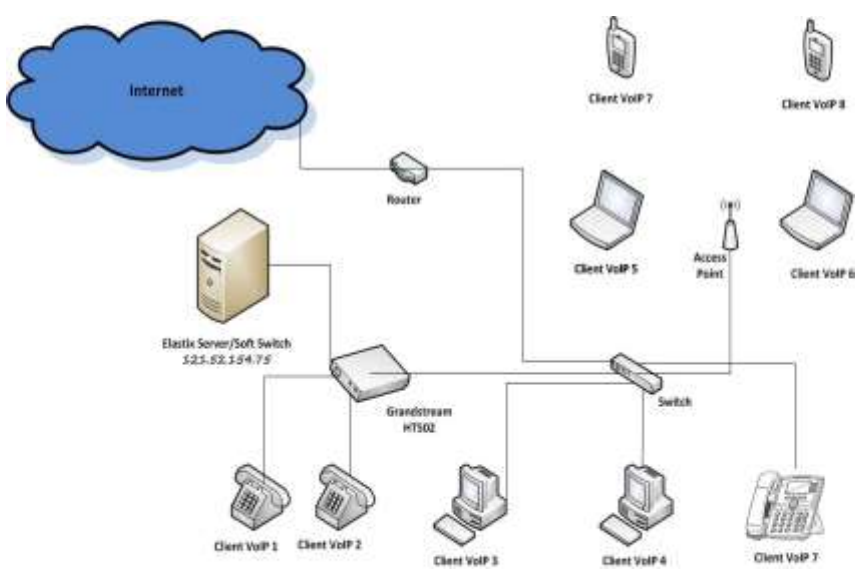

Fig. 2 VoIP Network [4]

Implementation Phase

The implementation phase is divided into two parts, namely the installation and configuration. First installing the soft phone on the PC/Laptop or on the android device and second configuring it to the Server using registration process.

Implementing the features of Ozeki server

Voice call: The voice call is the basic property of unified communication system, here the voice call is based on sip protocol. The communication is allowed only for those who are registered to the server. The server will check for the connection and will not allow any unauthorized access.

Video call: To enable video calls we need to configure /etc/ozeki/ sip_general_custom.conf to: video support=yes, allow=h264, allow=h263 and allow-h263p.

Voice mail: Voice mail is configured to handle calls that can not be answered. Voice mail is generally made is to call user group. Flow chart of voice mail can be seen in Fig. 3

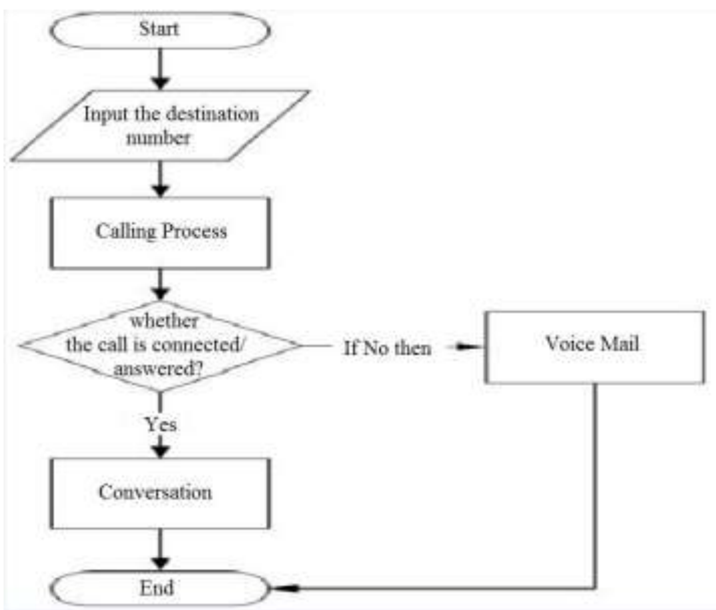

Fig. 3 Voice Mail flow chart

Personalized voice mail is a feature that allows callers to leave messages on phone. Voice mail permits to record user outgoing message, so that when calls are routed voice mail callers will hear greeting and have the option to leave a message. The voice mail message will also provide a time stamp so user know when user caller contacted.

PROPOSED WORK

Registration of the VoIP user

Registration process need users to install the soft phone on their device. The soft phone will direct the user to the registration page, where user has to fill the details and then click the register button. After this the entered details will be send to the server and the server will check for the validity of the user registration. Validation of the user detail will provide a number to the user that can be used to connect with the user. The registration process is shown in the fig. 4

Calling fellow user

In the process of dialing phone user will require the phone number of the fellow user which is given at the time of registration on the PBX Server. The Soft Phone will connect to the server in the presence of the internet, and then server will forward the call to the fellow user, and when the fellow user accepts the call request the connection between them will be established. The connection will vary on the basis of internet bandwidth. If the fellow user will be not able to accept the request, the caller will be connected to the voice mail of the fellow user. Fig. 5 is an illustration of sesame user VoIP calls.

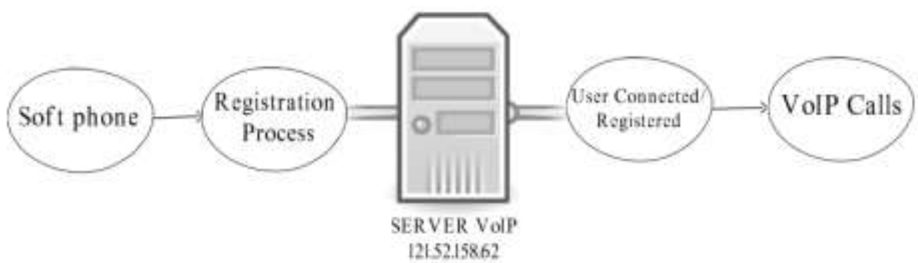

Fig. 4 Registration process of the user

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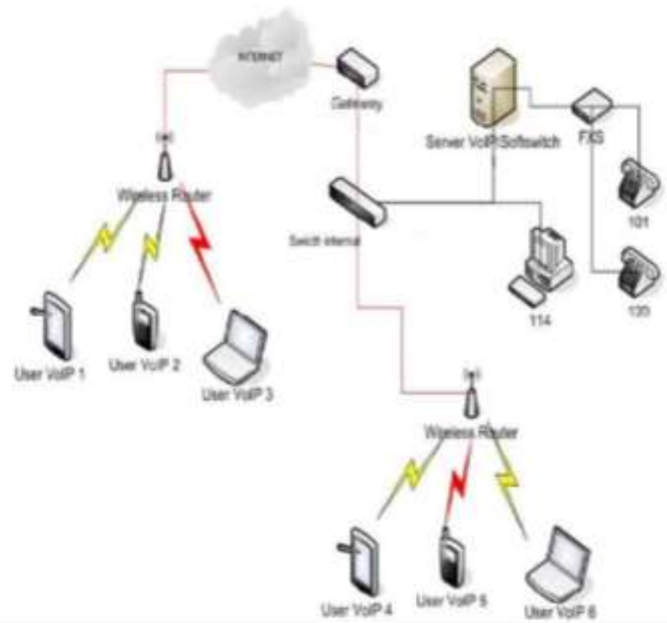

Fig. 5 Calls between VoIP users [8]

Incoming calls

Incoming calls will be received and handled by the IVR (Intelligent Voice Response), then from there the call will be transferred in accordance with the purpose of the caller.

RESULT

System testing process

Steps taken in the testing of the system is user registration on the IP PBX server, calls to fellow user, incoming calls from the laptops, soft phones, Android based IP telephone to a user.

Registration of VoIP user

Registration of VoIP user using a PC/Laptop is adding phone functionality on user PC/Laptop that is by installing a soft phone application. In the testing that has been done with some application soft phone, our soft phone has major advantages compared to other soft phone programs, namely:

- Can do call forward and call transfer
- Great Usability
- Easy to understand the operation

In addition to these, there are some problems in the process of registration of VoIP users, this can occur for many reasons, here is the list of various errors that can be occur while registering:

400 Bad Request

Requests cannot be understood by the server, the blame lies on the Soft phone SIP profile, complete reconfiguration on SIP soft phone is required.

401 Unauthorized

Request requires user authentication, user authentication error on SIP soft phone profile, complete reconfiguration on SIP soft phone is required.

404 not found (User not found)

Registration request cannot be accepted, because the proxy configuration on the soft phone user cannot find the proxy in question.

403 Forbidden

Requests can be understood by the voip server, but not biased implemented. The blame lies on the soft phone profile, complete reconfiguration on SIP soft phone is required.

VoIP calls to fellow user

In the process of dialing phone, Internet (VoIP user) did not experience a lot of problems, but with the provision that user is active / online at the time of the call. Users who have low connection more often fail in doing the calling. It simply cannot be avoided, unless the user adds VoIP bandwidth used.

Incoming calls

Calls will be received and handled by the IVR (Intelligent voice response), then the call will be transferred in accordance with the purposes of the caller. One of the identification of the inbound problem is when there is an incoming call and the call is not handled by the IVR, but only a ringing tone on the caller side can be hear.

CONCLUSION & DISCUSSION

This project can be implemented in the university campus to provide free voice and video calls. It’s a most effective way to diminish the large phone call bills. The service is secure and allows only the registered user to place calls. Moreover, all the calls placed using the Ozeki server are encrypted thereby avoiding hackers to intercept an ongoing phone calls. Ozeki based voice exchange provide us with a much better alternative solution. It’s not only cost effective but also provides us with various features which we generally don’t get with the conventional circuit switched based PBX. Moreover, the system also provides for unlimited expansion and since it runs on a secure operating system, it's much less prone to viruses, worms and hackers.SIP is less complex than other protocols too. The quality of service is only not good while communicating phone to any SIP phone, when there is noise from the wireless network or due to ATA(Analogue Telephone Adapter).

FUTURE WORK

A. Integration with PSTN Network

Ozeki can connect with the existent PSTN by using FXO telephony card, so it is possible to be used as the VoIP gateway this will increase portability and decrease cost.

Integration with GSM Network

This project can also be integrated with GSM network through gateways. This will increase portability and decrease cost. This project can call from our all communication devices to any Skype ID.

High security for large scale enterprise network

When developing the large scale enterprise network by connecting multiple Ozeki servers located in different sites based on IAX2, to solve this issue, VPN method could be established by using open VPN.

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