ISSN No: 2348-4845
International Journal & Magazine of Engineering, Technology, Management and Research
A Peer Reviewed Open Access International Journal

Design and Implementation of VOIP Setup Using CISCO 2811 Routers

P.Prasad
UG Student
Department of ECE,
Sri Chuindi Ranganayakulu Engineering College,
Ganapavarm.

G.Harishankar
UG Student
Department of ECE,
Sri Chuindi Ranganayakulu Engineering College,
Ganapavarm.

Sameer Kumar
UG Student
Department of ECE,
Sri Chuindi Ranganayakulu Engineering College,
Ganapavarm.

Roshan Kumar
UG Student
Department of ECE,
Sri Chuindi Ranganayakulu Engineering College,
Ganapavarm.

Dara Naveen
Assistant Professor
Department of ECE,
Sri Chuindi Ranganayakulu Engineering College,
Ganapavarm.

R.Sambasiva Nayak
HOD
Department of ECE,
Sri Chuindi Ranganayakulu Engineering College,
Ganapavarm.

ABSTRACT:

Whenever there is a new need of communication, it has parallel effect on communication technologies. From symbolic and sign language to the current Internet, the role of communication technologies drastically changed their face. Today, we are provided with various kinds of communication technologies, which are shaped up according to the user needs. Whether it is a simple Fax or a telephonic conversation or an E-mail, each one is intended to satisfy a specific need of user. Though each mean of communication has its own advantages & significance, apparently voice communication is more popular and predominant in our society. The unique advantage of real time experience of telephonic conversation made it more popular among all other means of communication. Though it is popular, providing such communication at reasonable costs is still remains a challenge.

Considering the fact that cost reduction is not possible in the current telephonic system the only option left to us is to develop alternative voice communication technologies which can provide services at low cost.

The development of Internet has opened the doors to explore the alternative means of voice communications. The Internet telephony is not just a mere alternate of voice communication, but it is a revolutionary concept, since it is the most effective way of reducing the costs of voice communication.

The Internet telephony communication is done by using VOIP and is an acronym for Voice over Internet Protocol, or in more common terms phone service over the Internet. To establish this communication we have to develop VOIP Client & VOIP Server.

KEYWORDS: *Voice over Internet Protocol, Internet protocol, Public Switch Telephone Network, Public Land Mobile Network, Session Internet Protocol (SIP), Real Time Protocol, Local Area Network, Wide Area Network.*

INTRODUCTION:

VOIP is an acronym for Voice over Internet Protocol, or in more common terms phone service over the Internet. Other terms commonly associated with VoIP are IP telephony, Internet telephony, broadband telephony, and broadband phone service. A major development that started in 2010 was the introduction of mass-market VoIP services that utilize existing broadband Internet access, by which subscribers place and receive telephone calls in much the same manner as they would via the public switched telephone network. Full-service VoIP phone companies provide inbound and outbound service with direct inbound dialing. Many offer unlimited domestic calling for a flat monthly subscription fee. This sometimes includes international calls to certain countries. Phone calls between subscribers of the same provider are usually free when flat-fee service is not available. A VoIP phone is necessary to connect to a VoIP service provider. As the term says VoIP tries to let go voice (mainly human) through IP packets and, in definitive through Internet. VoIP can use accelerating hardware to achieve this purpose and can also be used in a PC environment.

PSTN SYSTEM:

For the past many years people have relied on the PSTN for voice communication. The two parties using the line. No other information can travel over the line, although there is often during a call between two locations, the line is dedicated to plenty of bandwidth available. Later, as data communications emerged, companies paid for separate data lines so their computers could share information, while voice and fax communications were still handled by the PSTN.

More than 30 years ago Internet didn't exist. Interactive Communications were only made by telephone at PSTN line cost. Data exchange was expansive (for a long distance) and no one had been thinking to video interactions (there was only television that is not interactive, as known). Today we can see a real revolution in communication world: everybody begins to use PCs and Internet for job and free time to communicate each other, to exchange data (like images, sounds, documents)and, sometimes, to talk each other using applications like Net meeting or Internet Phone.

Particularly starts to diffusing a common idea that could be the future and that can allow real-time vocal communication: VoIP. Presently telecommunications system employs one of the three types of the following networks based on the end-user requirements. Those are

1. PSTN: It was basically developed and engineered for giving voice connectivity to the wire line subscribers.
2. PLMN: It has been developed to provide voice services for wireless subscribers.
3. Data Network: This network was basically designed for accessing remote files and servers for defense people and universities but now days nobody can think of living with data network services. The basic and most popular application of data networks is Internet.

DISADVANTAGE:

1. Not suitable for IP environment
2. No convergence features
3. High CAPEX & OPEX
4. Large Power, Cooling requirements.

VOIP SYSTEM:

Today, with the rapid adoption of IP, we now have a far-reaching, low-cost transport mechanism that can support both voice and data. A VOIP solution integrates seamlessly into the data network and operates alongside existing PBXs, or other phone equipment, to simply extend voice capabilities to remote locations. The voice traffic essentially "rides for free" on top of the data network using the IP infrastructure and hardware already in place. This system is based on VOIP concept.

What is a PBX?

Asterisk is a software implementation of a PABX. A PABX, usually called a PBX, is a Private Automatic Branch Exchange. A PBX is private because the enterprise owns it, not the telephone company. The telephone company can still be a supplier or service provider. Originally, PBX equipment was analog, more recent PBX equipment is digital. A PBX is cost

ISSN No: 2348-4845
International Journal & Magazine of Engineering, Technology, Management and Research
A Peer Reviewed Open Access International Journal

attractive because it is less expensive to use a PBX than a separate phone line for every user in the enterprise and because it provides more services. With a PBX, lines from the telephone company can be shared instead of having a separate line to the telephone company for each user. A PBX provides a place for trunk (multiple phones) lines to terminate at the enterprise. A PBX is a telephone system that services an enterprise by switching calls between enterprise users on local lines and by sharing the external phone lines. The PBX has the intelligence to switch calls within the enterprise and outside the enterprise. A PBX provides features and capabilities not available with direct connections to the Public Switched Telephone Network (PSTN.) A PBX moves telephone functions from the phone company to the enterprise. A PBX provides additional functions and features like interactive voice response, call waiting, conferencing or voice mail, paging, transferring calls, or three ways calling that wouldn't be available with separate telephone lines. A PBX usually has a console for use by an operator.

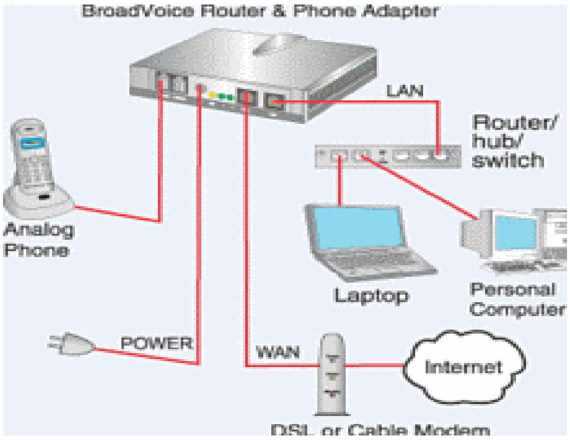
How Does Asterisk Compare to a PBX?

ET systems, and Asterisk, provide interoperability between a local system and the PSTN. Many features in a legacy PBX system are rarely used. Some features may have been developed for a single user to make a single large sale. Because of this, Asterisk does not yet have all the features of all PBX systems from all vendors. Because Asterisk is an open platform features are easy to add and many new features are being added all the time. If Asterisk does not yet have a feature you want it is either already under development or easy to add. Any feature added to Asterisk by any user will be available for you to use. This is because Asterisk is an open source product distributed under a GPL license.

What is Asterisk?

Asterisk is open source. It implements communications in software instead of hardware. This allows new features to be rapidly added with minimal effort. You can easily make your own changes or additions. With its included support for internationalization, rich set of configuration files, and open source code, every aspect of Asterisk can be customized to meet your needs. New interfaces and technologies are easily added to Asterisk. With Asterisk you can take control of your communications. Once a call is in your Linux sever with Asterisk, anything can be done with it Asterisk gives you fine-grained control over every aspect of your communications

VOIP ARCHITECTURE:



In This paper involves design and implementation of VOIP services over IP platform for providing communication set up with 2811 CISCO routers.

Protocols: 1. SIP 2. RTP 3. H.343

ADVANTAGE:

- Low call rate
- Extend the functionality of the corporate IP voice, video and data solutions to remote office locations.
- Lower equipment administration costs
- Centralized network control and management
- Increased customer satisfaction through the use of distributed call center applications
- Reduces infrastructure cost and complexity
- Increased communications capabilities and productivity for remote and mobile employees

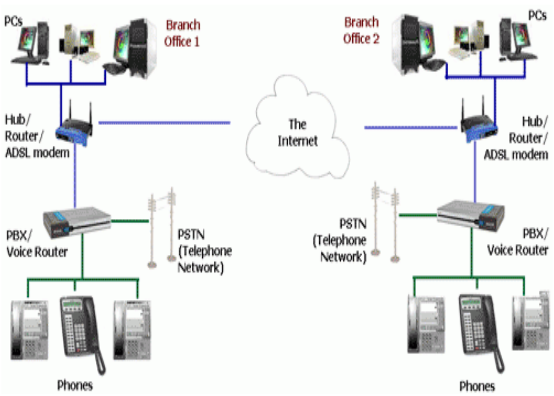
ISSN No: 2348-4845
International Journal & Magazine of Engineering, Technology, Management and Research
A Peer Reviewed Open Access International Journal

- Mobility: a user can move an IP phone from location to location, state to state while still keeping the same telephone number and set of feature.

DISADVANTAGE:

- Internet Requirement: IP phones must have access to the Internet to work.
- Quality of Service (QoS): Quality of Service is a major issue in VOIP implementations. The issue is how to give warranty that packet traffic for a voice or other media connection will not be delayed or dropped due to interference from other lower priority traffic.

RESULTS:



A VOIP test tells you if your Internet connection is equipped for a cloud phone system. If your connection does not meet the test's standards, your calls could suffer from subpar quality.

CONCLUSIONS:

VOIP Network provides voice communication for both Analog and IP Phones. VOIP treated as an approach towards complete IP communication and so it is called as future telecom network for Voice.

REFERENCES:

[1] CCITT Recommendation, G.711: Pulse Code Modulation (PCM) of Voice Frequencies, Geneva: International Telecommunication Union, 1972.

[2] H Chang, M Hsua, M. Chang, V Hsua, "The Internetworking functions of VoIP protocols", http://www.ccrc.nthu.edu.tw/excellence/ activity/911015/2/s-2/3.pdf.

[3] J. Glasmann, W. Kellerer, "Service Development and Deployment in H.323 and SIP", in Proceedings of 6 Th IEEE International Symposium on Computers and Communications (ISCC), 2001, pages 378-385.

[4] M. Goncalves, "Voice Over IP Networks", McGraw-Hill Companies, Inc 0-07-913783-0, 1999.

[5] M. Handley, H. Schulzrinne, E. Schooler, and J.Rosenberg, "SIP: Session Initiation Protocol", IETF RFC2543, March 1999.

[6] M. Huh, W. Hyun, S. Kang, D. Kim, " Call Management Mechanism for Internet Phone Services based on SIP", Proc. of the 5th International Conference on High Speed Networks and Multimedia Communication, 2002, pages 66-70.

[7] W. Hyun, M. Huh, S. Kang, "An Implementation of SIP Servers for Internet Telephony", Proc. of the th International Conference on High Speed Networks and Multimedia Communication, 2002, pages 61-65.

[8] D. Pan,"Digital Audio Compression", Technical Journal, Vol. 5, No. 2. Spring 1993.

[9] J. Rosenberg and H. Sculzrinne, "Models for multi party conferencing in SIP", Internet Draft, IETF, Nov.2000, Work in Progress.

[10] H. Schulzrinne, J.Rosenberg, "Signaling for Internet Telephony", Proceedings of the Sixth International Conference on Network Protocols, October 1998.

[11] H. Schulzrinne, E. Wedlund, "Application Layer Mobility using SIP", In Proc. of IEEE Service

Portability and Virtual customer Environments, December 2000, pages 29-36.

[12] H. Zou, K. Wang, W. Mao, B.Wang, S. Focant, K. Handekyn, D. Chantrain, N. Marly "Prototyping SIP-based VoIP services in Java", Proc. of WCC - ICCT