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Low Cost VoIP Architecture Using Open Source Software Component in Tertiary Institutions

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Abstract— Governments and their agencies are often challenged by high cost and flexible telephonic, Web based data services. Emerging technologies, such as those of Voice over Internet Protocol (VoIP) that allow convergent systems where voice and Web technologies can utilize the same network to provide both services, can be used to improve such services. This paper describe VoIP system for the enterprise network (e.g. company, university) that have been developed based on Asterisk which is a kind of open source software to implement IP-PBX system. Through the development and evaluation, we have confirmed that VoIP system based on Asterisk is very powerful as a whole and most PBX functions to be required for the enterprise network can be realized. Interesting findings include that the University of Kufa has a potential to implement the project. By connecting multiple Asterisk servers located in different sites based on IAX2, large scale enterprise network can be developed. Since the software recommended for installation is open source, the project could be used as a source of valuable information by students who specialize in real-time multi-media systems.

Index terms — Open source VoIP; VoIP Planning, VoIP implementation.

1.Introduction

From several years ago, wireless networks have been achieving great popularity because of low cost deployment of these networks, while provide us quite mobility and scalability [1]. These networks have evolved quickly to meet the needs of the users: more security and bandwidth. Therefore, it is evolved from IEEE802.11b, which is used to supply a theoretical bandwidth of 11Mbps, to IEEE802.11a and IEEE802.11g which provides a theoretical bandwidth of 54 Mbps [2]. Although IP telephony was firstly deployed for the wired network, it can also be deployed on the wireless network. This technology is always in progress. IP telephony, or VoIP (Voice over IP), enables voice communications over networks based on the Internet Protocol (IP). Therefore, it allows significant advantages. One of the main advantages of the IP telephony over a wireless network is that it allows mobility of the people while they are talking.

On one hand, IP phones communications within the intranet are free. This is most interesting for companies or institutions which have several branches or for mobile employers who are moving inside the intranet with their mobile devices. On the other hand, the huge investments that have to be done by the companies or institutions to purchase a Private Branch Exchange (PBX) can be reduced by using PBXs based on free software. They provide the same functionality as a traditional PBX. Currently, most PDA's and Smart Phones incorporate Wireless LANs connectivity that is a considerable advantage because such devices can be used as either cell or VoIP phone. It is a very important feature because, as long as we have WLAN coverage we will be able to make VoIP calls [3].

2. Voice Transport Network

The delivery of voice over IP networking infrastructure generally uses a number of advanced routing protocols and technologies. In this section, some of real-time audio communication is discussed. The Internet phone or voice over IP is an example of application of the real-time interactive audio/video, where the people communicate with one another in real time. Video conferencing is another example that allows people to communicate visually and orally.

VoIP is the service of making face to face communication with more flexibility over data networks versus legacy PSTN (Public Switched Telephone Network). Although deploying VoIP has considerable advantages but implementing VoIP in actual networks has some problems that should be considered.

VoIP gives its users the ability of real time communication through using the internet. VoIP reduces the cost of call transmission by passing voice and video packets through the available bandwidth for data packets. In this service users could initiate their communication from any location. Other services such as voice and video conferencing, text chat, caller ID, voice mail, call forwarding and other features could be achieved using VoIP. Using this service is simply with software phones, phones with adapters or mobile phones. VoIP could be simply deployed and is extensible and interoperable with other services [4].

Using VoIP advantages needs the recognition of the problems that exists in its deployment in real networks. QoS (Quality of Service)— Mechanism in IP and other data networks that allows some data flows to have higher priority over other flows that share the same communications line or device. It is one of the most important problems in transferring voice over IP. Legacy IP networks are designed for non-real time applications. These applications aren't sensitive to packet delay, jitter or packet loss and can work at current best effort networks with no problem. However, transferring voice and video needs certain level of QoS. IP networks should be enhanced by mechanisms to support real time traffic. Bandwidth, delay, jitter and packet loss directly impact on the VoIP quality. Security is another one of the VoIP problems. VoIP service is vulnerable as other services of the internet. Denial of Service (DoS) is a way of attack to VoIP servers by overwhelming the proxy server with invalid INVITE packets. INVITE packet is sent for requesting initiate a call. There are some techniques for identifying these invalid packets and discarding them [5].

Man in the Middle (MitM) is an attack in which the VoIP packets are stored, changed and transmitted to anywhere attacker wants. An effective solution against this attack is encryption as used in SIPS. Reliability, availability for users and simple extensibility of service are also problems that should be considered in enhancing the VoIP service [6].

3. Open Source VOIP

The concept behind open-source software is that not only is the application available to the general public at no cost, but the actual source code that created the application is also available and freely distributable. This allows other users of the application to make changes to the source code and optionally pass these changes on to other users of the software.

Asterisk is basically a telephony toolkit enabling developers to create numerous types of applications that interface with telephone networks. The most obvious application is that of a PBX. Asterisk can also be used as an IVR (Interactive Voice Response) system, for teleconferences and as a voicemail system. These functions can also be combined to create a powerful multi-faceted telecommunication system. Asterisk and the other components of TrixBox, however, are released under the GPL (General Public License) that comes with the caveat, so that when someone makes changes to the original code and decides to distribute the modified version they have to provide it under the same license. There is no obligation to distribute the source code, but if we must to choose release it under the GPL [7].

Asterisk is, however, most commonly used to build hybrid PBX systems that utilize modern PCI cards instead of banks of switches and relays, and software instead of custom hardware. By using relatively simple PCI cards in a standard x86 computer system running on Linux, the cost to build a working system is greatly reduced as compared to the often expensive and inflexible traditional PBX.

The architecture of Asterisk have been shown in Fig. 1 where the Channel portion consist of various logical communication interface modules and the Application portion consist of the additional PBX service modules. The following sections describes the main modules of the Channel and Application portions [3].

3.1 Channel Modules

Private Branch Exchange (PBX) channel modules consist of various logical communication interface modules which enable Asterisk to communicate with different applications [8]:

1. **DAHDI** (Digium/Asterisk Hardwae Device Interface): to connect with the ordinary existent telephone terminal it is necessary to insert the telephony card (e.g. telephone card of Digium or of Voicetronix) as the physical interface and then the DAHDI interface module will be used. In case of connecting with existing POTS (Plain Old Telephone Service), FXS (Foreign eXchange Subsciber) and FXO (Foreign eXchange Office) interfaces will be used. In case of connecting with ISDN terminal it is necessary to insert the extension card as the physical interface.
2. **SIP**: This is the most basic signaling protocol to perform call processing in Asterisk and RTP/RTCP that are used in order to transmit user data (e.g. voice data).
3. **IAX2** (Inter-Asterisk eXchange2): It is Asterisk proprietary protocol to connect with multiple Asterisk servers located in the different sites. The same port (i.e. 4569 as the default port) is used to transmit the call control signal and voice data.

3.2 Application Modules

Private Branch Exchange (PBX) serve many applications as well as VoIP as shown below [9]:

1. **Voice Conference**: The voice conference service in Asterisk is called as “MeetMe”. User can join the conference by inputting the designated number as the service number.
2. **IVR** (Interactive Voice Response): The automatic voice response can be performed by integrating voice response data file and dial number plan
3. **AGI** (Asterisk Gateway Interface): It is an API to connect the outside program with Asterisk in order to include some additional functions. Various programming language (e.g. C, Java, Perl, PHP, Bone Shell) are supported.
4. **SLA** (Shared Line Appearances): Multiple telephone terminals can share a subscriber line.

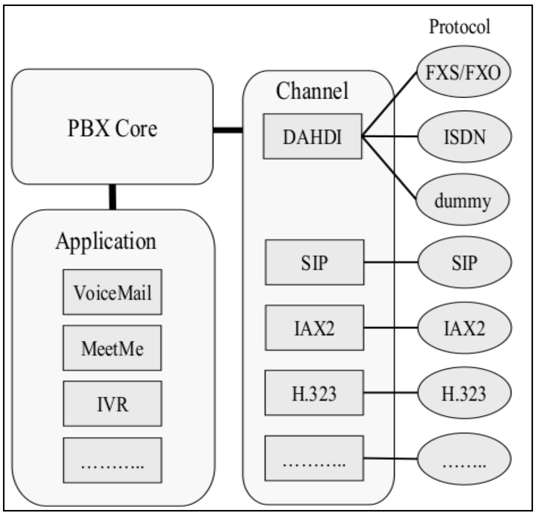

Figure 1: Asterisk Architecture.

4. VoIP Planning

Deployment of new applications and technologies in an enterprise network is always a challenge for all the network administrators and managers. Before deployment, there is need to carry out good planning, analysis and assessment of the current communication and network environment. Then an evaluation and purchase of hardware and software is conducted based of the assessment of the current environment [10].

In cases where consideration is being made to put together voice and data on the same network, a need to ensure that the existing networks can take on this additional load is necessary. This will require an analysis of the current network for congestion and a plan for bandwidth in the case of WAN links [10].

VoIP can be implemented using several methods. One way is to find a vendor who supplies a commercial software product and equipment. Another way is to implement it internally through the use of OSS (Open Source System) such as Asterisk. Various configurations and design can then be made to it in order to customize the system.

Planning is an important aspect to consider for successful implementation of VoIP. Further, Walker and Hicks emphasizes on planning, analysis and assessment of current data and voice networks in order to make projections for the VoIP model. This assists in determining the kind of management of hardware and software resources that would be involved for the purpose of continuity and enhancement [11].

The University of Kufa consists of the main Campus in AN-Najaf province -Iraq and three colleges around the campus. The University has 17 colleges and each college has its own IT center as well as main IT. Fig. 2 shows University of Kufa network infrastructure.

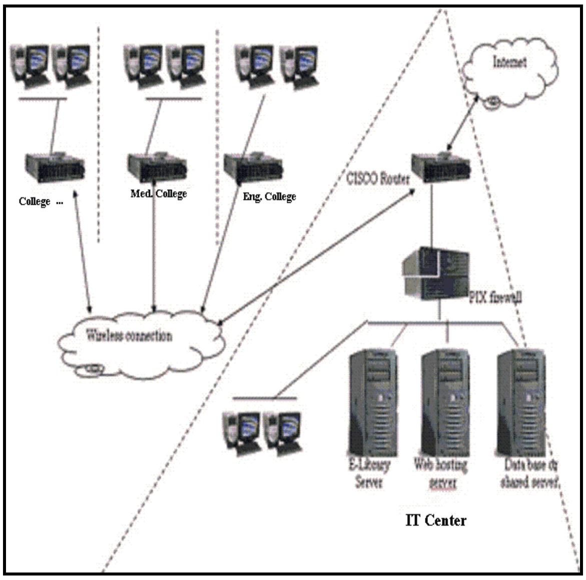

Figure 2: Infrastructure network of Kufa University

A soft copy of the call detail cost was collected and analyzed for a period of (June, 2008 to April 2011) per one month for UOK. The statistical studies of the cost of telephone calling are increasing through advent years, (see Fig. 3).

5. VOIP Implementation

VoIP service is based on signaling protocols for controlling communication process. SIP and H.323 are two important signaling protocols of VoIP. SIP is preferred because of easy implementation, high scalability and easy integration with many other IP services.

In SIP-based VoIP architecture, there are two main elements are user agent (UA) and SIP servers. UA is the end user which sends or receives SIP requests. The SIP servers include proxy server, registrar and redirect server. Proxy server establishes calls between users. It receives requests and resends them and changes fields of packets if necessary. Registrar server registers users with their information when receives a REGISTER request. Redirect server redirects URL of the end proxy server to the client which initiated request. This server is used in extended VoIP networks to prevent loops [10].

VoIP interoperability with PSTN network needs gateways or a soft switch for translating voice packets over IP to voice signals of PSTN based on related protocols.

Some companies have offered these tools as packages. For example ShoreTel [12] and Procurve have such products. Generally deployment and implementation of these packages are costly. Also, there are low cost services, like Skype which have public servers. Some providers aren't eager to use such services because of security problems in using public servers.

Fig. 4 shows the VoIP system that we have developed by using an Asterisk in the Intranet environment of University of Kufa (i.e. enterprise network). In Fig. 4 all telephone terminals are connected to one Asterisk server, but it is possible to use multiple Asterisk servers depending on the scale of the Intranet (i.e. the number of terminals). As the IP phones, Cisco PAP2T have been accommodated which, enables high-quality feature-rich voice over IP (VoIP) service through the broadband Internet connection, Dual standard telephone ports for use with analog phones or fax machines, with independent phone numbers, Compatible with all common telephone features, such as caller ID, call waiting, and voicemail (see Fig. 5).

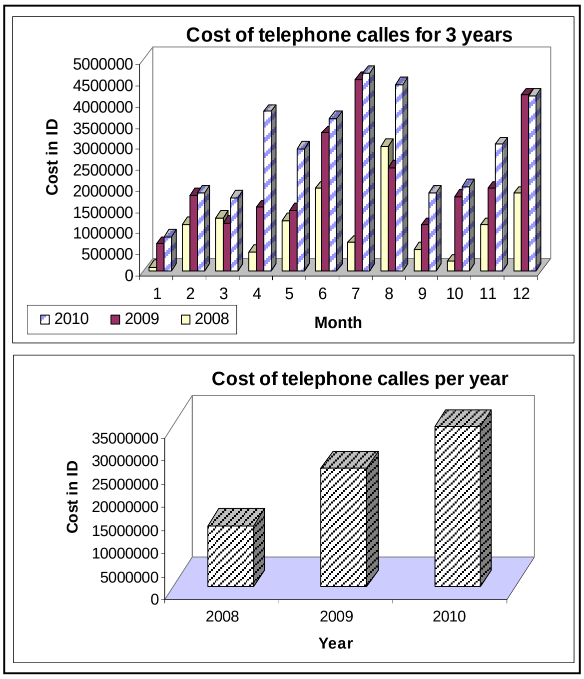

Figure 3: Telephone calling cost.

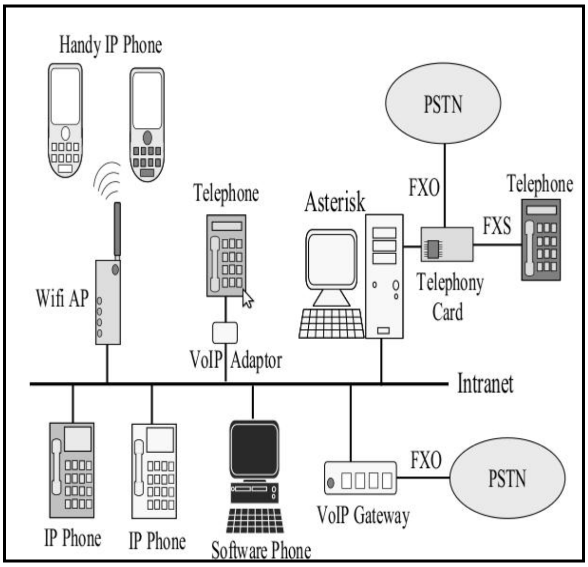

Figure 4: VoIP system development in intranet.

Figure 5: Cisco PAP2T

6. Conclusions

There are different ways for implementing VoIP. The differences are in cost, security, QoS, scalability and reliability. In this paper a reliable low cost structure was described.

The contribution of this paper is the outcome of a study from examining how, Asterisk, an open source VoIP software can be deployed to serve the needs of an educational institution. University of Kufa is the case study of the educational institution, which is currently using a conventional Private automated Branch Exchange (PABX) system for voice and fax communication services, as well as the local area network connected to Internet for Web and data services. Compared with the general SIP server, it can be said that Asterisk is more focused on providing basic functions. But Asterisk can connect with SIP server easily, so it is possible to implement the necessary additional functions by just connecting with other outside SIP servers. Also Asterisk can connect with the existent PSTN by using FXO telephony card, so it is possible to be used as the VoIP gateway.

It is possible to connect multiple Universities using Asterisk servers (see Fig. 5). So it is possible to connect Iraqi University which has Intranet connection together.

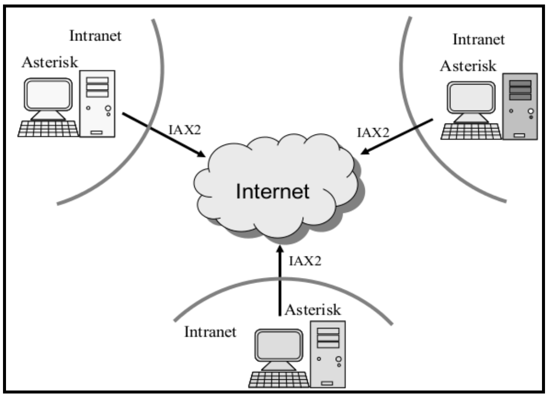

Figure 6: Connecting with multiple Asterisks.

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