"DEPLOYMENT OF DEPARTMENTAL INTERCOM USING IP TELEPHONY"

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ABSTRACT

The past decades we have witnessed a great progress towards applications of the wireless communication. It includes speech/ music transmission, computer related voice/music transmission. Wireless communication have been used in telephony such as cordless telephone, mobiles and Voice over IP (VoIP) is one of the most exciting new developments emerging within the telephony market. We are aware about the traditional PBX system. Traditional PBX systems can be replaced by IP PBX system in a cost effective way. Voice over IP (VoIP) has been implemented using different useful tools. One of the most efficient open source PBX systems is Asterisk PBX. Asterisk is released as open source under the GNU general public license (GPL) platform for IP PBX deployments. This paper provides architecture and deployment of IP PBX system using Asterisk.

Keywords: wireless communication, VoIP, wireless LAN, PBX, IP phone.

1. INTRODUCTION

Telephony history includes PSTN (public switch telephone network) which is based on circuit switching and data exchange network. It is very difficult to handle voice and data packets independently over the PSTN. The next trend developed in telephony to utilize all network resources is Voice over IP (VoIP) or IP telephony. It is used to transmit phone calls over the data network using Internet Protocol (IP). The term VoIP, not necessarily related to the internet; the protocol can be used on intranets and local area networks, in addition to both private and public wide area networks. IP-PBX is proposed system that includes the development of IP and soft switch which makes it possible to connect internet and telephone network.

2. IP-PBX SYSTEM

Deployment of departmental intercom using IP telephony includes configuration of IP-PBX server which is considered as soft switch that uses packet switching technology. The second necessary requirement is one or more internet phones. Its working principle includes registering the clients i.e. soft phones to the server using shared wireless LAN. When client needs to call, it sends connection request to the server. Server has all SIP address of the users which is used to connect an internal call or external call using VoIP service.

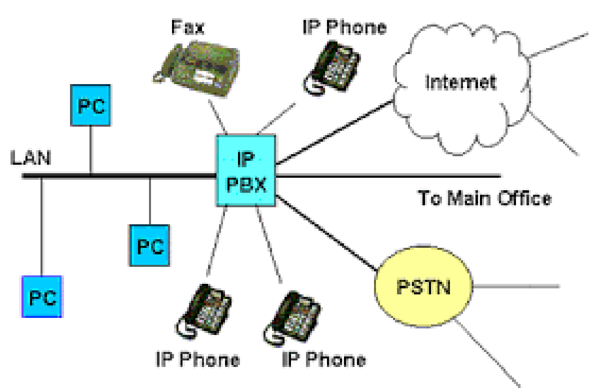

Figure 1: Architecture of IP PBX

Following are the advantages of IP-PBX over traditional PBX system.

- IP-PBX system is easy to install & configure as it is based on IP protocol.
- IP-PBX system provides significant cost savings using VoIP and Eliminate phone wiring in the case of addition of new phones in the network.
- Traditional PBX system requires external devices whereas the IP-PBX system is highly integrated.
- IP-PBX system is highly user friendly because of GUI compared to traditional PBX system.

3. ASTERISK PLATFORM

Now a day's most of the businesses are migrating to IP-PBX system instead of traditional PBX system because of low cost and high reliability. One of the most reliable IP-PBX is Asterisk Platform. Asterisk is released as open source under the GNU general public license (GPL) platform for IP PBX deployments. It provides all the features of the PBX using software and makes use of the LINUX environment. Commercial licensing is available from Linux Support Services, Inc. (http://www.linux-support.net) for applications in which the GPL is inappropriate.

Unlike many modern "soft switches", Asterisk can use both traditional TDM technology and packet voice (Voice over IP and Voice over Frame Relay) protocols. Calls switched on TDM interfaces provide lag-less TDM call quality, while retaining interoperability with VoIP packetized protocols.

To understand the basic difference between VoIP calls and ordinary phone calls, it is important to compare them with each other.

Packet Switching	**Ordinary Telephony**
• Standardized protocols and packet Formats • Very limited internal state • No session state in the network • Services can be added by anyone • No central control • Unclear what the role of operators is (or even who is an operator)	• Standardized interfaces • Lots of internal state (i.e., each switch & other network nodes) • Built in services by the network (hard to add new services) • Centralized control • Clear operator role

Table 1: Comparison between VoIP and Ordinary Telephony

The characteristics of Asterisk are [8],

- Supports for traditional analog telephony devices and digital telephone equipment.
- Supports for major VoIP protocols such as SIP, H.323, MGCP etc.
- Its own specific protocol IAX2 is used to communicate between the server and Asterisk.

3.1 Session Initiation Protocol (SIP)

Session Initiation Protocol is the IETF's standard for establishing VOIP connections. It is an application layer control protocol for creating, modifying and terminating sessions with one or more participants

[18]. The architecture of SIP is similar to that of HTTP (client-server protocol). SIP has INVITE and ACK messages which define the process of opening a reliable channel over which call control messages may be passed. SIP makes minimal assumptions about the underlying transport protocol. This protocol itself provides reliability and does not depend on TCP for reliability. SIP depends on the Session Description Protocol (SDP) for carrying out the negotiation for codec identification. The services that SIP provides include [19].

- User Location: determination of the end system to be used for communication
- Call Setup: ringing and establishing call parameters at both called and calling party
- User Availability: determination of the willingness of the called party to engage in communications
- User Capabilities: determination of the media and media parameters to be used

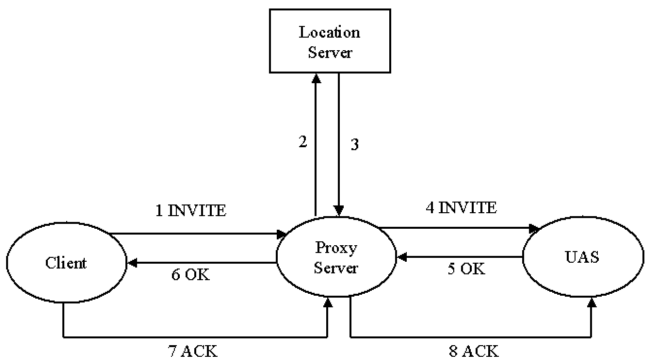

Figure 2: Sample SIP operation [19]

3.2 H.323 Standard

This is the ITU-T's (International Telecommunications Union) standard approved in 1996 used to promote Voice over IP service. H.323 is considered to be the standard for interoperability in audio, video and data transmissions, Internet phone and voice-over-IP (VoIP). It provides capability of addressing call control and management for both point-to-point and multipoint conferences as well as gateway administration of media traffic, bandwidth and user participation.

3.3 Media Gateway Control Protocol (MGCP)

MGCP is used for migration from PSTN to IP telephony, ISPs, and carriers by converting TDM circuits into voice packets. It is a protocol that defines communication between call control elements (Call Agents) and telephony gateways [2]. Protocol operates between a Media Gateway (MG) and a Media Gateway Controller (MGC), also known as Call Agents or Soft Switches, allowing the Media Gateway Controller to control the Media Gateway. MGCP is considered as part of the convergence movement that brings voice and data together on packet-switched Internet.

3.4 Supporting Protocols

SIP works in conjunction with RSVP (Resource Reservation Protocol), RTP/RTCP (Real-time Transport Protocol), RTSP (Real-time Streaming Protocol), SAP (Session Announcement Protocol) and SDP (Session Description Protocol). RTP/RTCP is used for transporting real time data, RSVP for reserving resources, RTSP for controlled delivery of streams, SAP for advertising multimedia sessions and SDP for describing multimedia sessions [19]. H.323. too works in conjunction with RTP and RTCP (Real-time Control Protocol). The present day voice gateways usually compose of two parts: the signaling gateway and the media gateway. The signaling gateway communicates with the media gateway using MGCP (Media Gateway Access Protocol). MGCP can interoperate with both SIP and H.323.

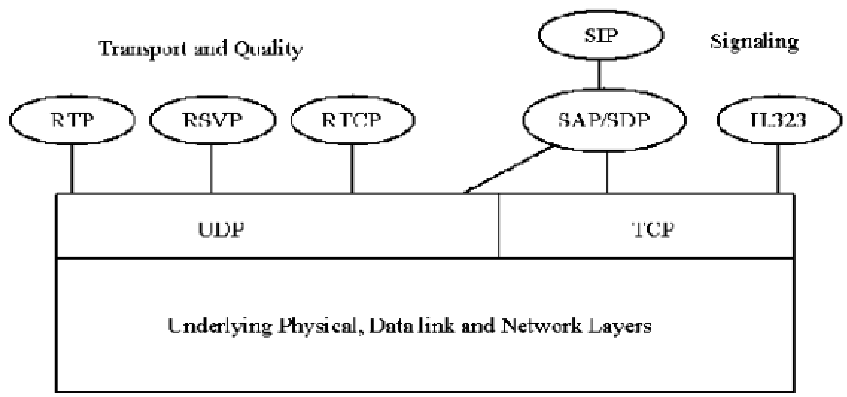

Figure 3: Supporting protocols for H.323 and SIP

4. ASTERISK PBX ARCHITECTURE

Asterisk architecture is fundamentally very simple and configured as the core of an IP or hybrid PBX, switching calls, managing routes, enabling features, and connecting callers with the outside world over IP, analog (POTS), and digital (T1/E1) connections. Asterisk runs on Linux, Mac OS X, Open BSD, FreeBSD and Sun Solaris and provides all features that are available in the traditional PBX system. Asterisk Telephony applications include features such as call bridging, conferencing, voicemail, auto attendant, custom IVR scripting, call parking, intercom and more. Asterisk has been carefully designed for maximum flexibility. Most of Asterisk's usefulness and flexibility come from the applications, codecs, channel drivers, file formats, and more, which plug into Asterisk's various programming interface [12].

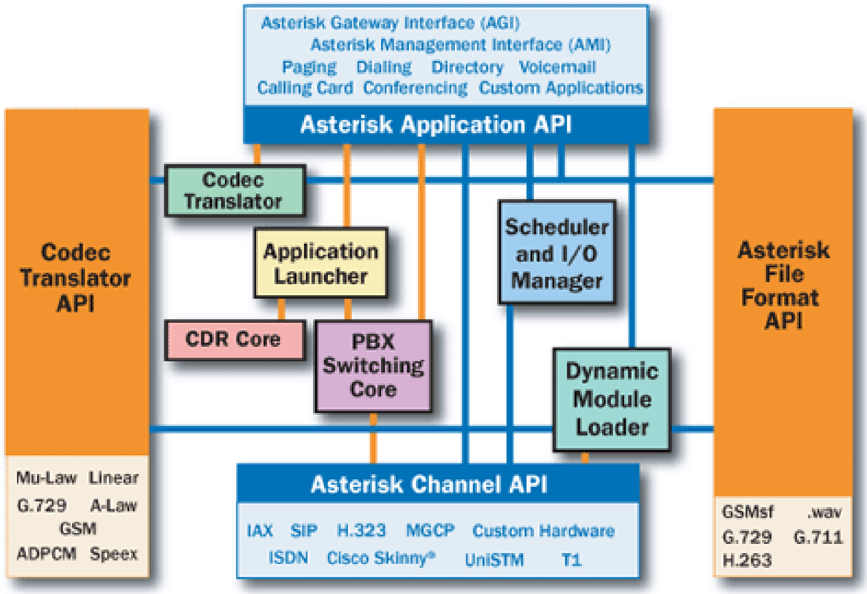

Figure 4: Architecture of Asterisk PBX

5. EXPERIMENTAL RESULTS

In this section we present measurements of various parameters of proposed Voice over IP-PBX system. Voice over IP (VoIP) is susceptible to network behaviors, referred to as delay and jitter, which can degrade the voice application to the point of being unacceptable to the average user. Prior to deploying VoIP applications, it is important to assess the delay, jitter, and packet loss on the data network in order to determine if the voice applications work. The delay, jitter, and packet loss measurements can then aid in the correct design and configuration of traffic prioritization, as well as buffering parameters in the data network equipment.

5.1 The Goals of the Measurements

The goals of the measurements were four-fold:

- To gain understanding of the delay behavior of IP Voice;
- To measure the overall packet loss in VoIP links;
- To measure the total end-to-end delay (max. delta) including the processing delays at the workstations;
- To test the maximum inter-arrival jitter estimation of RTP in a new environment: IP Switching.

5.2 Choosing the Measurement Environments

In a VoIP PBX system, the main components are soft phones, Asterisk Call Manager and Wireless LAN. The Asterisk Call Manager software is the call-processing component of the Unified Communications system. It is an open source IP telephony call-processing solution. We are interested in IP voice delay, delay variance characteristics, maximum jitter and packet loss. These parameters are strongly affected by the type and amount of other traffic in the network. And hence we tried to put some load on our IP PBX server by originating more number of calls.

In this experiment, six IP PBX extensions (5000 to 5006) are created in the Asterisk server database that means six users are created in to the PBX system. A real-time VoIP traffic is generated by originating more number of VoIP calls using soft phone software named X Lite/CSipSimple. It is VoIP Telephony software which is not designed to simply duplicate the traditional telephone system. It is a suite of instant messaging and internet telephony software. PBX users can send and receive voice calls to or from other users on the Wireless LAN. In this test case, three calls are originated one after other.

A simple test bed was setup to transmit the voice packets through Asterisk Call Manager which is installed on Asterisk server computer connected to the Wireless LAN. One client is used to initiate calls and the other one receives calls on Wireless LAN with the help of soft phones.

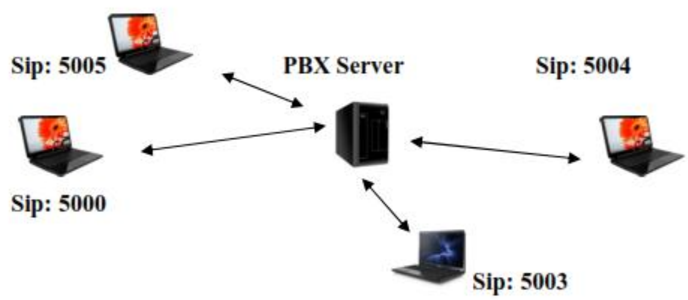

Figure 5: Measurement test bed setup

5.3 Data Capture and Analysis Tools

Wireshark is the primary protocol analysis software used to decode the trace file. From the decoded data, call setup processes and QoS parameters such as Maximum Delay, Maximum Jitter, Packet Loss and protocol mix of VoIP traffic can be determined. Wireshark is a popular network analyzer widely used by network professionals for troubleshooting, analysis, software and protocol development, and teaching. It reads packets from either the network or a trace file, decodes them, and presents them in an easy to understand format.

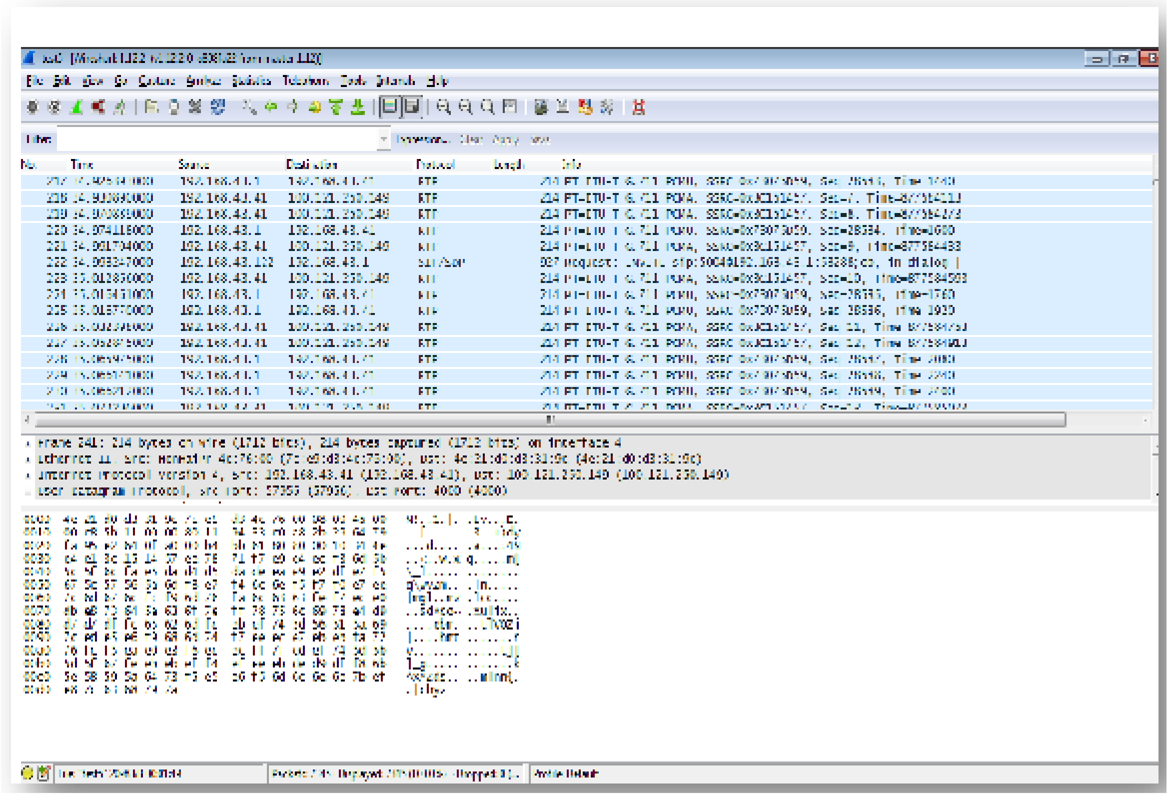

Figure 6: Wireshark captured file

Wireshark was chosen as the tool to use because it is an actively-maintained open source program and its graphical user interface is easily configurable and easy to use.

Following are the primary features to be utilized from Wireshark:

- It can capture data from the network or read from a captured file.
- It supports Tcpdump format capture filters.
- It runs on over 20 OS platforms, both UNIX-based and Windows.
- It supports over 480 protocols, and because it is open source, new ones are contributed very frequently.

5.4 Results: In this section, experimental results of test bed have been discussed along with the statistics are presented in detail.

5.5 Statistics

The information in the trace files of Wireshark is summarized in Table 2. In this experiment, a trace of three calls has been collected with approximately two minutes of voice data for each call. The traffic was captured from all the three bidirectional calls.

Attributes	Measured Values
Max. Delta	68.74 ms
Max. Jitter	13.23 ms
Lost RTP Packets	0 (0.00%)

Table 2: Overall statistics for the trace

CONCLUSION AND FUTURE SCOPE

The performance of IP PBX system is monitored on real-time basis and the performance (QoS) parameters are measured experimentally and are compared with the standard QoS parameters. Thus we can conclude that the measured values of QoS parameters are within the specified limits and hence proposed IP PBX system can be implemented in real-time environments. Lots of additional services can be implemented in the existing setup. These services may include interconnection with PSTN and Internet, advanced speech, web, and streaming video capabilities, Interactive Voice Response (IVR) services and many more. The total number of extension also can be increased with the help of dedicated PBX server.

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