Integration of Open Source and Enterprise IP PBXs

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Abstract— **Over the last few years, Voice over Internet Protocol (VoIP) evolved from enabling voice communications between computer terminals to providing much wider functionality of the Public Switched Telephony Network (PSTN). The idea of deploying VoIP in Private Branch eXchanges (PBXs) is gaining credence among service providers, and this paper addresses some of the design issues related to an IP PBX implementation. Specifically, the paper explores the interoperability between Cisco Call Manager (CCM) and Linux-based Asterisk PBXs corresponding to the enterprise-class and Open Source IP PBXs, respectively. The architecture of the networking testbed is documented for running various VoIP protocols between multi-vendor devices and IP PBXs. The results presented can be broadly defined into three areas of investigation: (i) the use of Session Initiation Protocol (SIP) and Inter Asterisk eXchange (IAX) trunks to integrate CCM to Asterisk PBXs and Asterisk to Asterisk PBXs, respectively; (ii) examining interoperability between the CCM and Asterisk PBX with advanced features like Call Display, Voicemail, Call Detail Records (CDR) and Voicemail to e-mail transfer; and (iii) testing the resilience of voice quality against packet loss and delay in the Open Source PBX trunks with National Institute of Standards and Technology (NIST) network simulator.**

Keywords-Component; VoIP,PSTN,PBX,SIP,IAX,H.323,CCM,Asterisk PBX, NIST-Net.

I. INTRODUCTION

Convergence of computer and communication industries in the 1990s resulted in a number of new and disruptive technologies such as VoIP. Initially, VoIP provided voice communications over generic packet switched Internet with specialized software like Net meeting running on each computer terminal to establish voice communication. In the last few years, VoIP deployment entered into even wider commercial aspects of modern telecommunications by being considered as a replacement for legacy PBXs in the conventional PSTN. VoIP typically uses two types of protocols: signalling and media transfer. Signaling protocols are used to set up a voice conversation while the media transfer protocols are used for the transfer of voice traffic once the session is successfully established [1].

PBX, short for Private Branch eXchange, is a private telephone network that is used within a mid-size company to connect to PSTN using a reduced number of lines; i.e., users of the PBX share a certain number of outside lines for making telephone calls external to the PBX. The premise for the development of PBXs was that the majority of telephone calls were staying within the company and only a small proportion of calls are connected to the PSTN. Most medium-size and larger companies use a PBX because it is less expensive than connecting an external telephone line to every telephone in the organization. In addition, it is easier to call someone within a PBX because the number you need to dial is usually just 3 or 4 digits.

With the VoIP revolution, some vendors of the networking equipment realized that there was a potential of entering telecommunications equipment PBX domain and they introduced an enterprise class of PBXs using VoIP technology, which in this paper we refer to as Enterprise IP PBX. Cisco with its Call Manager and Unity solutions is an example of this paradigm shift in the PBX implementation space [2]. It did not take much time to introduce an Open Source converged telephony platform to implement IP PBX designed primarily to run on Linux, hereafter referred to in this paper as the Asterisk PBX [3]. Before serious deployments of the Asterisk PBX, the integration of the enterprise and Open Source PBXs is of great interest to the telecommunication industry community. It is because of this interest that the interoperability issues in CCM and Asterisk have been chosen for investigation in this paper. In particular, we configure Asterisk implementations of Open Source PBXs, Cisco Call Manager and Unity boxes to be interoperable between themselves and PSTN. This implementation provides comparable intelligent services feature sets like Call Display and Voicemail. This paper also integrates two Asterisk PBX servers to examine the scalability of Asterisk-based PBX solutions. Integration of the PBXs explored in this paper cover (i) the various ways of setting up trunks between the PBXs and (ii) configuration of the effective and functional trunks in the testbed. Additionally, the resilience of a VoIP PBX solution measured through voice quality for different Packet Loss Rates (PLRs) and delays is tested on the Open Source PBXs. The testing of the setup is conducted at Dalhousie University using equipment within the Advanced Internetworking Laboratory (AIL) environment developed through the Canada Foundation for Innovation (CFI) grant [4].

II. VOIP TECHNOLOGY

The section provides an overview of the standards and technologies related to the testbed presented in this paper. Specifically, subsection II.A discusses different VoIP call control and signalling protocols. This is followed in II.B by an explanation of the various types of VoIP devices used in this project. Subsection II.C describes configurations of CCM and Asterisk PBXs.

A. VoIP Protocols

The two standardized VoIP protocols are H.323 and Session Initiation Protocol (SIP). The lightweight nature of the SIP Protocol is the key reason why, in recent years, this protocol has been the subject of extensive research. SIP is used with Real Time Transport Protocol (RTP) for real time media transfer in the VoIP communication. Recently, IAX (Inter Asterisk eXchange Protocol), a third VoIP protocol, has gained attention in the Open Source community [5]. IAX has not been standardized yet by the IETF. The primary design goals for the IAX protocol are:

- To provide native support for NAT transparency.
- To minimize bandwidth usage for both control and media with specific emphasis on individual voice calls.

IAX is a peer-to-peer media and signalling protocol. With respect to media, the sequencing and timing information is included in IAX frames. The IAX protocol does not use the RTP for transport of media and is thus different to SIP. In IAX, both signalling and media share the same UDP port number and hence IAX does not suffer NAT traversal problems that are associated with SIP. IAX is a binary protocol. This design choice was made for bandwidth efficiency. The IAX protocol is used to create trunks between the two Asterisk PBX servers while the CCM version 4.1 and Asterisk PBX were integrated with SIP trunks to provide seamless communication between the two IP PBX classes (Enterprise and Open Source) in the testbed. [6]

B. VoIP Devices

A wide variety of VoIP devices have been deployed in the testbed, such as soft phones, Analog Telephone Adapters (ATA), VoIP phones and Wi-Fi VoIP phones. The specific models and software used in each of these categories are listed in Table 1.

TABLE I. VOIP DEVICES IN THE TESTBED.

Soft phones	ATAs	VoIP phones	Wi-Fi VoIP phones
Virbiage Firefly, X-Pro Pocket PC, Pocket PC IAX	Linksys PAP2	ATCOM SIP phone, Cisco 7970	Cisco 7920, UTStarcom F1000.

In addition, the testbed includes Cisco Aironet 1200 series access points to provide Wi-Fi access for the Cisco 7920's. The testbed and its setup are explained in Section III with the detailed configurations provided in [7]. This also applies to the IP PBX servers, the most important components of the testbed that all these devices are connected to.

C. IP PBX Servers

An IP PBX delivers dial tone, the ability to conference, Call transfer, and most importantly dials other employees by extension number. It provides standard features of the PSTN like Call Display as well as features specific to VoIP like voice to e-mail forwarding. All of this being possible because voice transmissions are carrying via data packets over a data network instead of circuit-switching in the traditional phone network.

We classify IP PBX servers in two types. Enterprise IP PBX like Cisco Call Manager, Nortel Meridian 61C and Open Source IP PBXs like Asterisk PBX, SIP Express Router and SIPX.

The two PBXs used in the testbed described in this paper are Cisco Call Manager and Asterisk as examples of the enterprise-class and Open Source PBXs, respectively. The paper aims at configuring these two types of IP PBXs for seamless interoperability and integration. Special attention is paid to analyze the performance of these IP PBXs in the presence of IP packet loss emulated using the NIST network simulator.

The CCM requires the Cisco Unity box for the overall system to deliver powerful unified messaging (e-mail, voice, and fax messages sent to one inbox) and intelligent voice messaging. The CCM used in the testbed was version 4.1. Cisco Call Manager serves as the software-based, call-processing component of the Cisco IP Telephony Solution for the enterprise part of Cisco AVVID (Architecture for Voice, Video and Integrated Data). The integration of Call Manager and Unity was already set up in the Advanced Internetworking laboratory prior to the implementation of this project.

The Open Source IP PBX that is used in the paper is the Asterisk PBX version 1.2.1, the latest version when the testbed was originally set up. The latest version of the Asterisk PBX at the writing of this paper is 1.2.14. Asterisk is a complete PBX software solution. It can be run on Linux, Free BSD, Centos, and Debian. It provides all the features that one expects from a PBX. Asterisk implements voice over IP using three protocols SIP, IAX and H.323 and, when using SIP or H.323, RTP is used for audio packet delivery. It can interoperate with almost all VoIP standard-compliant telephony equipment using relatively inexpensive hardware.

Asterisk needs no additional hardware for Voice over IP. For interconnection with digital and analog telephony equipment, Asterisk supports a number of hardware devices, most notably all of the hardware manufactured by Asterisk's sponsors, e.g., Digium. Hardware from vendors like Sangoma and RedFone is also supported by Asterisk. Digium has single and quad span T1 and E1 interfaces for interconnection to PRI lines and channel banks as well as a single port FXO card and a one to four-port modular FXS and FXO card.

The Asterisk core handles these items internally using [5]:

- PBX Switching - the Switching Core transparently connects callers arriving on different hardware and software interfaces. To provide native support for NAT transparency.
- Application Launcher - launches applications required for various services, such as Voicemail, file playback, and directory listing.

- Codec Translator - uses codec modules for the encoding and decoding of audio compression formats used in the telephony industry. A number of codec's are available to suit diverse needs and arrive at the best balance between audio quality and bandwidth usage.
- Scheduler and I/O Manager - handles low-level task scheduling and system management for optimal performance under all load conditions.

Asterisk PBX fits in various places of a conventional soft switch network as a PSTN gateway, conferencing server, voicemail server, music on hold server, soft switch, and POTS (Plain Old Telephony Service) gateway.

III. Architecture and Implementation

The section III.A presents the network design of the testbed followed in section III.B by experimental verification that the testbed is completely functional.

A. *Network Design*

Figure 1 describes the network design and the architecture of the testbed. SIP and IAX extensions are created on the Asterisk PBX servers while H.323 extensions are created on the CCM. Cisco Unity is configured as well to store the voicemails and other related info of extensions created on the CCM.

Figure 1 illustrates the following features of the design:

1. It integrates two IP PBX Systems, CCM and Asterisk PBX using SIP trunks.
2. Two Asterisk PBX Servers are interconnected using IAX trunks.

The testbed uses the following components of which some were discussed in Section II, while the others are discussed in the subsequent sections:

1. Cisco Call Manager (Publisher and Subscriber) and Cisco Unity.
2. Two Asterisk PBX servers connected with an IAX trunk.
3. Cisco 3750 switch and Cisco 3845 router with 2 FXO ports.
4. Cisco Aironet 1200 series access point used by the Cisco 7920's to register to the CCM.
5. Linksys wireless router used by Toshiba Pocket PC and UT Starcom F 1000 to register a SIP client to the Asterisk server and for registration of VoIP phones to Asterisk servers 1 and 2.
6. Cisco 7970's registered to CCM.
7. Linksys PAP2 ATA used to register analog phones as SIP clients to Asterisk servers using the Linksys wireless router.
8. SIP Trunks between CCM and Asterisk servers.

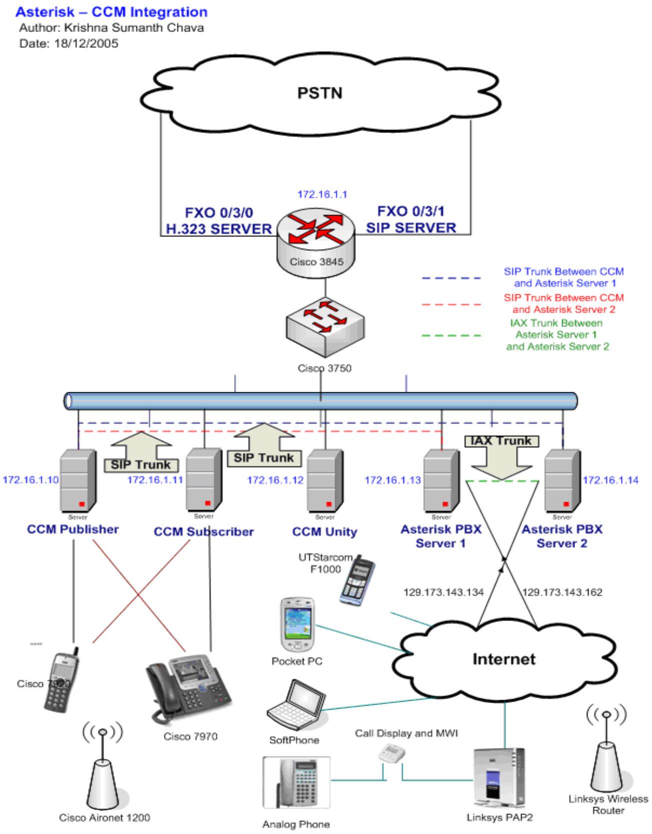

Figure 1. Network design of the testbed.

The CCM and Asterisk PBXs were configured first individually in the testbed using well-documented in the literature procedures [9], while this paper focus is on investigating the setup of trunks to accomplish seamless integration between the CCM and the Asterisk PBXs. Also, the paper discusses the trunk setup between the two Asterisk servers from the scalability perspective. Both CCM and Asterisk PBXs support the SIP trunks, and hence steps were taken in configuring the SIP trunk between the IP PBXs. The two Asterisk servers were integrated using the IAX trunks as being more efficient over using the SIP trunks in the testbed.

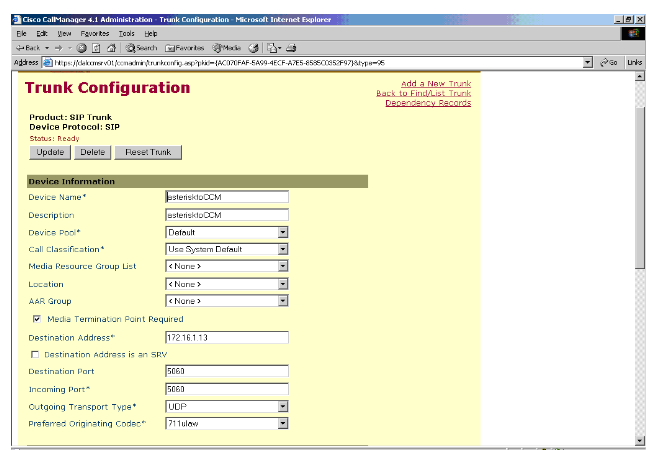

Figure 2. CCM SIP trunk configuration.

The SIP trunk Configuration on the CCM is shown in Figure 2. Asterisk servers send the SIP calls to the SIP UA configured in the Cisco 3845. Similarly, a H.323 gateway is set up on the CCM to send its PSTN calls through the FXO port on the Cisco 3845 router. The Cisco 3845 router is used to terminate the SIP calls from the Asterisk servers to the PSTN.

B. *Testing*

The research objectives in testing the testbed are as follows:

1. To make CCM and the Asterisk interoperable and still provide their advanced features with good voice quality.
2. To analyze the problems that one has to deal with during the integration.
3. To verify the scalability of the Asterisk-based IP PBX solution by integrating two Asterisk PBX servers.
4. To use NIST Net network emulator for imitating the performance dynamics like voice quality, delay, and PLR (Packet Loss Rate) in the IAX trunks between the Asterisk servers, as discussed in detail in Section IV.
5. To document the test results of incoming and outgoing calls between PSTN to Asterisk and PSTN to CCM, and to capture the voice packets at different points in the network.

The IAX trunk state is registered on the two Asterisk servers and hence enables communication between the clients (SIP and IAX) registered to the two servers. Figure 3 shows the state of the trunk as seen from one of the Asterisk servers in the testbed.

The voice quality is checked for all calls. It is observed that PSTN calls cannot be dialed from IAX extensions of Asterisk without using a SIP peer on Asterisk. The reason for this limitation is because the Cisco 3845 router is configured to accept only SIP calls.

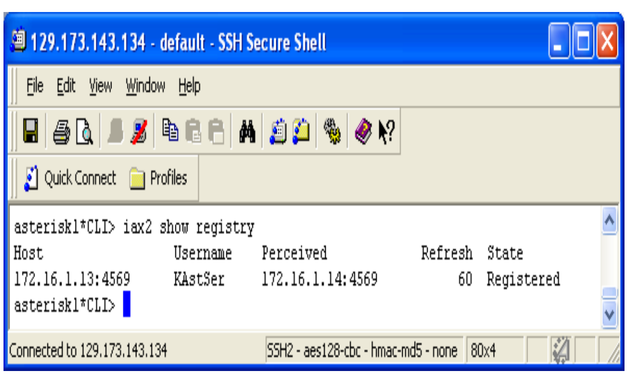

Figure 3. IAX trunk state on the Asterisk Server 1.

The IAX2 packet captured on the IAX trunk when a call is in progress between two SIP clients registered to the two Asterisk servers is shown in Figure 4. A close examination of captured packets in Fig. 4 shows that there are two IP subnets; this is because the specialized PC running NIST Net is acting as a router between the two networks with different IP PBXs to simulate congestion conditions in the realistic networks.

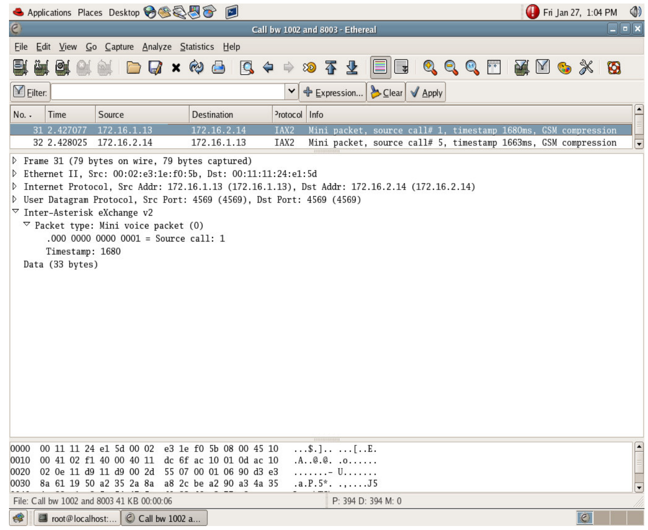

Figure 4. IAX2 packet captured on the IAX trunk.

IV. PERFORMANCE

This section examines the resilience of the VoIP network introduced in the previous section against IP packet transport imperfections, such as delay and packet loss in the IAX trunks, using voice quality parameters. Resiliency in the context of the project is the characteristic of being able to adapt under stress or faults in order to avoid failure. It is an attribute that contributes to achieving the required reliability but is not an independent measure of it [10].

In this section, NIST Net emulator is used in our investigations to drop the packets in real time and examine the voice quality at different Packet Loss Rates (PLRs) [11]. NIST Net emulator is used on the IAX trunk and between Asterisk PBX and its soft phone client. We examined the resilience against delay and packet loss by plying and recording voice messages in the IP PBXs with different choice of parameters for PLR and delay. Specifically, Section IV.A discusses the resilience against PLR using qualitative analysis, while Section IV.B discusses system resilience against delay.

Among many features supported in IP PBXs and verified to operate correctly in this project (e.g., Call Display, Voicemail, Voicemail to e-mail, recording of the calls), this paper documents in Section IV.C obtaining of Call Detail Records (CDR) as an example for the possibility of centralized accounting, billing and auditing functions.

A. *Resilience against Packet Loss Rate*

To verify the resilience against packet loss in Asterisk PBX, the NIST Net is placed on the IAX trunk with Asterisk servers on two different networks and investigated voice traffic being routed through NIST Net. We also placed NIST Net between an Asterisk server and a PC having a soft phone registered to Asterisk. The analysis of voice quality is made using the Mean Square Error (MSE) estimate of an error signal, which is the power of a signal being a difference between original voice signal and the voice signal recorder after emulating the packet loss.

1) On IAX Trunks (Asterisk-NIST Net- Asterisk)

The NIST Net was placed between the Asterisk servers and the packet loss was introduced from 0 to 30%. The voice quality started degrading very badly from 20% PLR. But the call stays connected, which means signalling information (SIP) is being sent. The reason for that is the signalling information is sent in TCP packets and the network stays up till we have some network problems.

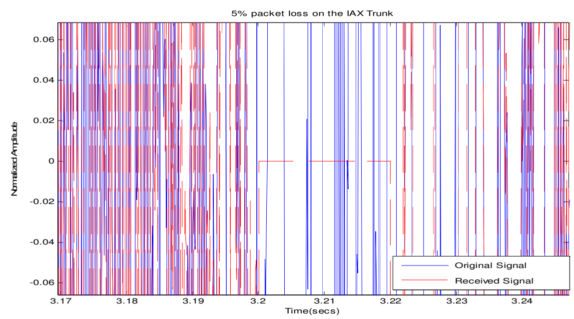

Figure 5. 5% Packet loss on the IAX trunk.

Figures 5 and 6 show time waveforms in Matlab [12] representing the original voice signal (in blue) and the voice signal recorded (in red) after emulating the packet loss on the IAX trunk with PLR scenarios of 5% and 30%, respectively. The graphs clearly visualize the periods when the packets are being lost with the continuous lines at zero level (red waveforms). In addition, Figures 5 and 6 also document some of the problems when analyzing speech signals in the time domain such as sample clipping (dynamic range limitations) and sample inversions which are the result of multiple transcoding of speech signals between different servers with different voice codec's, e.g., GSM and G711U.

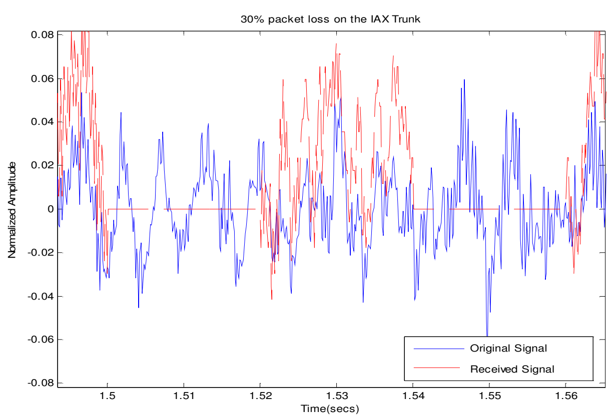

Figure 6. 30% Packet loss on the IAX trunk.

The quantitative analysis for the resilience against PLR is done using the Mean Square Error estimate. A call is sent over the IAX trunk, the packets are dropped, and the calls are recorded on both ends in wav format. The wave files are then read into Matlab to find the mean square error estimate at different PLRs. We calculated the cross correlation and had taken care in the alignment of the wave files with respect to the delay while finding the mean square error estimate. Figure 7 shows the Mean Square Error estimate values in different PLR scenarios for the setup described above. Due to the excessive clipping in Asterisk PBX while recording the calls, the results are not conclusive.

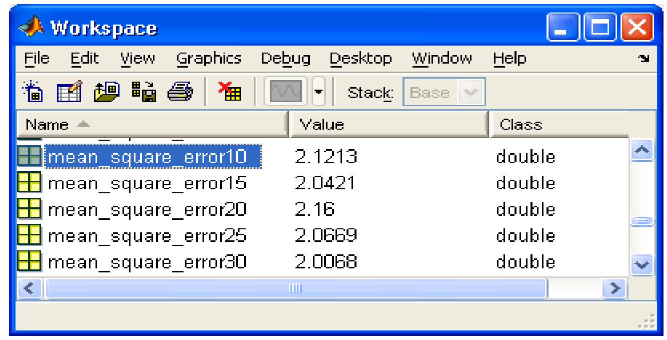

Figure 7. Mean Square Error estimates on the IAX trunk.

Because of this and other limitations of hardware and software tools for speech analysis in the AIL environment, we are unable to provide a more informative analysis of voice quality in the project.

2) Between a soft phone on a PC and a phone under control of the Asterisk server

For network performance analysis in this section, the NIST Net was placed between the Asterisk server and the PC having the soft phone. Packet loss was introduced from 0 to 30%. The voice quality started degrading very badly from 20% PLR, but the call stays connected which means signalling information (SIP) is being sent. The reason for this is that similarly as in the previous section, the signalling information is sent in TCP packets and the network stays up till we have some network problems. The voice can be understood with difficulty at 20% PLR and is understandable comfortably at or less than 10% PLRs.

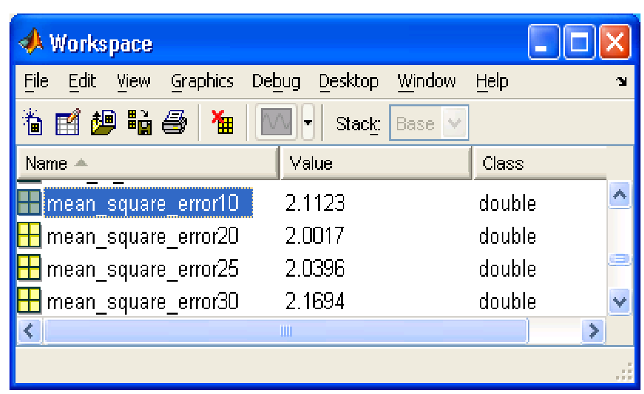

Figure 8. Mean Square Error estimates between soft phone and Asterisk.

The Packet Loss Rate is introduced in the NIST Net, the calls are recorded, and the wave files are read in Matlab to find the Mean Square Error estimate similarly as described in previous section. Figure 8 shows the Mean Square Error estimate values with PLRs set at 10, 20, 25 and 30%. Due to the excessive clipping in Asterisk PBX while recording the calls, the results are not as convincing for the same reasons as discussed in the previous section. Needless to add, the results obtained could be more conclusive had we utilized a Hammer Call Analyzer or some other more sophisticated speech analysis tools that could minimize clipping introduced by the Asterisk servers [13].

B. *Resilience against delay*

Delay is introduced using NIST Net in the two scenarios that we had discussed in Section IV.A. When delay is introduced the receiver hears the caller with a delay at his end. As expected, when a delay of 150 ms or above is introduced in the network, we are getting a noticeable echo during the calls.

C. *Call Detail Records*

The Call Detail Records (CDR) plays an important role in PBX servers for billing purposes. Figure 9 (a) contains the CDR between two Asterisk servers using the IAX trunk, while Figure 9 (b) contains the CDR between Asterisk Server and CCM. The results obtained corroborate that the CCM phones to the Asterisk extensions and vice versa can also be maintained with Asterisk Server.

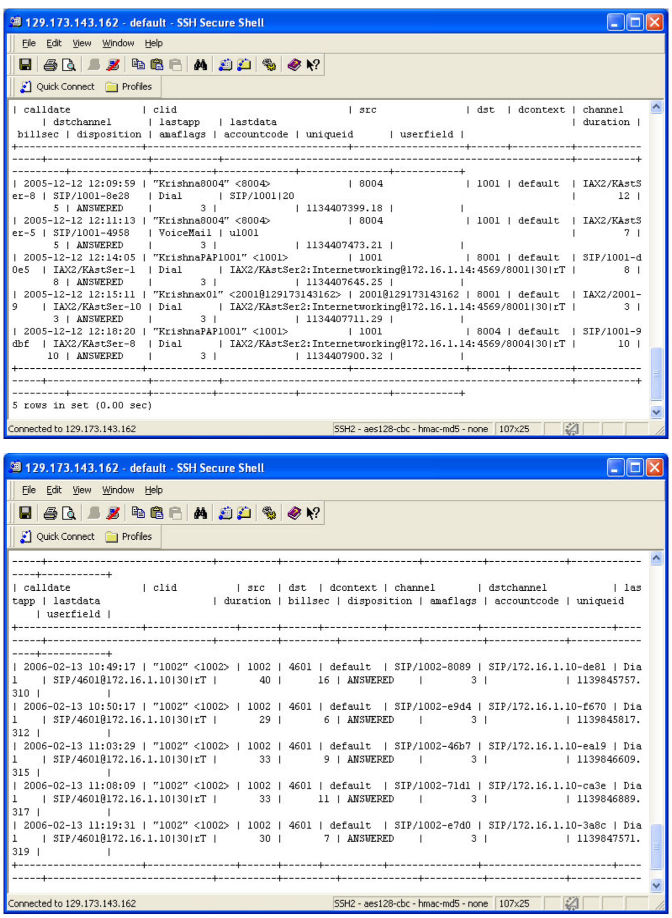

Figure 9. Asterisk Server 1 CDRs with a) Asterisk and b) CCM extensions.

V. CONCLUSION

The paper discussed the VoIP technology in respect to the IP PBXs implementations. The testbed presented has been used to study the interoperability of various IP PBXs / PSTN domain call control and transmission of voice traffic over real-world and emulated networks. Both the enterprise class and Open Source IP PBXs are made operational in the project. An innovative design of integrating the Open Source with enterprise class PBXs is executed successfully, and all the features like Call Display, Voicemail, Voicemail to e-mail, CDR, and recording of the calls are verified to operate correctly. The conclusion reached is that with a proper configuration of the network testbed designed in this project, Asterisk and CCM are interoperable. Also, voice quality analysis was undertaken in the project using Matlab to demonstrate the resilience of voice network designed against packet loss and delay in the system. Some shortcomings of the deployed Mean Square Error Estimate (MSE) methodology for measuring the voice quality were pointed.

ACKNOWLEDGMENT

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REFERENCES

[1] B. Khasnabish, “Implementing Voice over IP”, Wiley, 2003.

[2] D. Bateman, “Configuring Call Manager and Unity”, Cisco Press, 2005.

[3] J. Van Meggelen, J. Smith, and L. Madsen, “Asterisk The Future of Telephony”, O'Reilly Media, September 2005.

[4] N. Aslam, J. Ilow, S. Sivakumar, W. Robertson, “Establishing an Advanced Internetworking Research Laboratory at Dalhousie University”, Proc. 4th Annual Communication Networks and Services Research Conference, 2006. CNSR 2006, pp. 100-103, Moncton, NB, Canada, May 2006.

[5] Asterisk.org, “Features and Architecture of Asterisk PBX”, http://www.asterisk.org/features, accessed in March, 2006.

[6] Internet Draft, “Inter-Asterisk EXchange (IAX) Version 2 draft-mspencer-iax2-01”, http://www.cornfed.com/iax.pdf, accessed in January 2005.

[7] C. Krishna Sumanth, “Integration of Open Source and Enterprise IP PBXs”, Internetworking Program, Dalhousie University, M.Eng project, Apr 2006.

[8] T.Abbasi, S.Prasad, N.Seddigh, I. Lambadaris, “A comparative study of the SIP and IAX VoIP protocols,” IEEE Canadian Confernce on Electrical and Computer Engineering, pp. 179–183, Saskatoon, SA, Canada, May 2005.

[9] Dual Phone System Integration Guide for Cisco Unity 4.0, http://www.Cisco.com/univercd/cc/td/doc/product/voice/c_unity/integuid/multi/itmultin.htm, accessed in Dec. 2006.

[10] A. Takahashi, H. Yoshino and N. Kitawaki, “Perceptual QoS assessment technologies for VoIP” IEEE Comm. Mag., pp.28-34, Vol.42, No. 7, July 2004.

[11] NIST Net Home Page, http://www-x.antd.nist.gov/nistnet/, accessed in Dec. 2006.

[12] Matlab from Mathworks, http://www.mathworks.com/products/matlab/, accessed in Dec. 2006.

[13] VoIP Testing and Monitoring Solutions from Empirix, http://www.empirix.com/, accessed in Dec. 2006