Implement VoIP Based IP Telephony with Open Source Asterisk Architecture

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ABSTRACT

Asterisk is a leading open source telephony software/system, easily implemented over intranet and internet. Asterisk empowers developers and integrators to create advanced communication solutions. An Asterisk system is a low cost type of a traditional PBX system. Any phone controlled by an Asterisk system can call a VoIP or analog phone controlled or managed by a traditional telephone system or by Asterisk telephone system. In this paper, the authors focus on the deployment and testing of various Open Source Asterisk Services in an enterprise level communication system. Selected services are listed in this paper that can be used to implement a telephone system with good Quality of Services (QoS) and good Quality of Experience (QoE) from the personal user to enterprise level users.

Keyword: Asterisk, IP Telephone, IVR, Linux Server, QoS, VoIP

INTRODUCTION

Asterisk is open source telephony Software/System; it easily implemented over Intranet as well as Internet with open source software. Asterisk allows new value added service of Communication for sharing Voice-Video-Data (Spencer, 2002). With its included support for internationalization, rich set of configuration files, and open source code, every aspect of Asterisk can be customized to meet user needs for providing better Quality of Service with low cost.

Asterisk easily adopts new interfaces and technologies for telephony. An Asterisk system is a fraction of the cost of legacy PBX systems; additional hardware that turns a small Linux server into a telephone system is inexpensive and ready available. Asterisk is incredibly efficient. A small PC will serve many telephone users at once adopting Asterisk. With Asterisk you can easily build a telephone system for the smallest or the largest enterprise, there are Asterisk server running thousands of phones right now. You can easily scale or combine Asterisk systems to serve a number of users in any number of locations (Mahler, 2004).

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When combined with low-cost Linux telephony hardware, Asterisk creates a PBX at a fraction of the price of traditional PBX systems. While an Asterisk system is a fraction of the cost of legacy systems, it provides better functionality than the most expensive proprietary systems. Asterisk includes features such as voicemail, interactive voice response IVR, and conferencing which are very expensive in proprietary systems.

**REVIEW OF LITERATURE**

Asterisk Implementation strategy with aspect of small-scale implementation discussed in Alam, Bose, Rahman, and Abdullah Al-Mumin (2007) present some important theoretical and experimental result regarding setting up a VoIP (voice over Internet protocol) server with the well known open source software - Asterisk, this research also enlightening implementation of GnuGk PBX and Asterisk server setup configuration as well. Asterisk Designing and Implementation showing as replacement of EPBX (Qadeer & Imran, 2008), in this research the design and implementation aspects of a VoIP based asterisk voice exchange, developing a fully functional voice exchange requires to set up a server based on Asterisk, connecting clients to that server with the help of soft phones and then configuring the soft phones with the server.

In (Qadeer & Imran, 2008) connection of the clients to the server with the help of IAX protocols also discussed and deployment configuration discussed. Important features of Asterisk deployed in Imran and Qadeer (2009), the services generally associated with an Asterisk based Voice Exchange i.e. conferencing, paging and voice mailing.

**Communication Services Provision with Open Source Asterisk**

Asterisk supports both US and European signalling standard used in telecommunication systems. Asterisk easily connect existing network infrastructure with next generation voice-data integrated networks. Asterisk supports traditional phone equipment as well as new equipment with additional capabilities to connect with a single backbone network.

Asterisk deployed communication infrastructure, able to exchange Voice-Video-Data over Intranet as well as Internet. With additional hardware support Asterisk also connect and share video-voice with POTS without Internet Connection. Calls over Asterisk seem to be much cheaper in cost, better in Quality and much more functionality over the traditional telecommunication system.

Asterisk allows centralized dictionary for contacts, Voice-Video conferencing, Call Parking, Voice Mail, Web based access to voice mail and many other facilities. Asterisk’s flexible dial plan allows seamless integration of IVR and PBX functionality. Asterisks Features are easily implemented using nothing more than extension logic and dial plans. Asterisk supports a wide range of protocols for handling and transmitting voice over traditional telephony interfaces.

Asterisk PBX’s and Interactive Voice Response (IVR) applications are rapidly created and deployed. The powerful command line interface and feature rich text configuration files support rapid configuration and real-time diagnostics of the whole system. Web servers provide easy deployment of dynamic content, for example movie listings or weather reports. Asterisk can deploy dynamic (Mahler, 2004) content over the telephone, with the same ease.

Inter-Asterisk Exchange (IAX) is a Voice over IP protocol specific for Asterisk to exchange the information between two Asterisk Servers and allow to communicate easily. IAX allows Asterisk to merge voice and data traffic seamlessly across disparate networks. When using Packet Voice, data like URL information and images can be sent in-line with voice traffic. This supports advanced integration of voice and data that is not available in legacy systems. Asterisk provides a central switching core, with four APIs for modular loading of
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