Review on : Voice over IP and SIP based Asterisk PBX

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Introduction

Asterisk is a software implementation of a telephone private branch exchange (PBX) it allows attached telephones to make calls to one another, and to connect to other telephone services, such as the public switched telephone network (PSTN) and Voice over Internet Protocol (VoIP) services. Its name comes from the asterisk symbol, U+002A * ASTERISK.

Asterisk is released with a dual license model, using the GNU General Public License (GPL) as a free software license and a proprietary software license to permit licensees to distribute proprietary unpublished system components.

Asterisk was created in 1999 by Mark Spencer of Digium . Originally designed for Linux Asterisk runs on a variety of operating systems including NetBSD ,OpenBSD , FreeBSD , macOS and Solaris and can be installed in embedded systems based on OpenWrt and on flash Asterisk

Asterisk is a Public Branch Exchange (PBX) software which consists of a PBX, a SIP Proxy, a built in Interactive Voice Response (IVR) menu and a server that supports the four major VoIP protocols that are currently in use around the world namely SIP, MGCP, H.323 and IAX. For large VoIP networks, Asterisk runs with Sip Express Router (SER) which is a SIP proxy that can support large numbers of clients on behalf of Asterisk. Asterisk is a complete PBX in software which is both standards based and Open Source. J.Hitchcock [2006] also mentions the ILanga PBX Proxy VoIP service which integrates different technologies seamlessly and was developed at Rhodes University to be a complete cost effective computer based PBX [J.Hitchcock 2006].

PBX or Private Branch exchange is a telephone exchangethat serves a particular business or office, as opposed to one that a common carrieror telephone company operates for many businesses or for the general public [Wikipedia 2007].

J.Hitchcock [2006] writes that Asterisk allows for easy service creation in VoIP systems because it allows various telephony protocols to communicate in a transparent manner and also Asterisk acts as a middleware between services and telephony technologies which makes Asterisk the ideal platform for developing services for a converged VoIP network. Asterisk at the moment has the potential to be expanded to do much more than the current subset of telephony applications that is being used for. Asterisk currently shows the same potential that the PC showed in the early eighties and went on to dominate the computer market[M .Spencer , 2004] so Asterisk has the potential to be used in a wide variety of services. From this paper and other literature surrounding Asterisk cited in this review, Asterisk seems like it is a very good candidate to be the service provider for this project and all the services that will need to be developed for this system. Successful development of all the edutainment services that were planned from this project has proved the suitability of the use of Asterisk as the PBX for the development of edutainment services for a VoIP network. All the required services were developed through the Asterisk Dial plan.

VoIP

M.Chetty, E.Blakeand E.McPhie [2006] define VoIP as referring "to a range of protocols designed to send voice over packet switched networks, traditionally the domain of internet traffic.**" In VoIP voice is sampled at a certain frequency which can be set to any desired value on the devices in use. The sampled voice is then digitised and then finally packaged into packets before being sent over the IP network. VoIP uses different protocols for the call setup and the actual conversation between two communicating telephones. Signalling protocols like SIP, H.323 and IAX are used for the call setup and then the Real Time Transmission Protocol (RTP) is used to carry the voice between the telephones. RTP has been specifically optimised to for the transmission of real time data. About the idea of creating additional services in VoIP systems which this project tested, J.Hitchcock [2006] in**

his paper entitled *DecoratingAsterisk: Experiments in Service Creation* *for a Multi-Protocol Telephony Environment Using Open Source Tools***wrote that "Voice over IP (VoIP) is no longer a single service application, but an array of marketable services** of increasing depth, which are moving into the non-desktop market."

M.Chetty et al [2006] also offer a couple of advantages that VoIP offers over the Traditional Public Switched Telephone Network (PSTN). These advantages include the fact that VoIP makes more efficient use of bandwidth by only transmitting when something useful is being sent. The other advantage of VoIP that M.Chetty et al [2006] also mention is that VoIP obviates the need to separate data and voice streams as will be necessary in traditional telephony since both types of packets can be carried on the same line at the same time. One of the major disadvantages of VoIP however is the difficulty to guarantee a certain level of quality of service. M.Chetty et al [2006] mention that this is because voice quality in VoIP networks varies with the audio or video codec that is being used and also such factors such as latency, packet loss and jitter may also degrade the overall voice or video quality of a conversation. To try and increase the voice quality in VoIP networks M.Chetty et al mention that in networks where the same network is also used for data transport it is necessary to give the VoIP packets a higher priority over the data packets in order to combat the deterioration in quality of service as the network gets congested. This issue on sound quality relates strongly to this project since the main function of the system, the Book Readout Function, is sound dependant and will be affected strongly with the sound quality that the VoIP network in use can deliver.

W.Yu, S.Chellappan and D.Xuan [2005] wrote a paper on VoIP entitled P2P/Grid-based overlay architecture to support *VoIP* services in large-scale IP networks which suggests using a P2P grid for a VoIP network instead of the traditional client/server model that is currently widely used. W.Yu et al [2005] give a detailed suggestion of how to implement a P2P network layout for a VoIP network but however do not give any comparisons between the P2P network layout and the client/server layout. W.Yu et al [2005] do however mention that experimental results demonstrate that the P2P + Hierarchy model for conferencing applications can achieve better performance than all other VoIP models in terms of minimizing the network bandwidth overhead. This certainly suggests an angle to the deployment of the system being developed in this project should the deployment become large scale in the future.

In *A Generic API for Interoperation between Heterogeneous Overlays for Peer to Peer SIP* by M.Tsietsi, G.Wells and A.Terzoli [2007], the authors describe an approach for inter-operability between heterogeneous peer to peer overlays for serverless SIP. This paper also mentions how SIP has risen in popularity in recent years and how peer to peer networks are essential for supporting distributed services. SIP or "Session Initiation Protocol is an application-layer control (signalling) protocol for creating, modifying, and terminating sessions with one or more participants. It can be used to create two-party, multiparty, or multicast sessions that include Internet telephone calls, multimedia distribution, and multimedia conferences. (cit. RFC 3261). SIP is designed to be independent of the underlying transport layer; it can run on TCP, UDP, or SCTP. VoIP- Voice over Internet Protocol also can be described as telephony over a computer network", [Wikipedia 2007]. The VoIP network that was developed in this project also uses SIP as its signaling protocol.

This project is about adding creating and analysing the VOIP based SIP protocol for day to day use in everyday life as an implementation on an asterisk based network,in asterisk based network of PBX can be deployed in school, colleges and Commercial Institutions the PBX is supported with various other Vendors for Interoperability features.

Asterisk is a core component in many commercial products and open-source projects. Some of the commercial products are hardware and software bundles, for which the manufacturer supports and releases the software with an open-source distribution model.

- AskoziaPBX a fork of the m0n0wall project, uses Asterisk PBX software to realize all telephony functions.
- Elastix uses Asterisk, HylaFAXOpenfire and Postfix to offer PBX, fax, instant messaging and email functions, respectively.
- FreePBX, an open-source graphical user interface, bundles Asterisk as the core of its FreePBX Distro
- LinuxMCE bundles Asterisk to provide telephony; there is also an embedded version of Asterisk for OpenWrt routers.

- PBX in a Flash/Incredible PBX and trixbox are software PBXes based on Asterisk.

There are also various add-on products, often commercial, which extend Asterisk in some manner.

As one example, the standard voice prompts included with the system are free. A business can purchase matching voice announcements of its company name, IVR menu options and employee or department names (as a library of live recordings of common names or a set of fully customised prompts recorded by the same professional voice talent) at additional cost for seamless integration into the system.

SIP (Session Initiation Protocol)

The **Session Initiation Protocol** (**SIP**) is a communications **protocol** for signalling and controlling multimedia communication **sessions** in applications of Internet telephony for voice and video calls, in private IP telephone systems, as well as in instant messaging over Internet **Protocol** (IP) networks.

SIP Channel Module

The SIP Channel Module enables Asterisk to communicate via VOIPwith SIPtelephones and exchanges. Asterisk is able to act as

- A SIP **client**: This means that Asterisk registers as a client to another SIP server and receives and places calls to this server. Incoming calls are routed to an Asterisk extension.
- A SIP **server**: Asterisk can be configured so that SIP clients (phones, software clients) register to the Asterisk server and set up SIP sessions with the server, i.e. calls and answers incoming calls. This said, Asterisk is not a full-feature SIP server like SIP express routeror OpenSER If you are going to have thousands of SIP phones, you should use SER or OpenSER and forward calls to Asterisk for voicemail or PSTN access.
- a SIP **gateway**: Asterisk acts as a Media gateway between SIP, IAX, MGCP, H.323 and PSTN connections. As an example, an Asterisk server can be connected to ISDN to give your SIP clients connectivity to the switched telephone network.

Telephones and Edutainment

Although at the moment there is a lack of literature on the use of desktop telephone handsets for edutainment, an article that appeared in an online magazine called eContent Magazine in May 2005 written by P.A Salz[2005] entitled "*Random House Gets VOICEL: Invests in Edutainment*", mentions that although most traditional publishing houses have been out of the mobile service industry, this is about to change because of the improvement in mobile telephone technology and also because of the large profits made from mobile content delivery by other industries such as the music and news industries.

Conclusion

VoIP has become common, and telephone handset prices have fallen rapidly, and we see rapid addition of new services and applications to its capabilities. Asterisk has established a niche for itself in the VoIP industry and has shown incredible flexibility in how it can be used as the base application in the creation of new services to VoIP networks. Because of this, the VoIP network is a good option to provide edutainment functions to the target end users of this system as content can be simplified enough for young children. Also the literature on edutainment in telephones is mostly available for edutainment over mobile telephones but because of the related cost, offering this edutainment functionality on a telephone network in Africa becomes practical if land line telephones are used in a closed classroom or school situation, since they are cheaper, comparatively easier to physically secure than mobile telephones. The services that have been indicated can also run satisfactorily on desktop IP telephones so this shows the practicability of the idea of SIP based asterisk PBX service for communication purposes.

Voice Over IP Primer. â€“ Signaling Protocols â€“ Codecs. Voice Over IP and Asterisk. â€“ H323 (chan_h323) â€“ SIP (chan_sip) â€“ IAX (chan_iax2). The NAT Traversal problem The Asterisk Application Gateway Interface (AGI). â€“ AGI Programming Basics â€“ AGI Simple Examples â€“ AGI Execution Flow and the DeadAGI concept. Questions and Answers. Voice Over IP â€“ VoIP for short. A Primer to Voice Over IP. Voice Over IP (VoIP) can be described as the ability to sample voice transmissions, packet them into well known chunks, and transmitting them over an IP network. VoIP traffic is split into 2 network transmission