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Voice Over IIT Developments And Integration With SIP.edu

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VoIP@IIT → Voice Over IIT → VoIIT

- VoIP is one of the numerous technologies that allows you to make phone calls over the Internet instead of only over the public switched telephone network. As an Information Technology and Management student expertise in the field of communication I felt that IIT must also be communicating using the latest technologies i.e. over the internet which is the VoIP communication. Interning and working in the VoIP lab under the guidance of Carol Davids helped me increase my technological and development skills. Under her guidance I was requested to build the a VoIP communication system at IIT for all the CPD staff and students. This way the Voice over IIT was born.
- The reduced costs, good network connection, and the various features which we could get were the reasons to shift to VoIP. The best choice for this is using open source software's such as Asterisk. Currently we have around 75 users registered to the IIT Rice campus Asterisk Server. It currently has various features and registered extensions in the format 1XXX@sip.iit.edu. These extensions can be used to dial any sip user agent or anyone can call this user agent.
- We are currently working on configuring our setup in accordance with the SIP.edu working group and follow the guidelines provided in their cookbook.

SIP.edu

The SIP.edu Working Group seeks to encourage experimentation with new campus real time communications services that use the Session Initiation Protocol (SIP) and its related protocols. The goals of the working group are to:

- Build a large base of SIP-reachable Internet2 users by making existing campus PBX, Centrex, and VoIP systems reachable via SIP*
- Facilitate the convergence of communications identities by promoting the use of email addresses for voice and multimedia communications*

Lower the barriers and reduce timelines for campus SIP voice deployments by:

- Publishing and maintaining a "cookbook" for deploying SIP infrastructure*
- Working with Internet2 corporate members and other suppliers to obtain special pricing for SIP hardware, software, and services*
- Building a knowledge base within the Internet2 community to assist others in deploying SIP infrastructure*
- Promoting the use SIP for additional media types, including presence, video, and instant messaging*
- Fostering experimentation with new SIP-based applications and services*
- Exploring ways in which SIP services can be used by faculty, students, and staff to enhance university activities*
- Developing best practices for scaling and securing campus SIP deployments*
- Collaborating with vendors and standards groups to help evolve SIP products and standards*

Asterisk

Asterisk is a leading open source telephony engine and tool kit. Asterisk has various inbuilt aspects and capabilities:

- **Asterisk as a switch (PBX)**
- **Asterisk as a gateway**
- **Asterisk as a feature/media server**
- **Asterisk in the call center**
- **Asterisk in the network**

I have used Asterisk as my primary application and used the above capabilities.

VoIP Packages

- **asterisk 1.4.22 (Rice VoIP)**
- **zaptel 1.4.11:** Required if the user is using an analog or digital hardware
- **libpri 1.4.6:** Optional unless the user uses a ISDN PRI interfaces
- **asterisk-addons-1.4.7:** The asterisk-addons package contains code to allow the storage of Call Detail Records (CDRs) to a MySQL database.
- **Asterisk-gui 2.0:** The Asterisk GUI is the interface that comes with the AsteriskNOW distribution or can be added to an existing Asterisk installation.

sip.iit.edu DNS

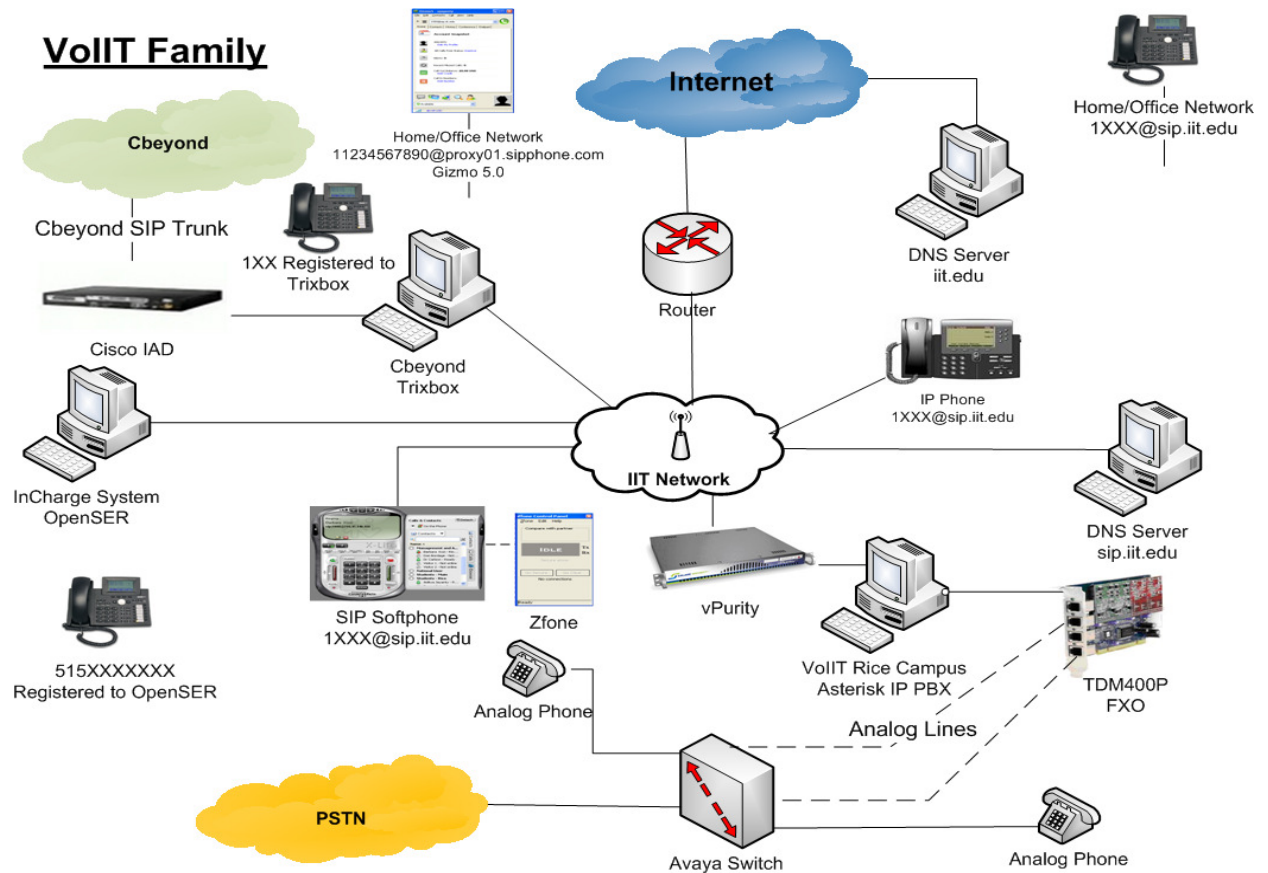
- DNS provides two record types relevant to SIP requests: SRV and NAPTR. Some implementations will use SRV records only. The DNS server with the respective SRV and NAPTR records for the sip.iit.edu is implemented.
- In the main iit.edu DNS the sip.iit.edu is specified as a Name Server.
- The main reason for this was we wanted to maintain our own DNS server and the IIT network were not ready to make any of these major changes in their primary DNS server, iit.edu.

```
$ttl 43200
@ IN SOA sip.iit.edu. root.sip.iit.edu. (
    2003032001
    10800
    3600
    604800
    86400 )
@      IN NS   ns.sip.iit.edu.
ns.sip.iit.edu.  IN A   1.1.1.1
sipserver.rice.iit.edu.  IN A   2.2.2.2
;("_Service._Proto.Name TTL Class SRV Priority Weight Port Target" )
_sip._udp.sip.iit.edu.  IN SRV 0 0 5060 sipserver.rice.iit.edu.
;("domain-name TTL Class NAPTR order preference flags service regexp target"
sip.iit.edu. IN NAPTR 0 0 "s" "SIP+D2U" "" _sip._udp.sip.iit.edu.
```

Slide 5

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VoIT Family



Slide 6

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Features of VoIT

- User Agents format sip:1XXX@sip.iit.edu
- 4 digit internal dialing 1xxx in the VoIT System
- Dialing Trixbox extensions 1XX provided by Cbeyond and OpenSER extensions 515XXXXXXX provided by Incharge Systems
- Receive and Make calls to and from any SIP User Agent of any network.
- User Agents Devices:
 - Xlite 3.0 Softphone with compatible USB hard phones, Nokia E-Series Phones, Nortel IP Phones, Cisco IP Phones
- Currently 75 configured VoIT User Agents
- Auto attendant to call the User Agents
- Audio and Video communication via SIP
- Inbound and Outbound calling via the PSTN
 - Outbound calling provided using the Cbeyond's Network
- Voicemail box and Voicemail box messages to email (attached as .wav file)
- Presence Feature for User Agent statuses
- Audio Conference (upto 15 participants)

VoIP and Digium - Zaptel PSTN Gateway Card Integration

- The Digium provides a broad selection of telephony interface devices. These devices help us to connect to the PSTN/ISDN PRI. Combined with Asterisk, the open source IP PBX, these cards provide an interface to traditional telephony equipments. Asterisk can extend the capabilities of its phone network, by acting as a bridge between traditional telephony systems and Voice over IP equipment. There are various cards which could be used but I used the TDM400P which is a *half-length PCI 2.2-compliant card that supports FXS and FXO station interfaces for connecting analog telephones and analog POTS lines through a desktop*. We use the two FXO modules in our setup that are required to connect to the telephone system to the public telephone system (PSTN).
- I submitted a detailed document that describes one possible way to configure a gateway using the Digium TDM400P and Asterisk IP PBX.



VoIT and ICS Products Integration

- Support the Incharge Systems team housed in the VoIP Lab to integrate the VoIT setup with the OpenSER SIP Proxy being used.

VoIT and Trixbox integration

- Support the Trixbox team to integrate the VoIT setup with the current Trixbox provided by Cbeyond. The primary purpose of this is to use the Cbeyond SIP trunk for outbound calling.

VoIT and Nokia E61i

- Integrate the Nokia E61i and E71 provided by Nokia and test its capabilities with the VoIT setup. Test the capabilities to use both the GSM and Internet calling from the cell phone.

VoIT and vPurity – Salare Product Integration:

- Support the Salare Team housed in the VoIP Lab to integrate the vPurity device which is used between the backbone and the VoIT server. This device is used for traffic analysis. It helps to monitor the connected server and provide detailed traffic related data.

VoIP and Zfone

- Setup and Integrate the Zfone (version 0.92 build 217) with the current VoIP setup. The mode selected is ZRTP PBX MiTM mode. *In this case the enrolled user makes calls to different outside users. Two separate ZRTP channels are established: between Alice and the PBX and between the PBX and Bob. This results in two separate media streams encrypted by different stream keys and with two different Short Authentication Strings. The PBX should transfer its SAS to the enrolled user for verification. If both users are enrolled, the PBX will choose one according to the ZRTP protocol.*
- The MiTM mode is configured in our current setup as we would like our users to trust our system and enroll there VoIP User Agents.



Future of VoIT

- Make use of a pure SIP Proxy like OpenSER for User Agent registrations and other features
- Integrate and test various security aspects into the VoIT setup
- Find methodologies to integrate whiteboard and video conferencing
- Integrate chat applications to make a UC setup
- Make the VoIT setup more stable, secure and create a full failover system.
- Monitor the traffic patterns and bandwidth utilization using various methodologies

References

- Asterisk: The Future of Telephony, published by O'Reilly Media, Inc. and written by Jim van Meggelen, Jared Smith, and Leif Madsen is released under a Creative Commons license (click the CC graphic for more information).
- Internet Communications Using SIP - Delivering VoIP and Multimedia Services with Session Initiation Protocol Second Edition, Henry Sinnreich and Alan B. Johnston
- <http://mit.edu/sip/sip.edu/>
- <http://voip.itm.iit.edu/VoIPT.php>

? Questions ?