

Voice over Internet Protocol (VoIP)

What is VoIP?

- Sending/receiving voice (like a telephone call) over an IP network (like the Internet or LAN)
- VOIP is a network service like HTTP, but sends voice data instead of web page data



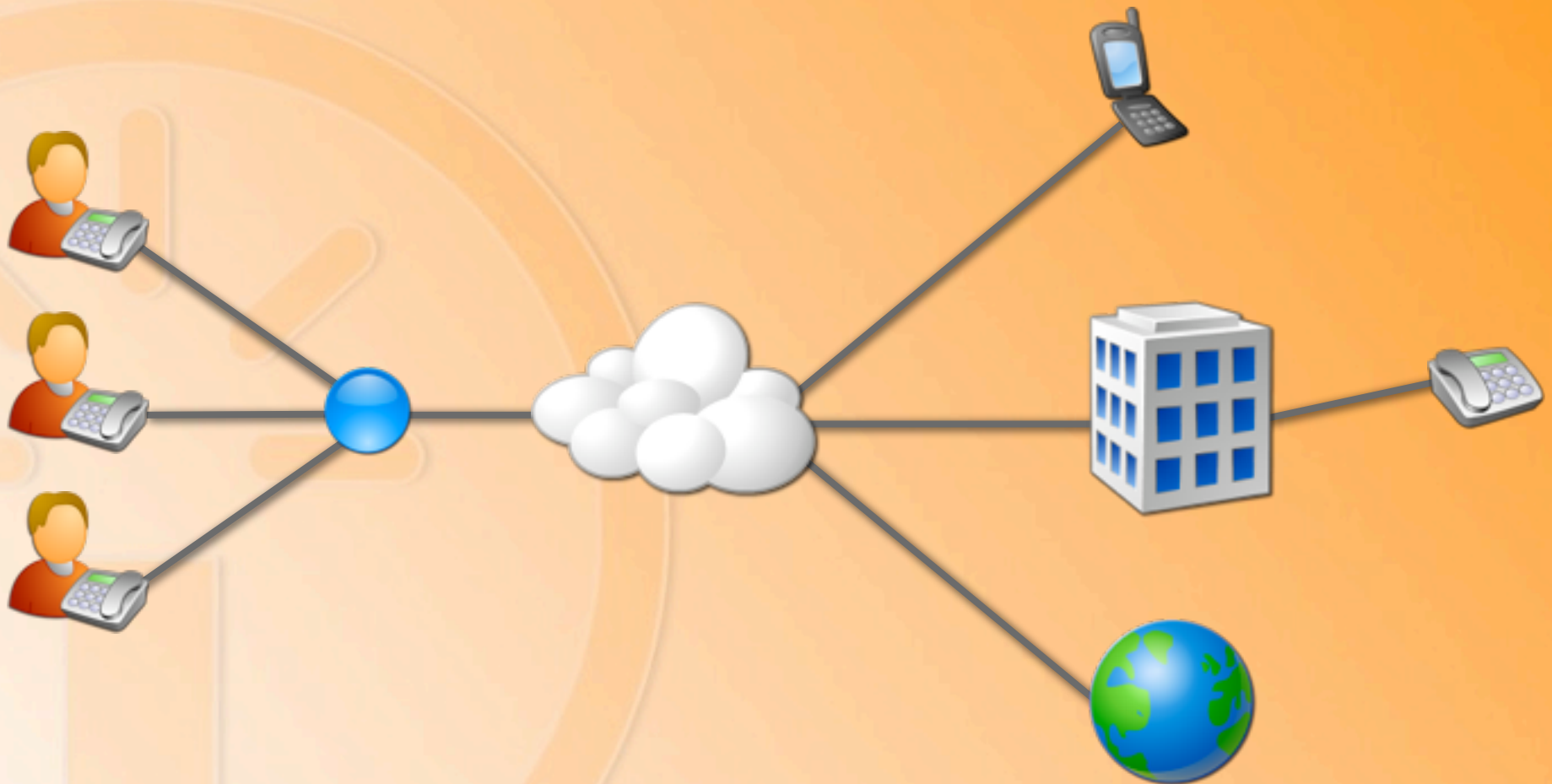
Why use VoIP?

- Save money
 - Value added: If you own the network, internal calls are free!
 - Cheap or free international calls via the Internet
- Provide professional telephone services
 - Voicemail, Voice menus, etc. at low cost

VoIP Choices

IP Only or **Internlinked**

- Do users need to connect to public telephone system (PSTN), cell network (GSM), the rest of the world?



VoIP Choices:

Peer-to-peer or Server Managed?

- Peer-to-peer: Point to point with 'no middle'
 - Skype, Google Talk, Yahoo Voice Chat
 - Internet required
 - PSTN interlink provided
- Server Managed: Central server negotiates all connections
 - Asterisk, other software options
 - Internet not necessary
 - Call routing
 - Voicemail, voice menus, etc.
 - PSTN interlink possible, but takes work

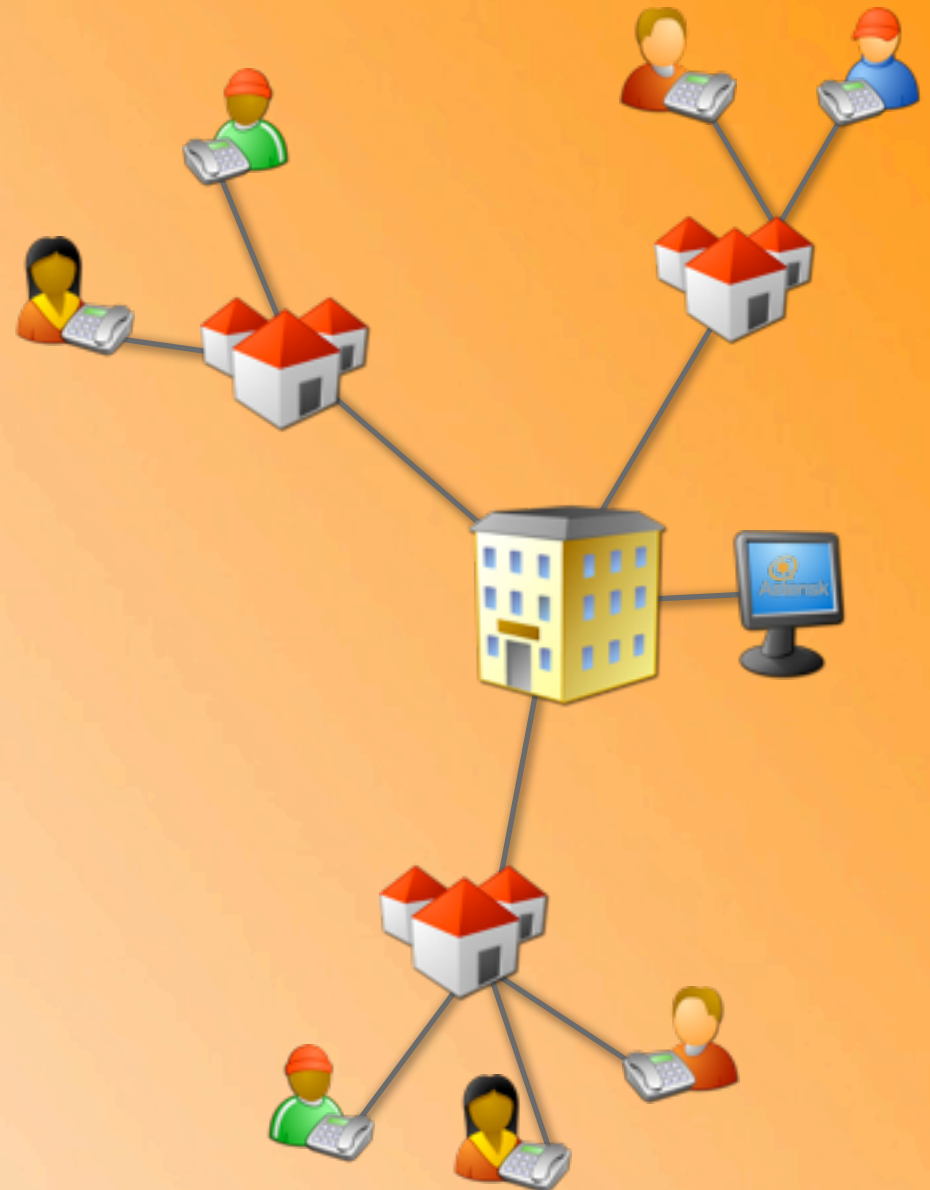


VoIP Choices

- How many users?
 - 10? 100? 1000?
 - Shared vs one user per phone
- Incoming calls necessary?
- 24/7 Operation?

VoIP Scenarios

- Telephones for villages
 - Calling other villages
 - Existing telephone service?
 - Calling family/friends working abroad



VoIP Scenarios

- Telephones for villages
 - Calling other villages
 - Existing telephone service?
 - Calling family/friends working abroad
- Telephones for health service
 - Connecting rural health offices with central hospitals



Peer-to-peer: Components

- Clients
 - Download client
 - Create account
 - Connect headset
 - Make calls
- Service provider takes care of Server and PSTN interlink



Server Managed: Components

- Server (Private Branch eXchange - PBX)
 - Asterisk or similar
- Clients
 - Analog telephone adapter (ATA)



Server Managed: Components

- Server (Private Branch eXchange - PBX)
 - Asterisk or similar
- Clients
 - Analog telephone adapter (ATA)
 - Hard phone



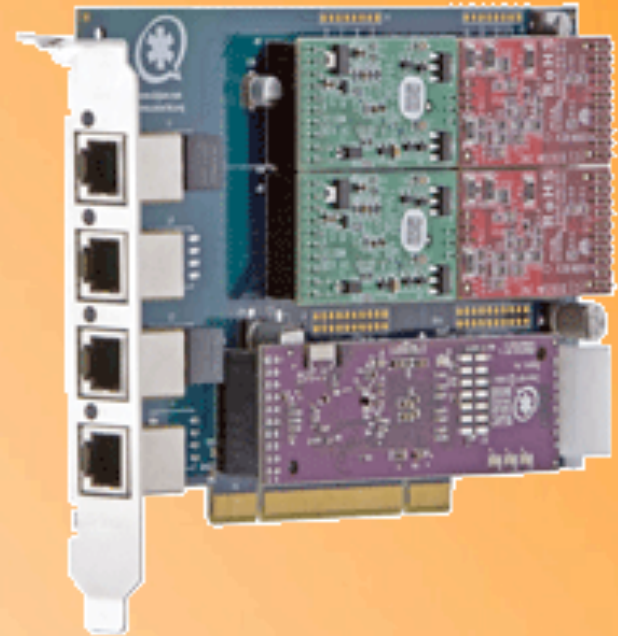
Server Managed: Components

- Server (Private Branch eXchange - PBX)
 - Asterisk or similar
- Clients
 - Analog telephone adapter (ATA)
 - Hard phone
 - Soft phone



Server Managed: Components

- Server (Private Branch eXchange - PBX)
 - Asterisk or similar
- Clients
 - Analog telephone adapter (ATA)
 - Hard phone
 - Soft phone
- Interlink
 - Digium Card
 - CDMA/GSM bridge
 - VoIP Service Provider (VSP)



Server Software

- Asterisk
 - Free and open source
 - Supported by commercial company: Digium
 - Also makes compatible hardware (PSTN cards)
 - Linux, Windows, Solaris (*best* on Linux)
- Commercial Software
 - Nortel, Alcatel, Lucent, Cisco
 - Very expensive: USD \$2000 - \$50000





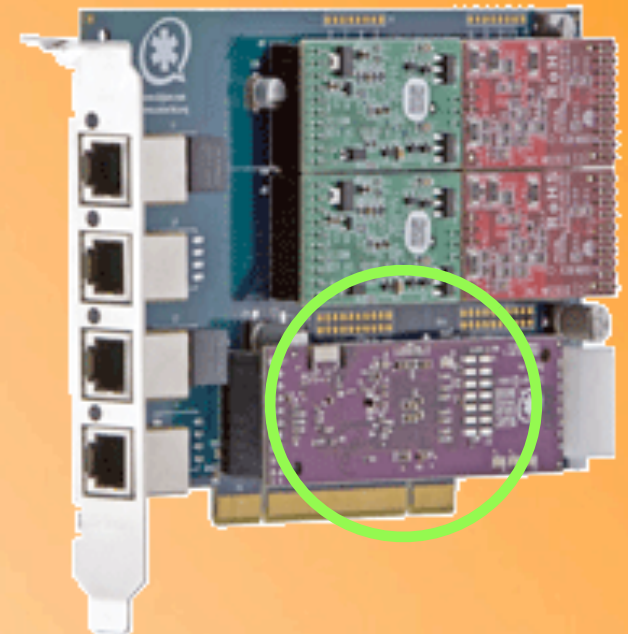
Managing Asterisk

- Asterisk GUI
- Built into IHL

The screenshot displays the Asterisk GUI interface. On the left is a vertical navigation menu with items: Home, Users, Conferencing, Voicemail, Call Queues, Service Providers, Setup Hardware, mISDN Config, Calling Rules, Incoming Calls, Voice Menus, Call Parking, Ring Groups, Record a Menu, Active Channels, System Info, Asterisk Logs, Global SIP options, Global IAX options, File Editor, Asterisk CLI, Backup, and Options. The 'System Info' item is selected. The main content area is titled 'System Information' and contains tabs for 'General', 'Ifoconfig', and 'Resources'. The 'General' tab is active, showing the following details: OS Version: Linux 16252-37681 2.6.18-5-686 #1 SMP Mon Dec 24 16:41:07 UTC 2007 1686 GNU/Linux; Uptime: 12:07:02 up 472 days, 50 min, 0 users, Load Average: 0.00, 0.00, 0.00; Asterisk Build: Asterisk 1.4.20.1, Asterisk GUI-version Revision: 2124 \$; Server Date & TimeZone: Tue Apr 21 12:07:02 PDT 2009; Hostname: 16252-37681.my-domain.com. At the bottom of the page, a yellow banner contains the copyright notice: Copyright 2006-2008 Digium, Inc. Digium and Asterisk are registered trademarks of Digium, Inc. All Rights Reserved. Legal Information.

PSTN Interlink

- Terminology
 - PBX: Private Branch eXchange (your server)
 - FXO: Foreign eXchange Office (acts like a telephone)
 - FXS: Foreign eXchange Station (acts like the wall jack)
- Physical Connection
 - Digium Card (goes in your PBX server)
 - 1-4 FXS ports (connect phones)
 - 1-4 FXO ports (connect phone lines)
 - Echo Cancellation (you want this)
- Modular, easy to fit to your budget!

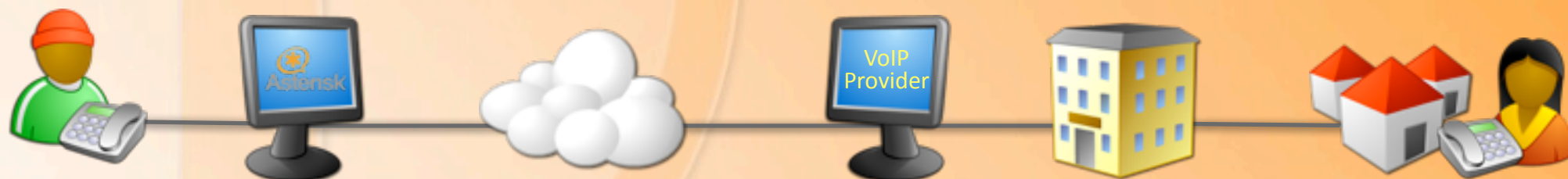


PSTN Interlink - Other Options

- Cellular (GSM/CDMA)
 - Bridge/modem



- VoIP Service Provider (VSP)
 - Configuration from VSP
 - username/password
 - server / SIP proxy
 - Calls go from your handsets, through your network, to your PBX, through the Internet, to your provider, out to the PSTN, to the called party



Technical Details

- Codecs - Converts sound into 11100101010101
 - Can be a very important decision, depends on your network
 - High quality, low compression
 - PCM/G.711
 - Medium quality, high compression
 - GSM
 - Speex
 - ILBC
 - SCCP/Skinny (Cisco)
 - G.729 (not-free)
- Protocols - Everything *except* the voice
 - Session-initiation-protocol/Real-time-protocol (SIP/RTP)
 - Inter Asterisk eXchange (IAX2)
 - Skype (closed)

References

- VoIP4D document
 - <http://www.it46.se/voip4d/voip4d.php>
- Asterisk Handbook
 - https://www.digium.com/elqNow/elqRedir.htm?ref=http://docs.digium.com/asterisk_handbook/handbook-draft.pdf
- voip-info.org active VoIP wiki with answers to everything!



Questions?

