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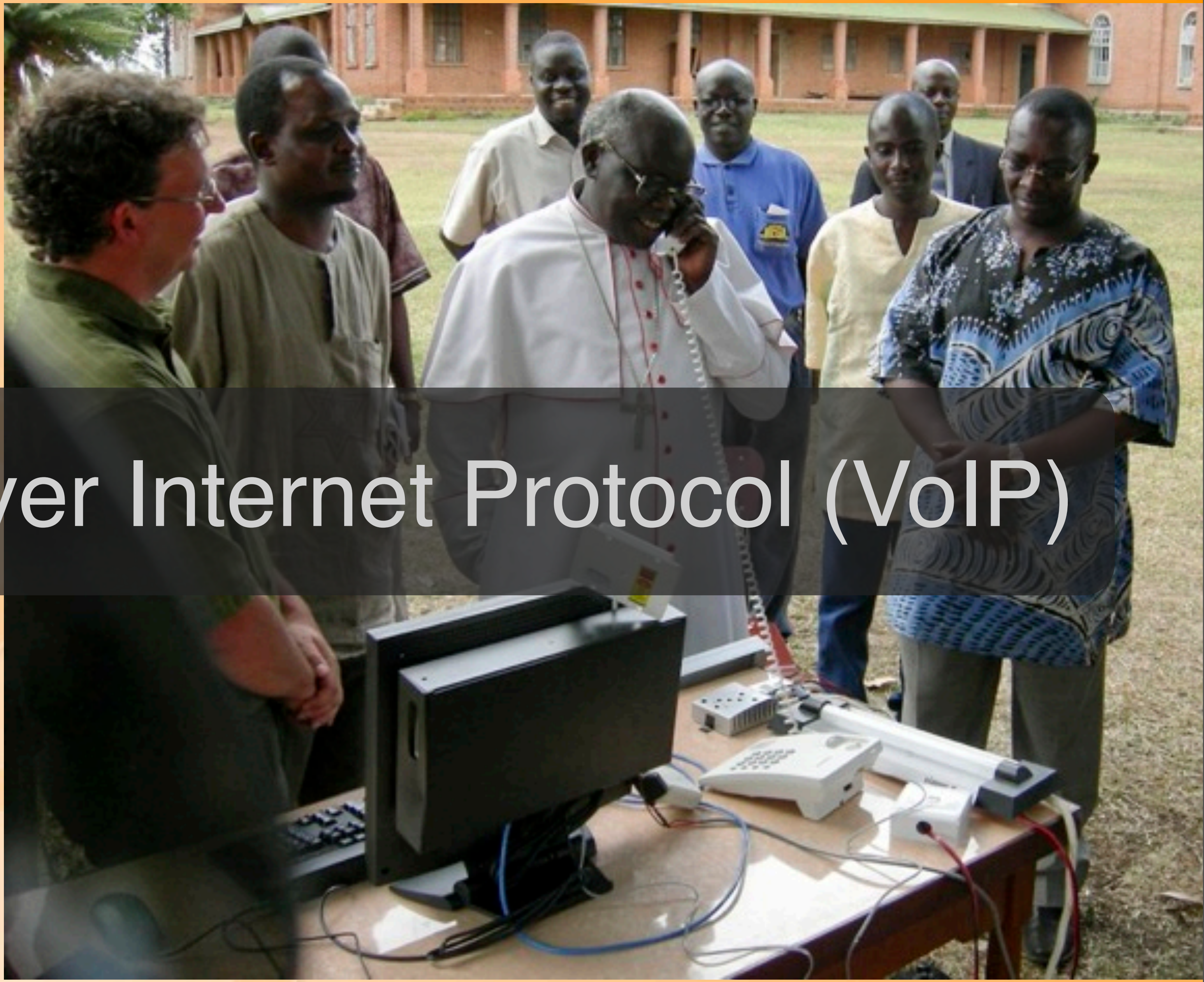
Voice over Internet Protocol (VoIP)



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

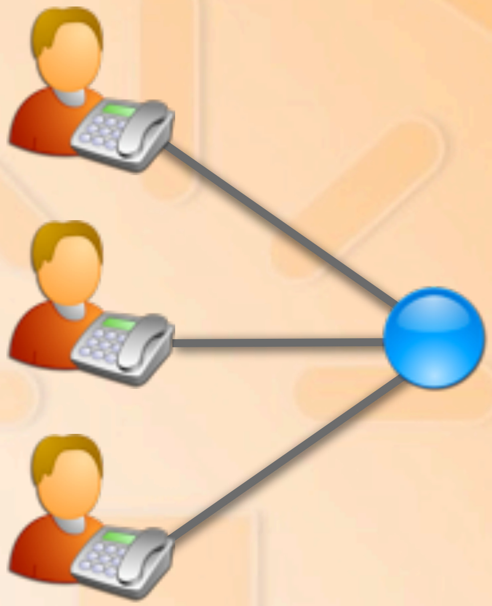
Why use VoIP?

- 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?



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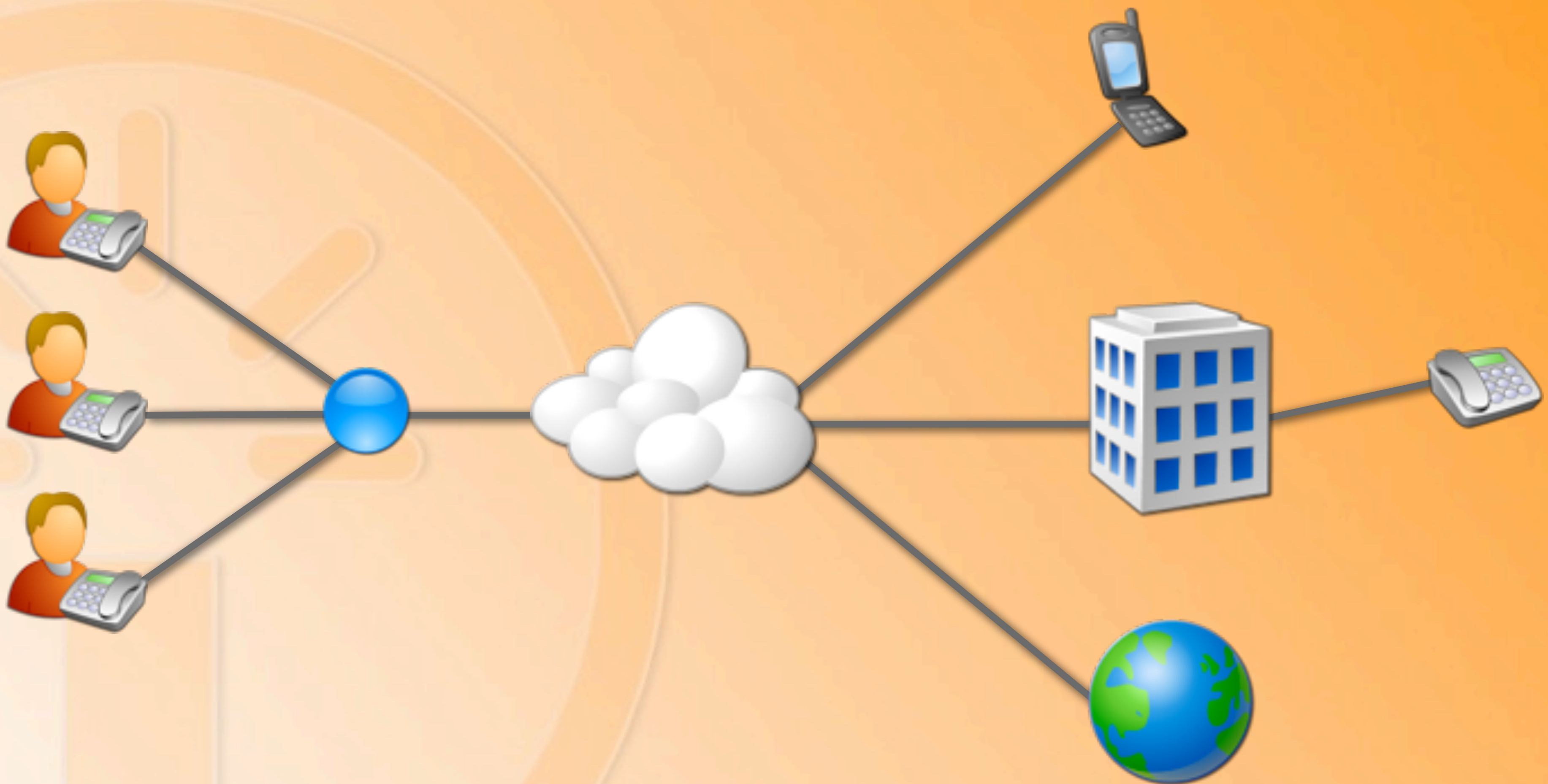


VoIP Choices:
Peer-to-peer or Server Managed?

- Peer-to-peer: Point to point with ‘no middle’
  - Skype, Google Talk, Yahoo Voice Chat
  - Internet <u>required</u>
  - PSTN interlink provided

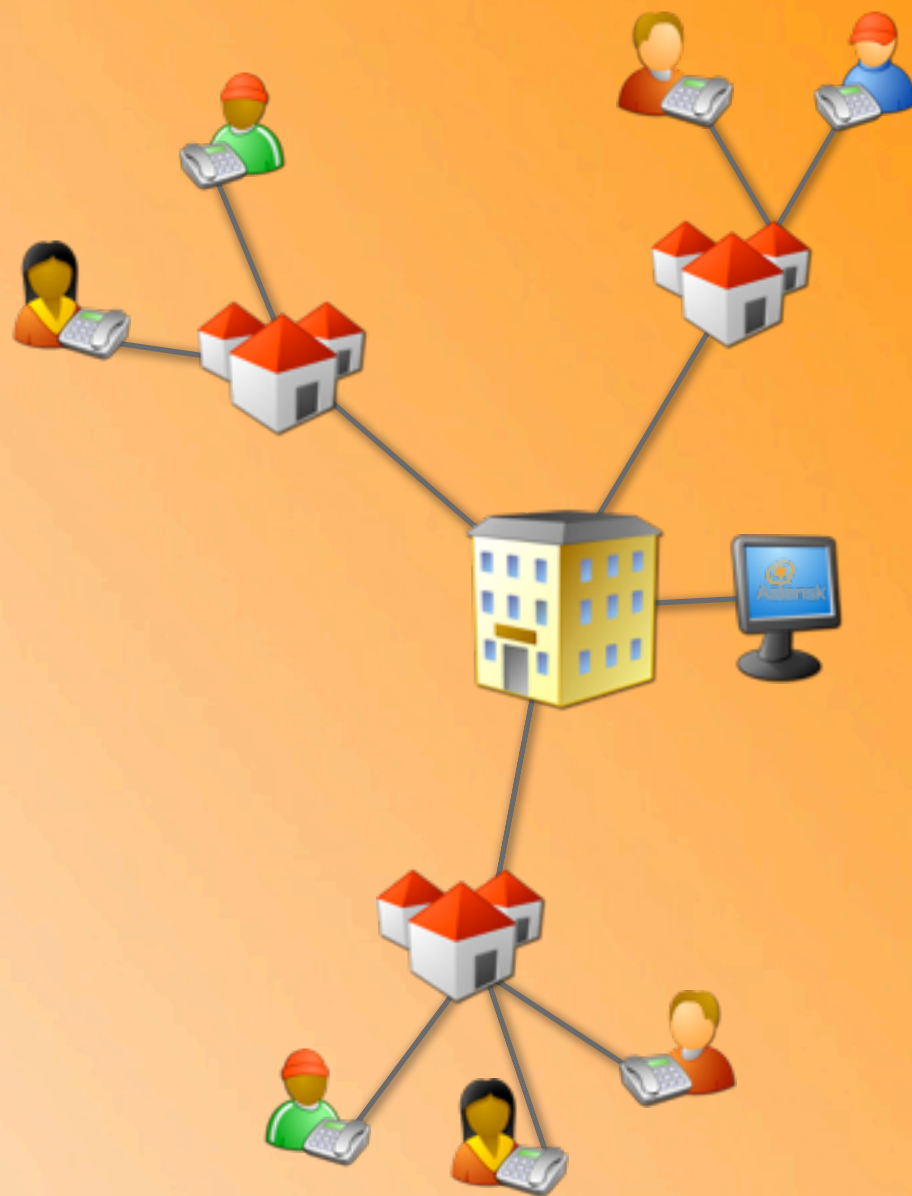
- Server Managed: Central server negotiates all connections
  - Asterisk, other software options
  - Internet not necessary
  - Call routing
  - Voicemail, voice menus, etc.
  - PSTN interlink possible, but takes work

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)

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  - Soft phone

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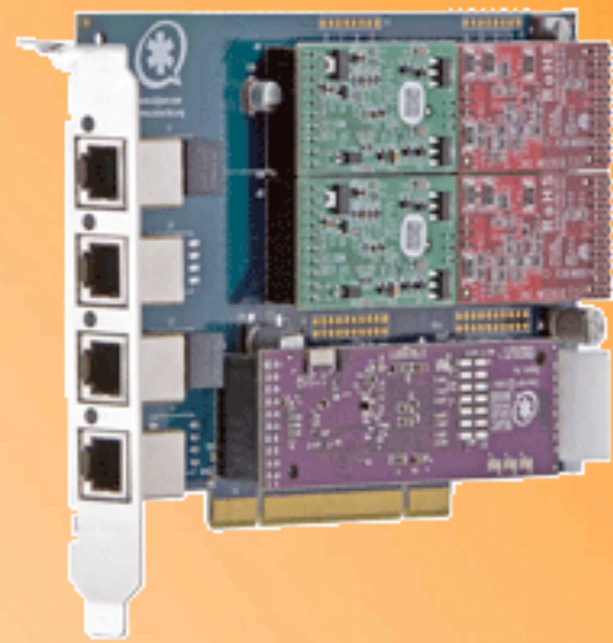
Server Managed: Components

- 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)

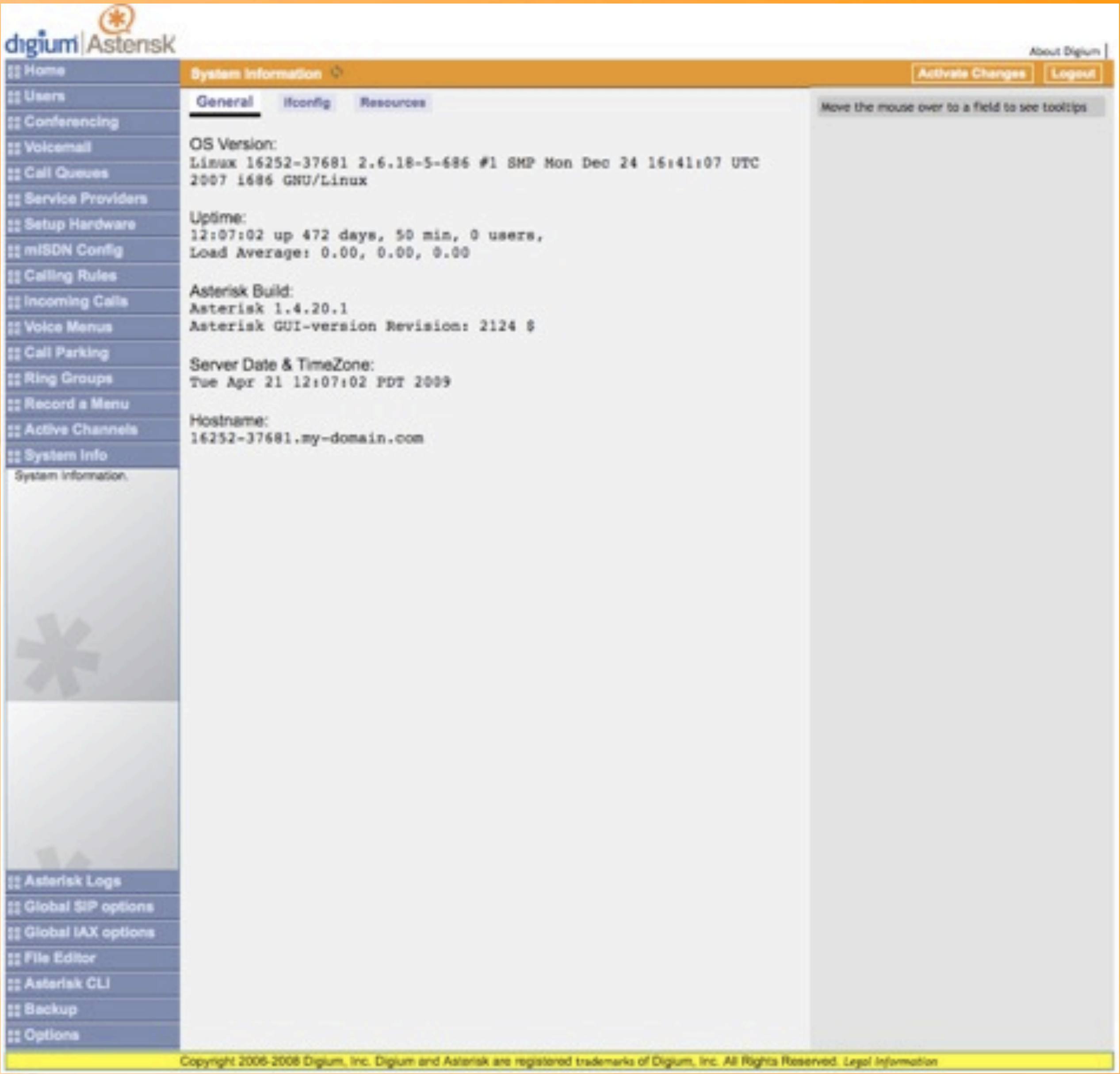


Server Software

- 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
  - <u>Very</u> expensive: USD $2000 - $50000

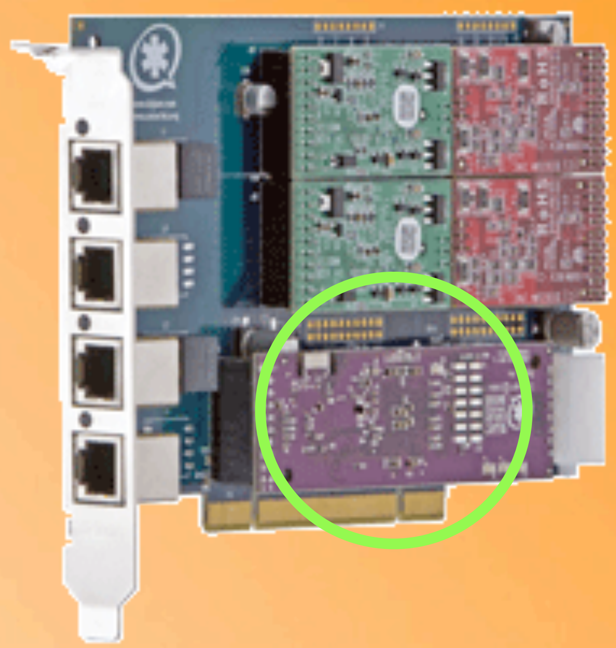
Managing Asterisk

- Asterisk GUI
- Built into IHL



PSTN Interlink

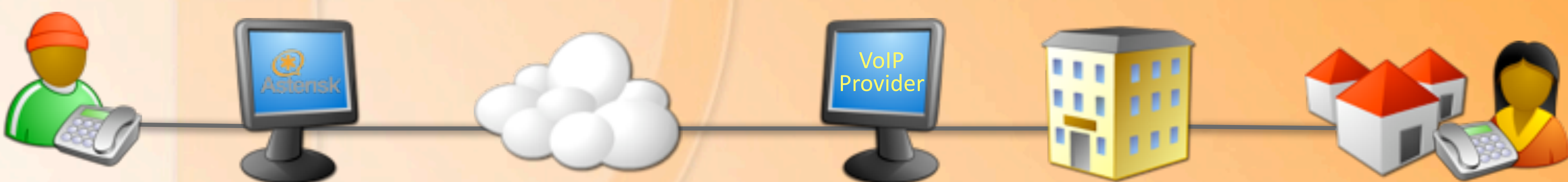
- Terminology
  - PBX: Private Branch eXchange (your server)
  - FXO: Foreign eXchange Office (acts like a telephone)
  - FXS: Foreign eXchange Station (acts like the wall jack)
- Physical Connection
  - Digium Card (goes in your PBX server)
  - 1-4 FXS ports (connect phones)
  - 1-4 FXO ports (connect 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?