Introduction to VOIP

Stephen Okay
Abdus Salam Int'l Center for Theoretical Physics
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Intro to VOIP

- Classic Telephony
- Data Networks(Review)
- VOIP
  - What it is
  - Protocols
  - Hardware
  - Software
  - Examples
  - Web Links

Classic Telephony in 1 slide

- Classic Telephony
  - Calls happen by electro-mechanical manipulation of voltage levels between the telco network and end-user phones.
  - voice “payload” and transport signals must be sent together over the same line/circuit.
  - “Modern” equipment is electronic, but must work w/ older equipment.
    - This makes upgrades and enhancements difficult and expensive.

VOIP in 1 slide

- Manipulate bits not volts
  - everything can now happen inside a PC
- Can be all-digital or a mix of digital and analog equipment
- Telephone Company/PTT not required
- Can use low-power, commodity hardware instead of expensive, dedicated gear

Intro to VOIP

- Review of TCP/IP
- VOIP protocols
  - H323
  - SIP
- VOIP hardware

TCP/IP Review

- 4-layer stack
- Packet-switched
- Error correcting
- Reliable delivery
- Designed in 1960s/1970s for use over slow, unreliable analog phone networks
- Application layer key to VOIP and other commonly used protocols

VOIP

- Uses formal protocol suites to provide:
  - Call routing, forwarding, voicemail, etc.
  - Compatibility w/ legacy PSTN/POTS systems
  - Quality of Service(QOS)
- Done mostly w/ software
- Can still do point-to-point calls

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Major VOIP benefits

- It’s all just bits.
  - VOIP is just another application running over a data network(usually TCP/IP)
  - Service expansion/integration often just a matter of writing the code, not tearing down a switching center.
- Per-call costs very low, often free
- Equipment is often commodity priced

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VOIP Concerns

- Regulatory issues
- Tax revenue issues
- Market exploitation and control
- Quality & Reliability of Infrastructure

VOIP Protocols

- H.323
- SIP
- MGCP
- IAX
- Others

H.323

- Specification defined by ITU
- Wide support among all telecom providers & manufacturers all over the world.
- Industry moving away from using it in new products.
- Meant to provide gateway for telephony devices into the PSTN

SIP

- “Session Initiation Protocol”
- Designed by Cisco Systems, Inc.
- Stand-alone computer-to-computer protocol
  - Does not presume a PSTN
- Calls routed/managed by a SIP Server
- No real “official” version, so there are lots of different implementations.
- Dominant open VOIP protocol

Protocol details

- Most VOIP protocols split signalling from voice data, unlike POTS
  - potential firewall issues
- QOS/capacity checked before calls are initiated

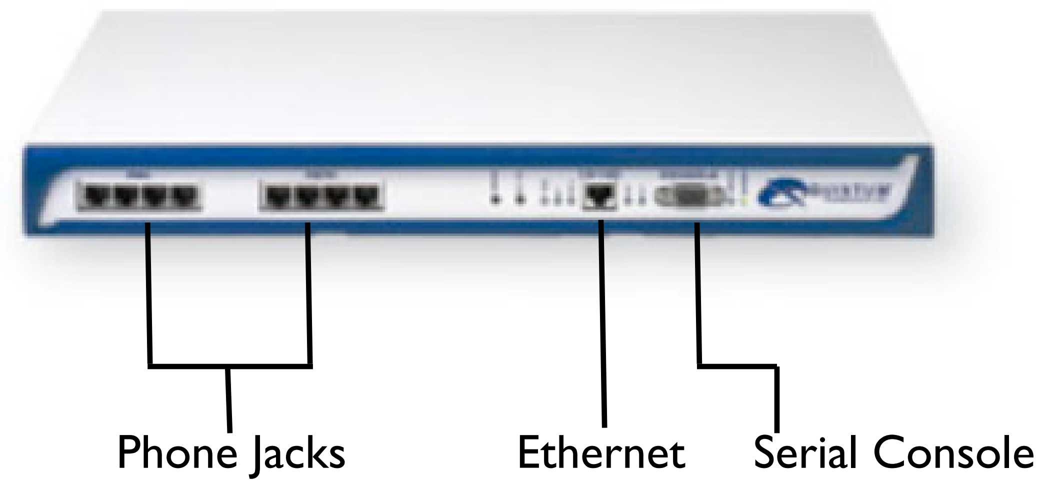
VOIP Hardware

- PSTN Gateways
- Telephone adapters
- Handsets
- Softswitch/PBX
- (Much of this can actually be done on a PC)

PSTN Gateways

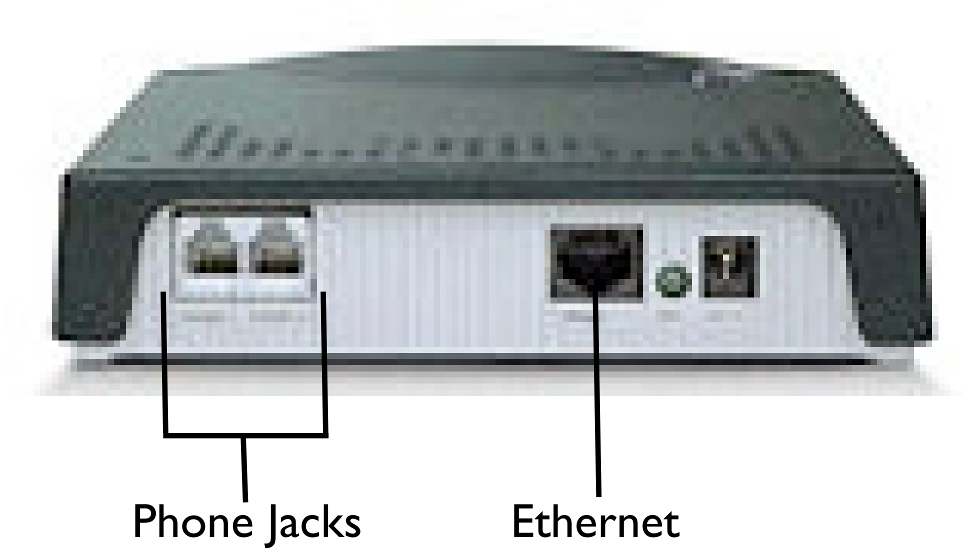
PCI Card

Office/Telecom System



- Provides a bridge between VOIP networks and the PSTN.

Analog Telephone Adapter



- Lets you use regular phone on VOIP network
- Avoid dedicated or “locked” adapters if possible
- $40-200 US

VOIP Phones

- (Really an embedded computer with VOIP client software)
- Ethernet or wireless (802.11b)
- $120-300

VOIP Phones

- USB Phones("Skype phones")
- Much simpler, much less expensive ($40-90 US)

Soft Phones

- Telephony functions completely in software
- Run on desktop, laptop and embedded systems.
- Some common softphone clients
  - SJPhone
  - XLite
  - KPhone
  - OhPhone

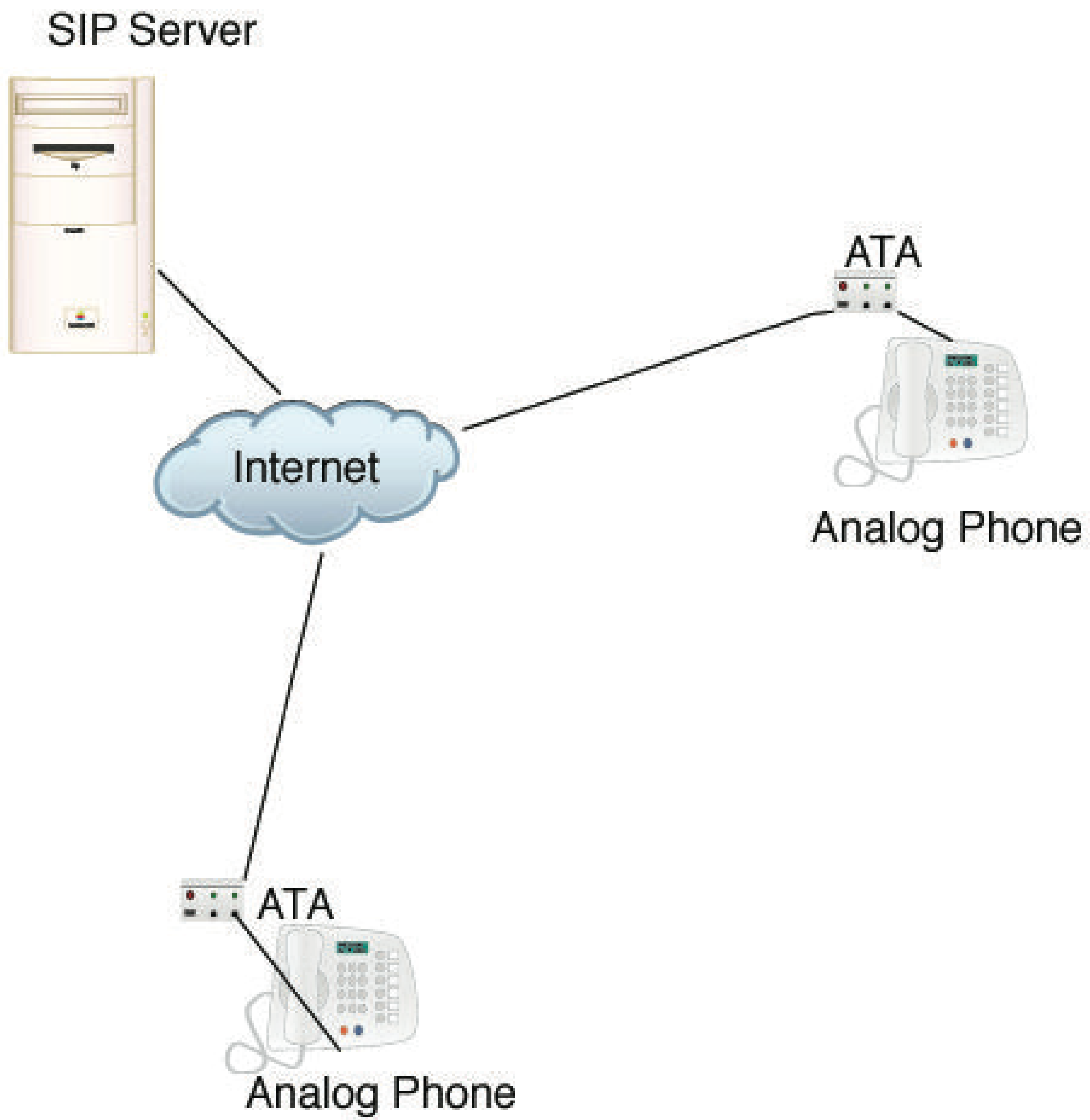
VOIP Networks

- Examples
  - PC-to-PC
  - PC-to-PSTN
  - VOIP-based POTS replacement
  - VOIP PBX using Asterisk

Asterisk

- Open Source GPL-Licensed PBX
- Runs under most popular versions of UNIX (Linux, BSD, OS X)
- Can replace a traditional office PBX
- Supports soft phones, SIP handsets, wireless phones, etc.

PC-PC VOIP Network with Analog phones



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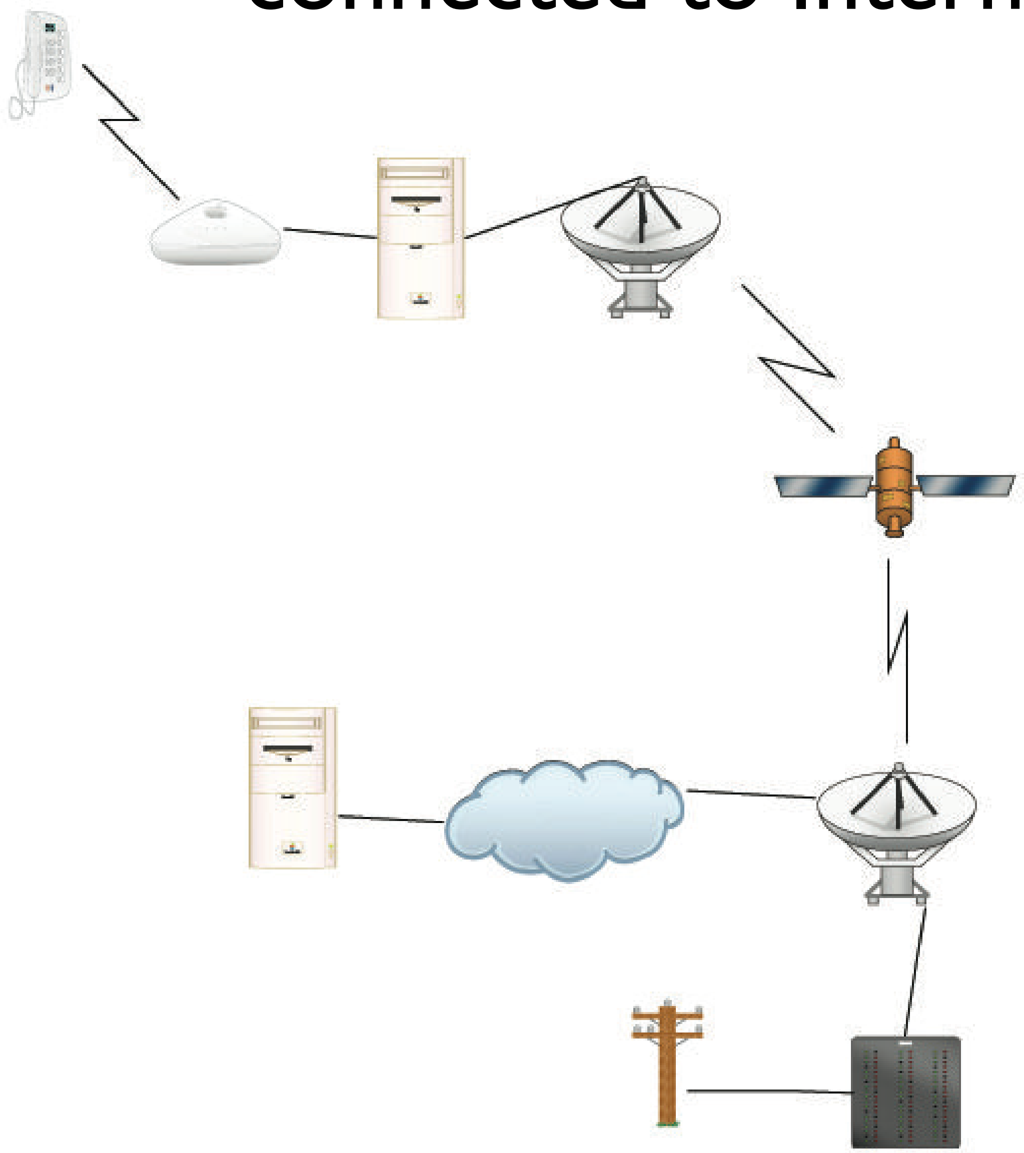
VOIP Network Using Satellite connected to Internet and PSTN



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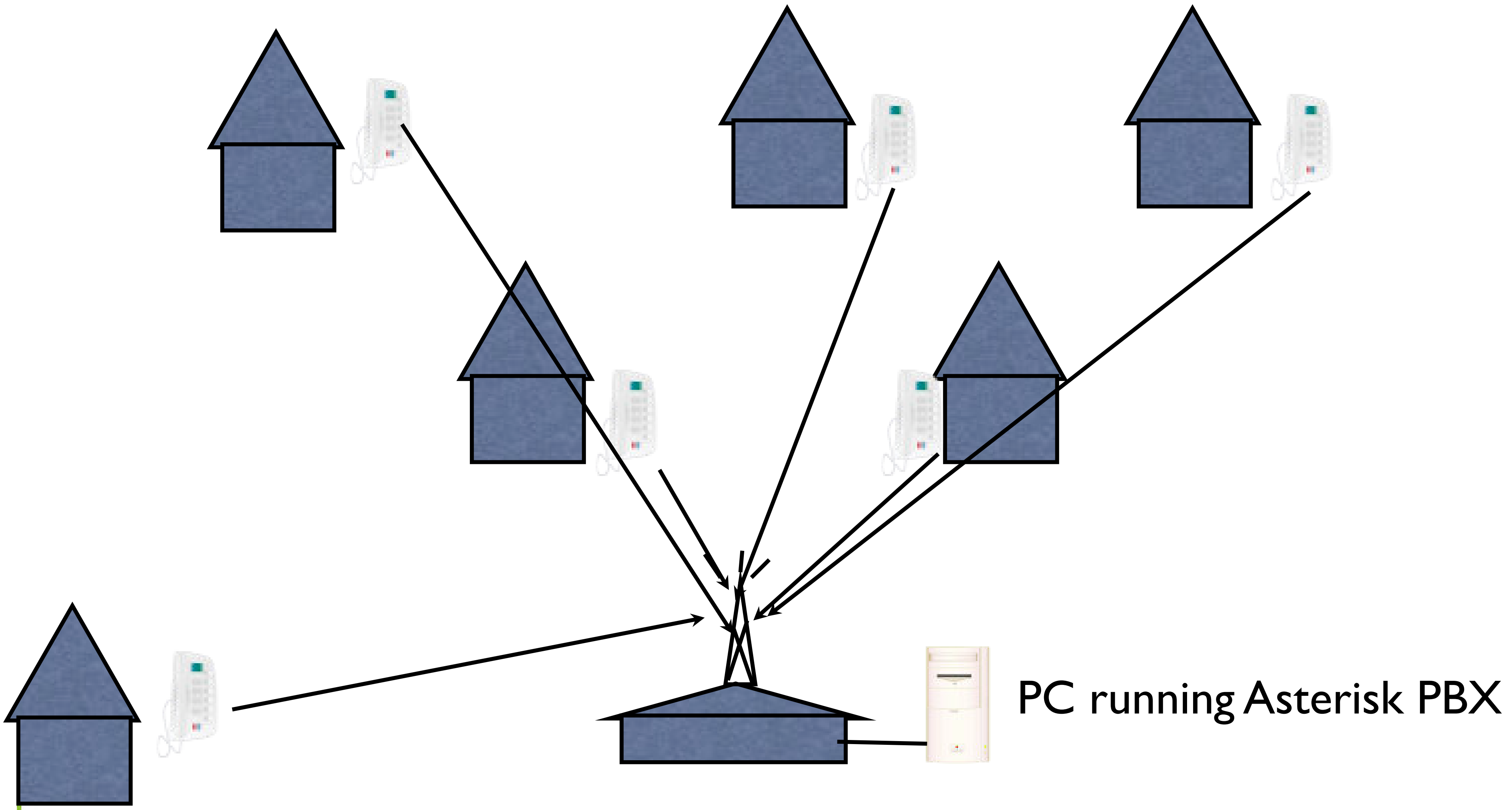
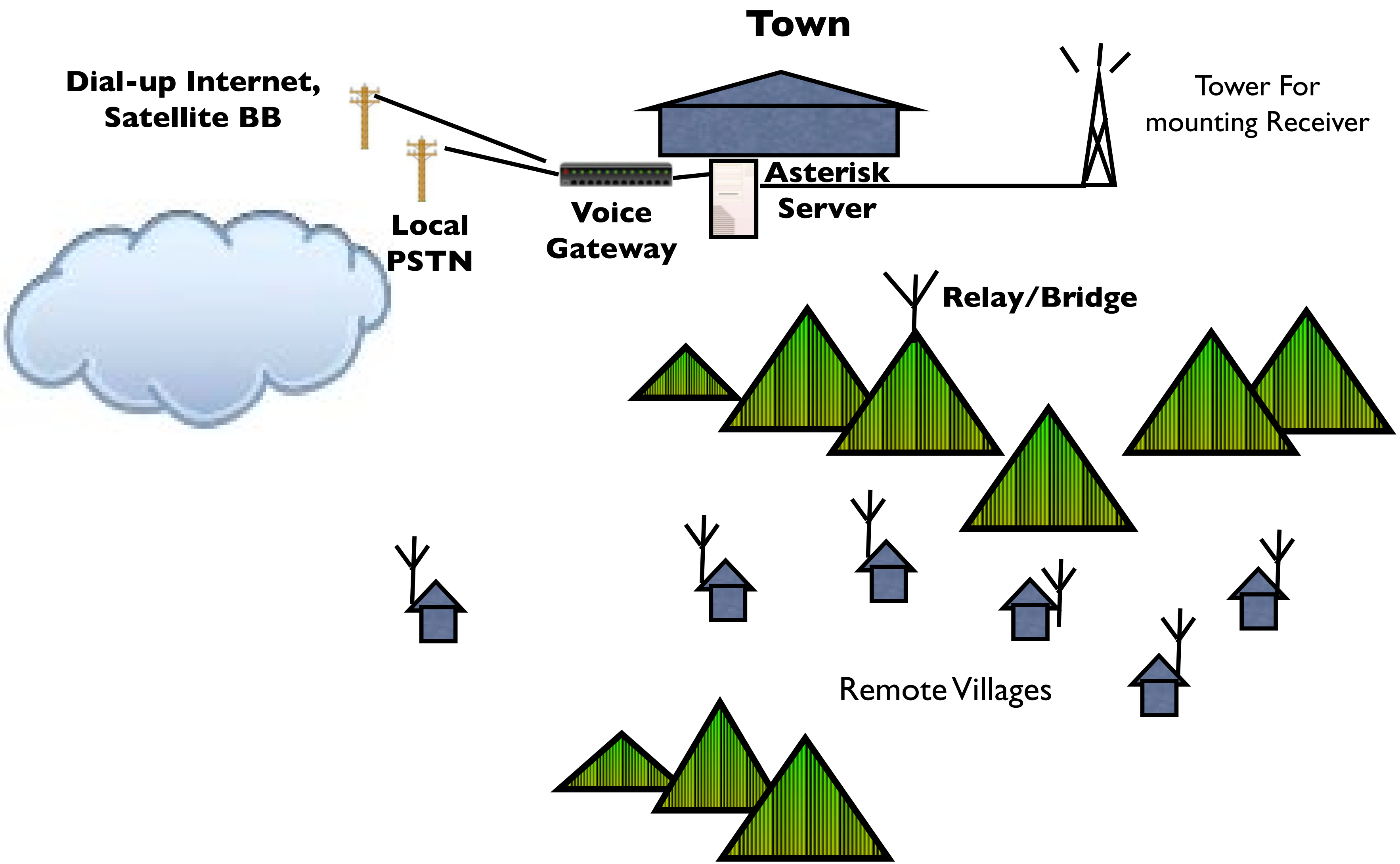
Local Village Wireless Network w/ SIP clients and Asterisk Server



Multi-Village Network w/ connection to PSTN



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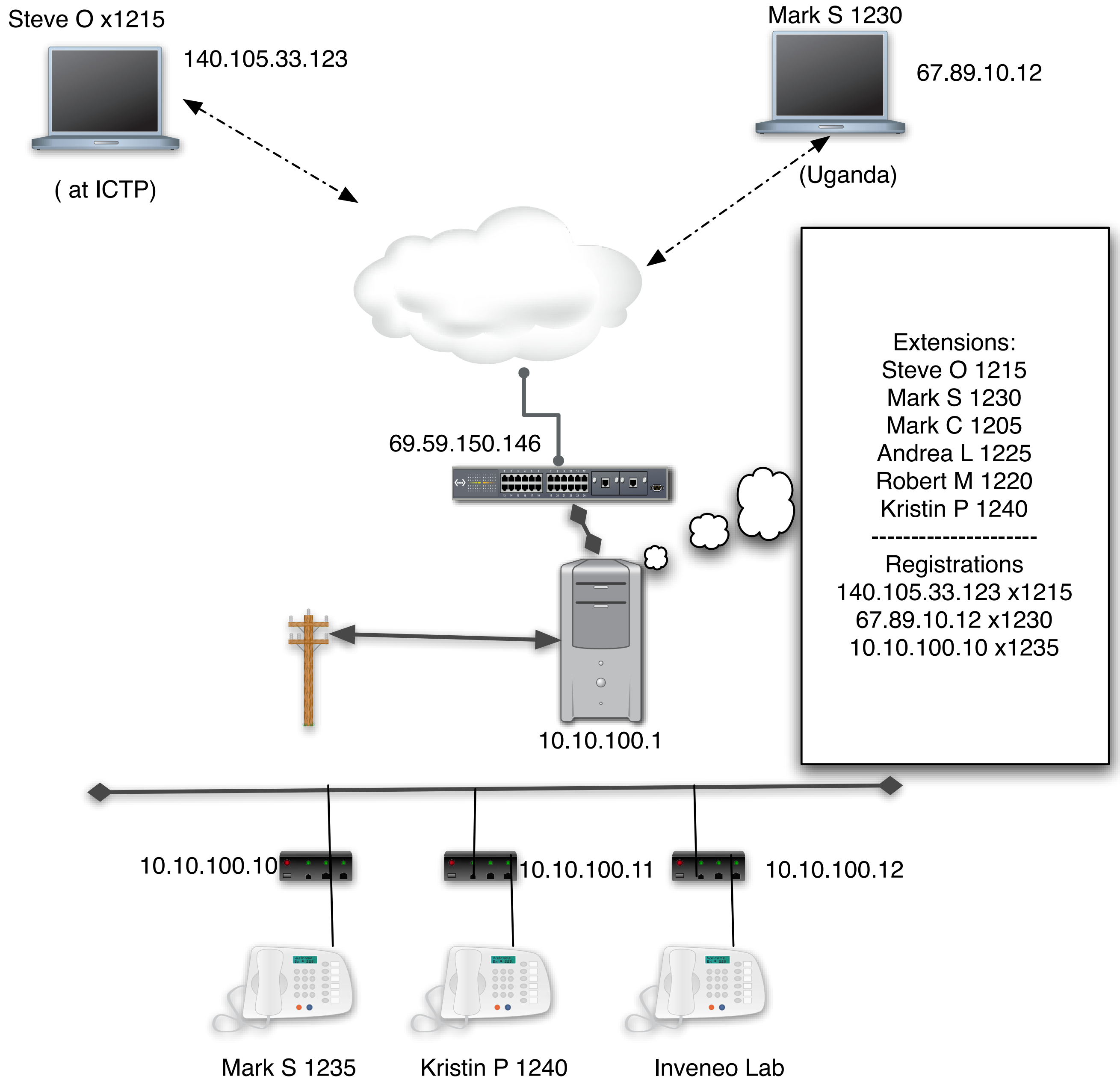
Open Source VOIP projects

- Asterisk(SIP-based PBX)
  - www.asterisk.org
- AstLinux
  - www.astlinux.org
- OpenH323(H.323)
  - www.openh323.org

Asterisk

- Open Source PBX project, in existence since 1999
- Available on Linux, OS X, Windows
- Supports SIP, H.323, H.264, IAX protocols
- Can route PC-to-PC, PC-to-PSTN, PSTN-to-PSTN calls
- Scriptable/Programmable

Registering extensions in Asterisk



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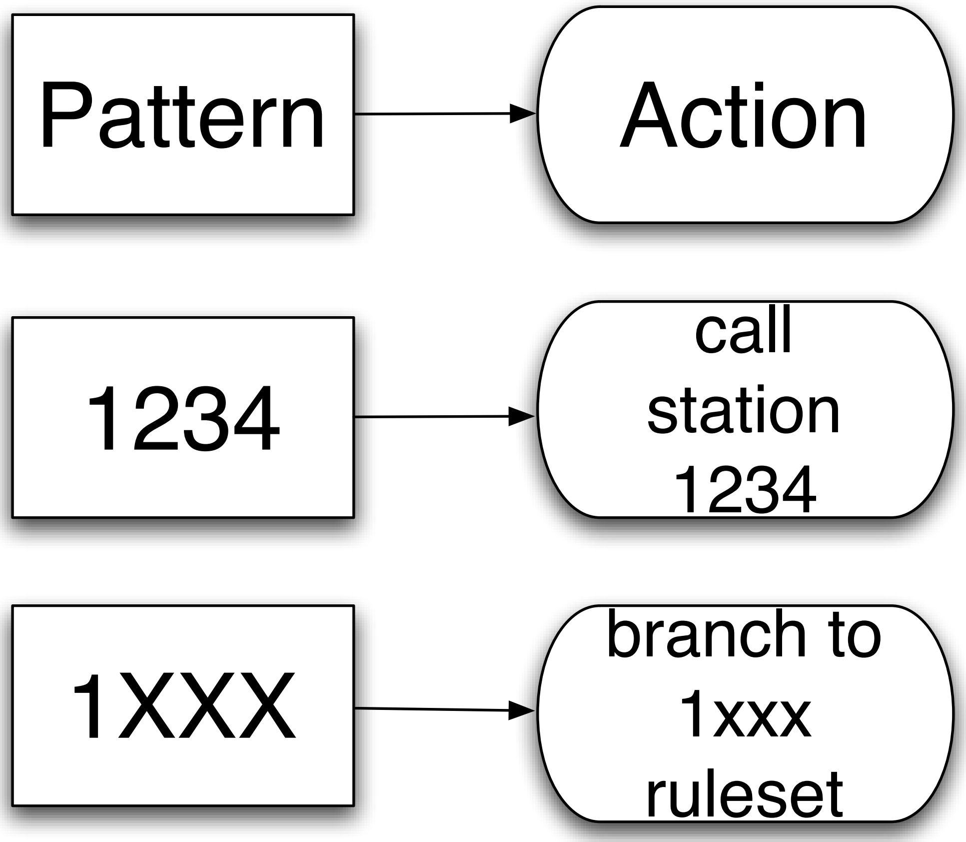
Call routing inside Asterisk

- Call-routing is done via a pattern-action mechanism



Call handling/routing in Asterisk

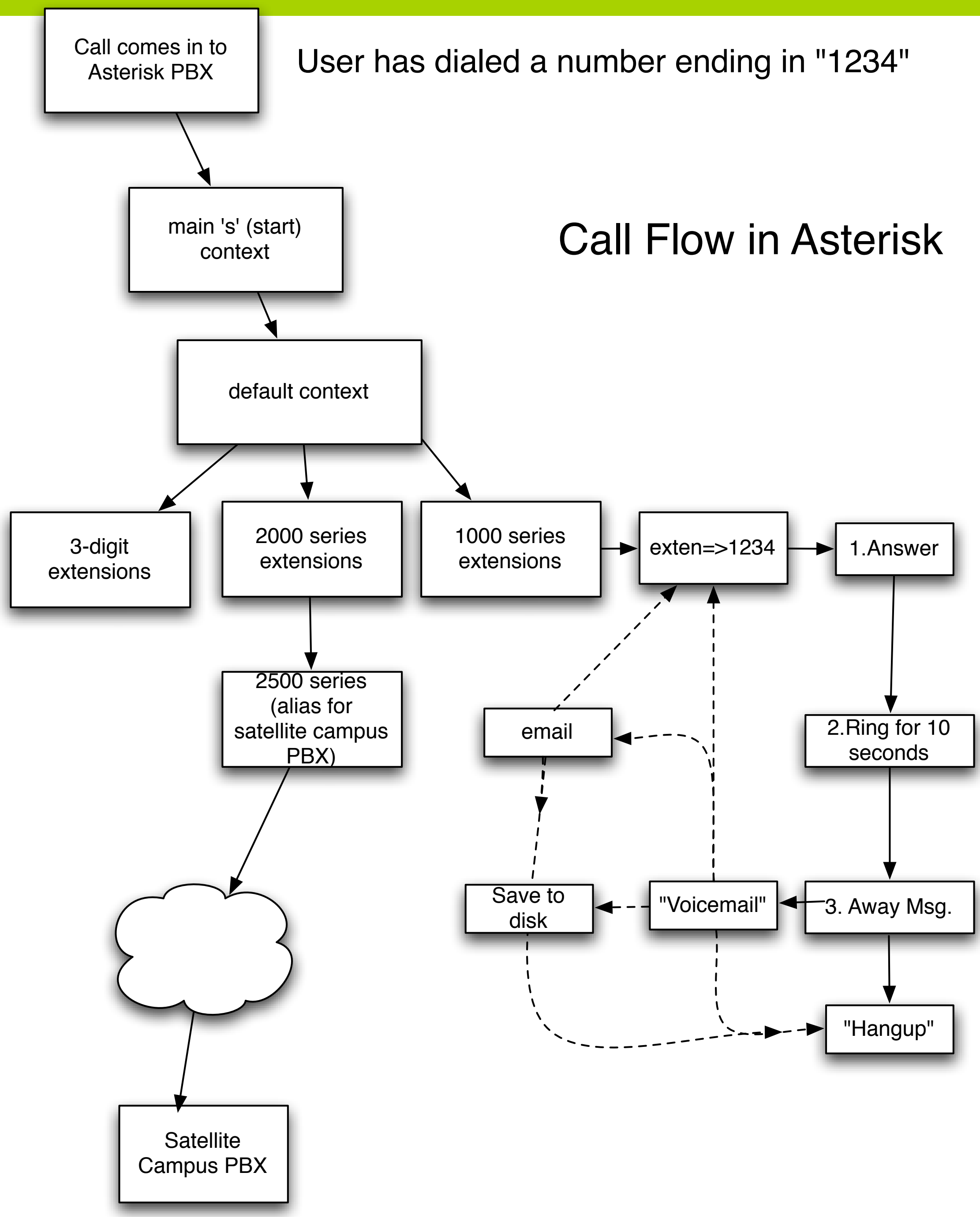
- Contexts
  - Each pattern and each action exists in a “context”
  - Contexts are groupings of patterns and actions
  - Patterns and actions can be in different contexts

Call handling/routing in Asterisk

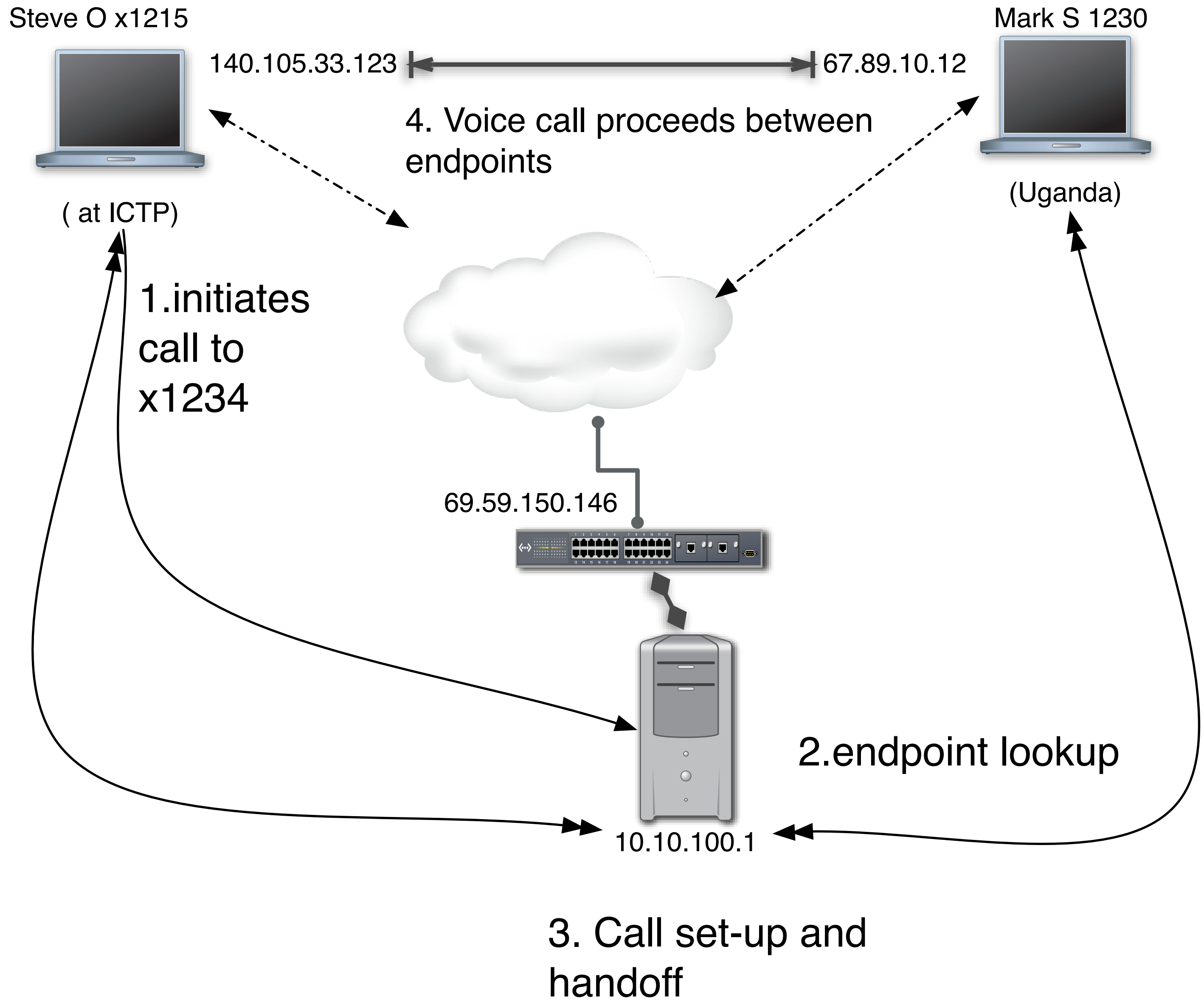
- Actions
  - are really small programs
  - can be further pattern matches
  - can do simple branching/looping
  - Actions may include running non-Asterisk programs
  - Actions MUST terminate

Call Flow in Asterisk

User has dialed a number ending in "1234"



Call handling in Asterisk



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Dialplans in Asterisk

- A dialplan is the sum total of the patterns and actions specified in the Asterisk configuration files
- Dialplan files in Asterisk
  - sip.conf --patterns
  - extensions.conf --actions
- Effective dialplans should be planned with the same detail you would plan a network

Voice CODECs

- A CODEC(enCOder/DECoder) is a method or algorithm for processing analog audio signals in a digital data stream
- Most commonly used for processing audio data sent over a network but can be streamed from a file.

Common VOIP CODECs

Name	Data Rate	License
G.711	64K	Free
G.726	16/24/32	Free
G.723.1	5.3-6.3	Proprietary
G.729A	8	Proprietary
GSM	13	Free
iLBC	13.3-15.2	Free
Speex	2.15-22.4	Free

Configuration issues on IP networks

- SIP -registration/setup on port 5060/5061
- H.323 defaults to ports 1718/1719
- IAX registration defaults to 4559
- typically runs into problems with multiple NAT layers

Asterisk Applications

- Actions are really applications/programs
  - Dial(), Playback(),Voicemail()
- Custom applications can be written using AGI, “Asterisk Gateway Interface”
- Support for Perl, Python, C, etc.
- Uses Linux stdio to pass data between external applications and Asterisk

Reference Links

- VOIP-Info Wiki - www.voip-info.org
- Asterisk - www.asterisk.org
  - Astlinux - www.astlinux.org
  - Trixbox - www.trixbox.org
- SJPhone (SIP client) www.sjlabs.com
- OpenH323 - www.openh323.org
- Asterisk-compatible VOIP hardware - www.digium.com
- Inveneo - www.inveneo.org