VoIP and Asterisk

K. M. Peterson

BBLISA • 8 May 2013

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http://kmpeterson.com/special/bblisa-asterisk

KM
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Overview

- The general map of things
- Hardware for VoIP
- SIP and Related Protocols
- How Asterisk is organized, and where to get it
- How VoIP protocols and practice are implemented in Asterisk
- An Example with Asterisk
- Issues: Voice Quality and Security

Why are we talking about this?

- How is it different from “the telephone”?
    - **The current network (the PSTN) was engineered specifically for voice**
    - **Design and implementation were performed in a very different environment**
    - **TDM vs. Packet Switching**

Why are we talking about this?

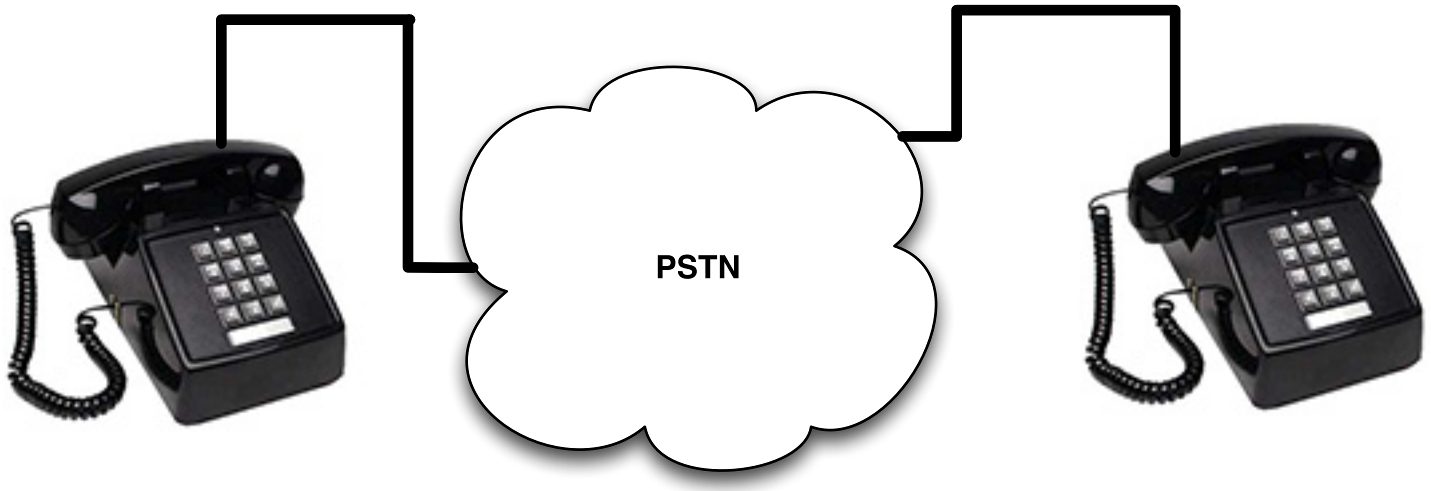
- The Challenges
  - **Everyone knows how telephones work.**
  - **Service is universal, along with numbering.**
  - **The Internet is not reliable. (See: jitter, latency, routing)**
  - **Entrenched interests ("The Phone Compan(-ies)")**

- Why Indeed?
  - **Cost**
  - **Features**

Goals

- For this presentation
  - **Understand enough of the technology to ask good questions**
  - **Implement it yourself**
  - **Help your organization/colleagues**
  - **Where to go next**
- Enterprise Implementation
  - **Costs**
  - **Functionality**
  - **Services**

The General Map of Things



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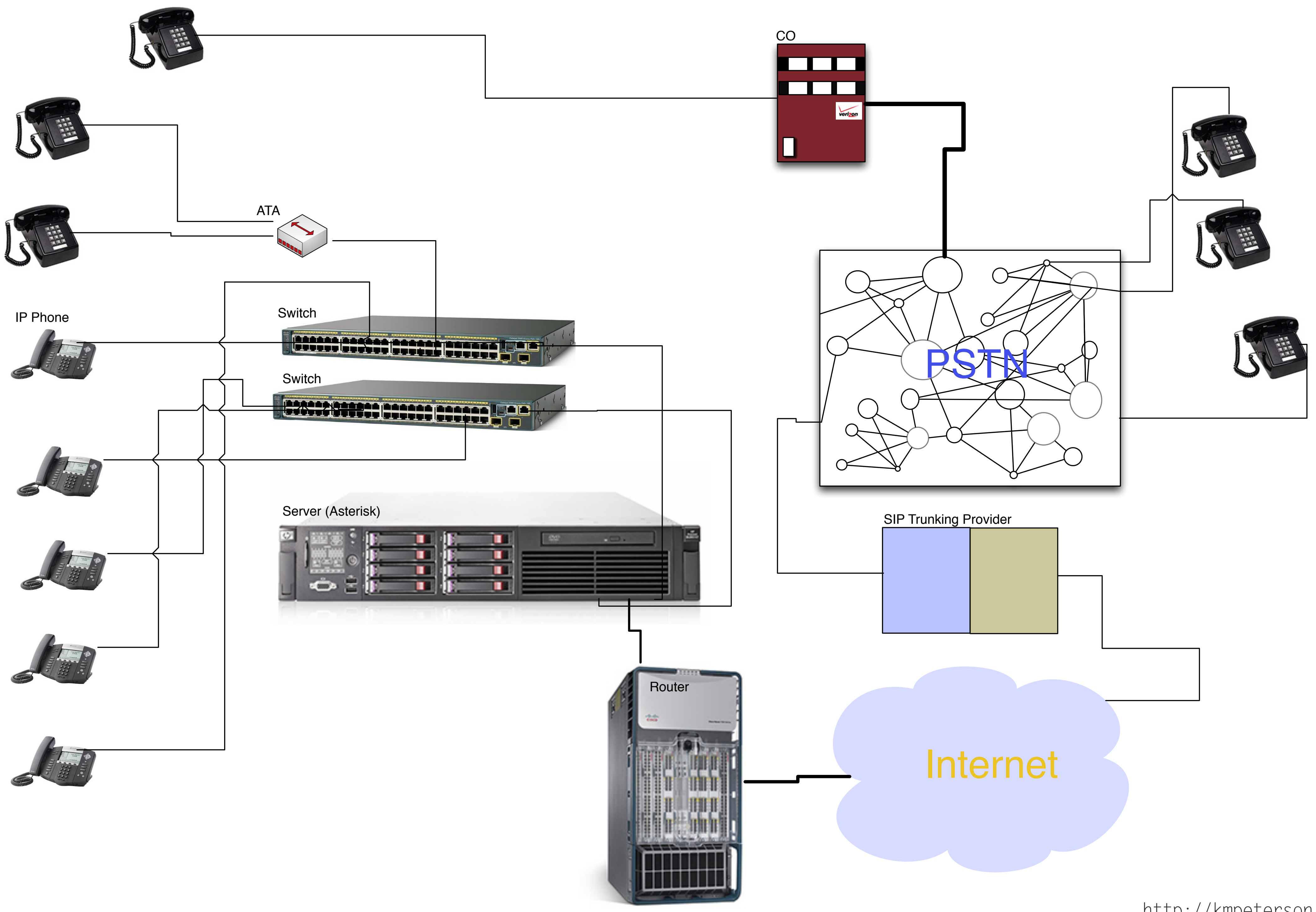
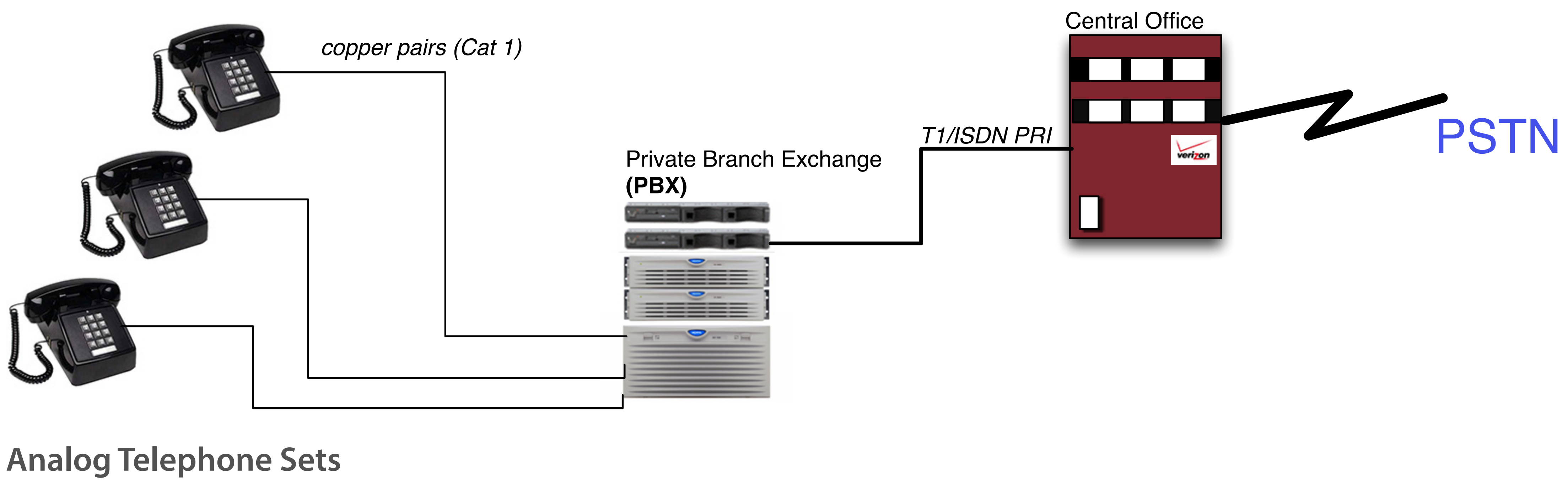


Corporate Classic



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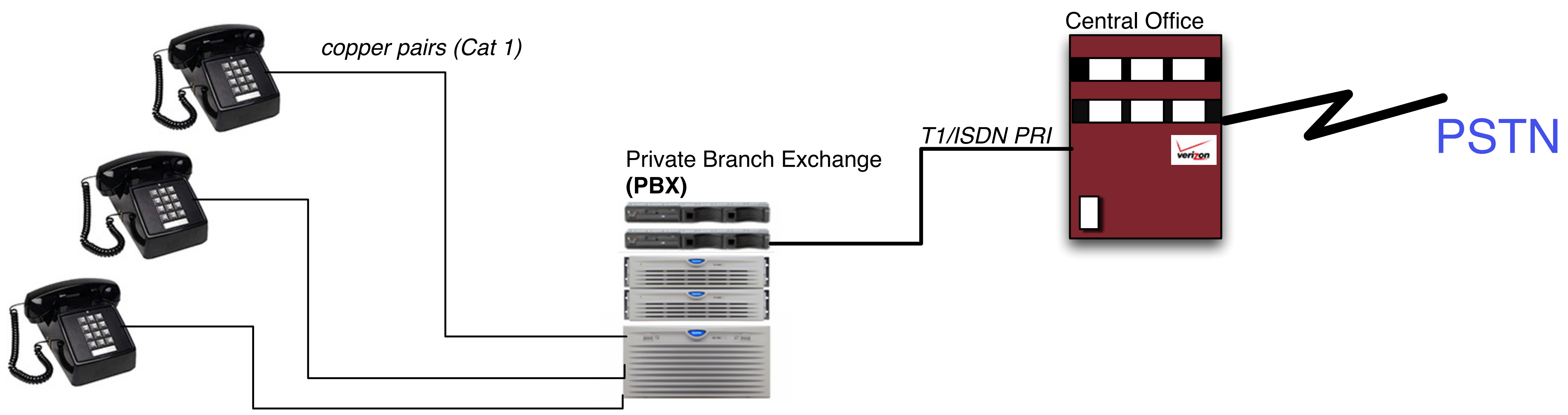

Analog Telephone Sets

Distributed Functionality

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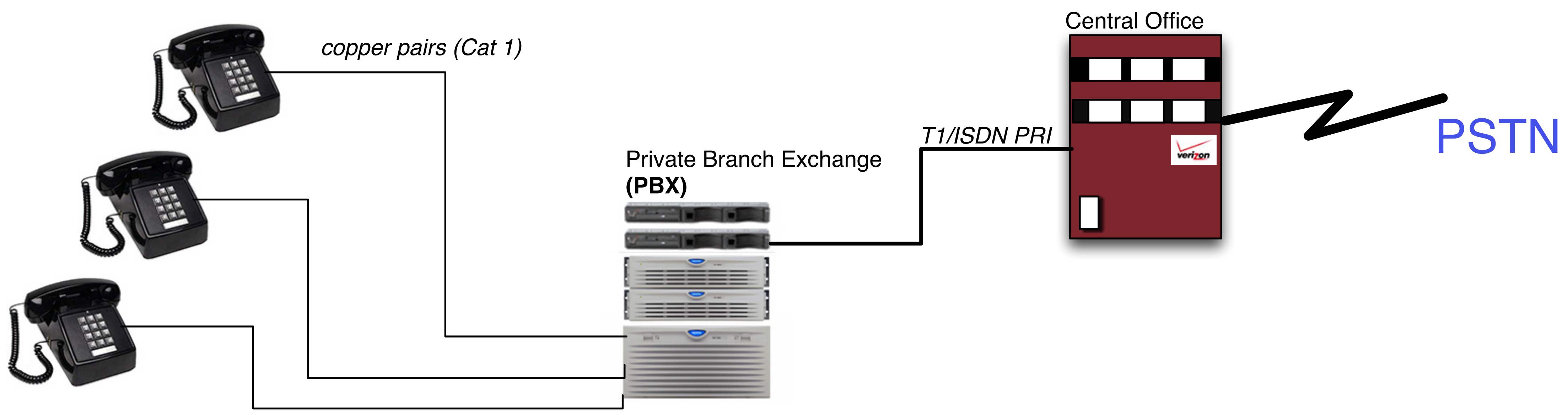
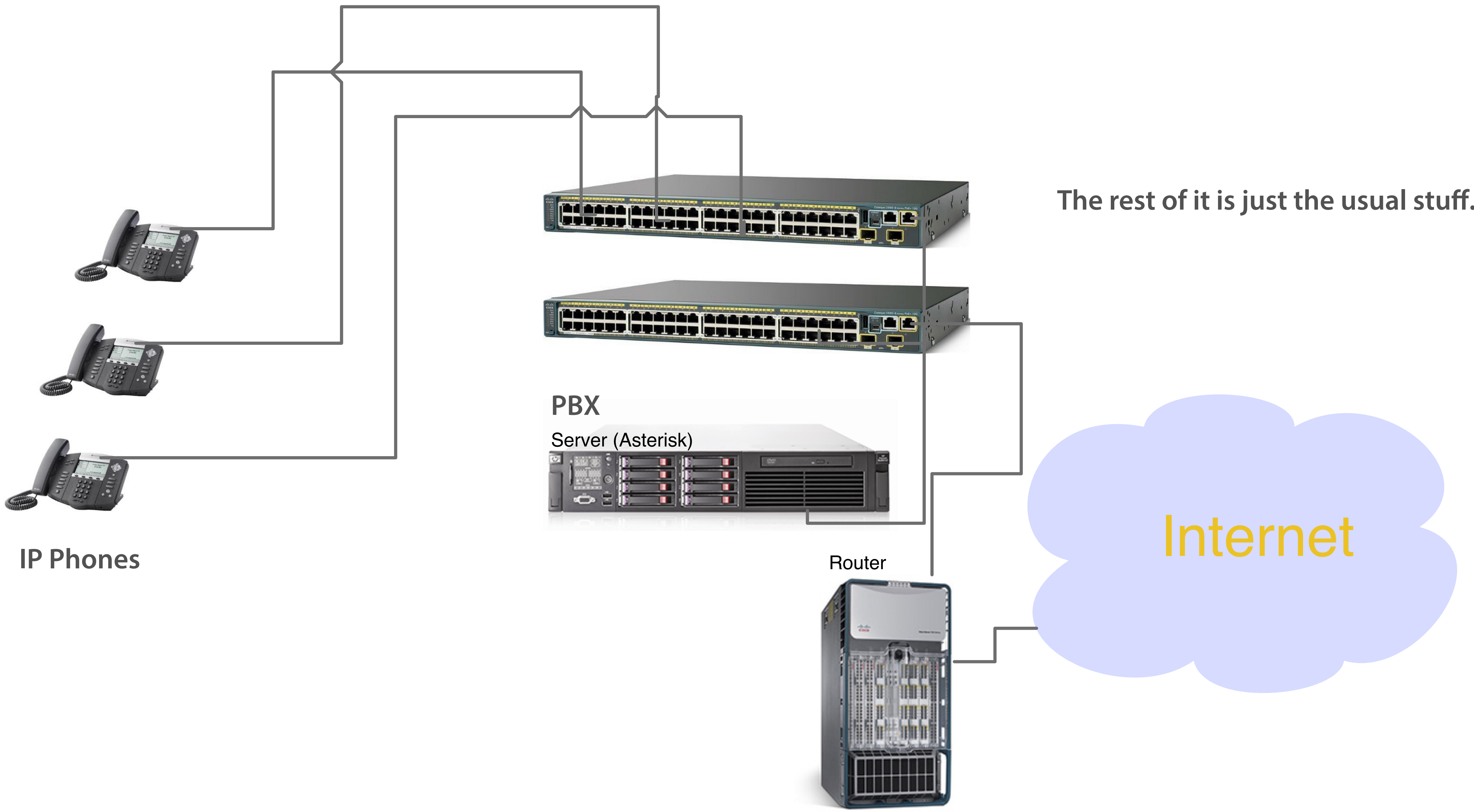


Analog Telephone Sets

PBX Features for Subscribers:
Local (extension) dialing
Call Pickup/Call Park
Voicemail
Billing/Accounting
Long-Distance Management
DID

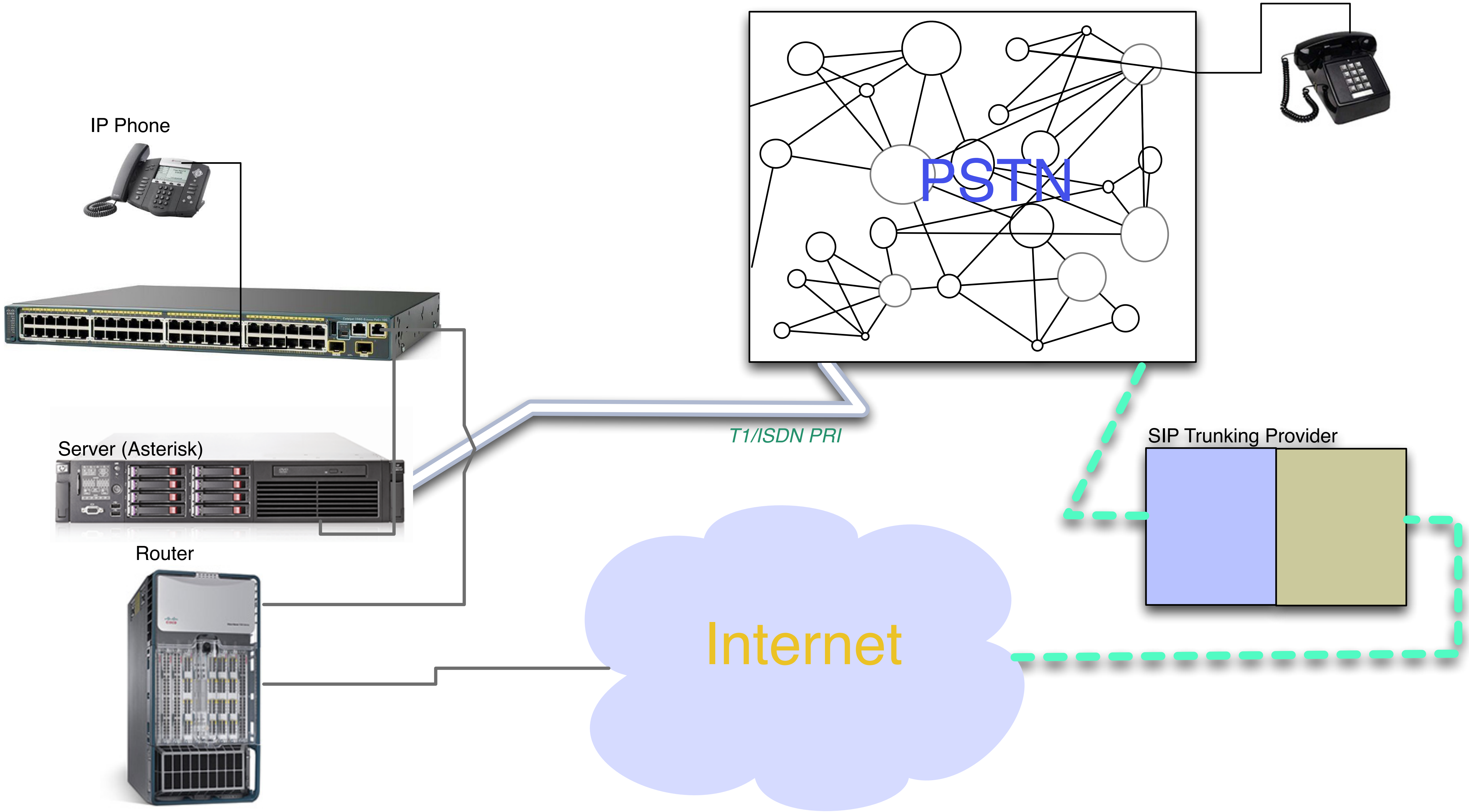
Distributed Functionality

Connect Phones to Network



The rest of it is just the usual stuff.

IP to the PSTN



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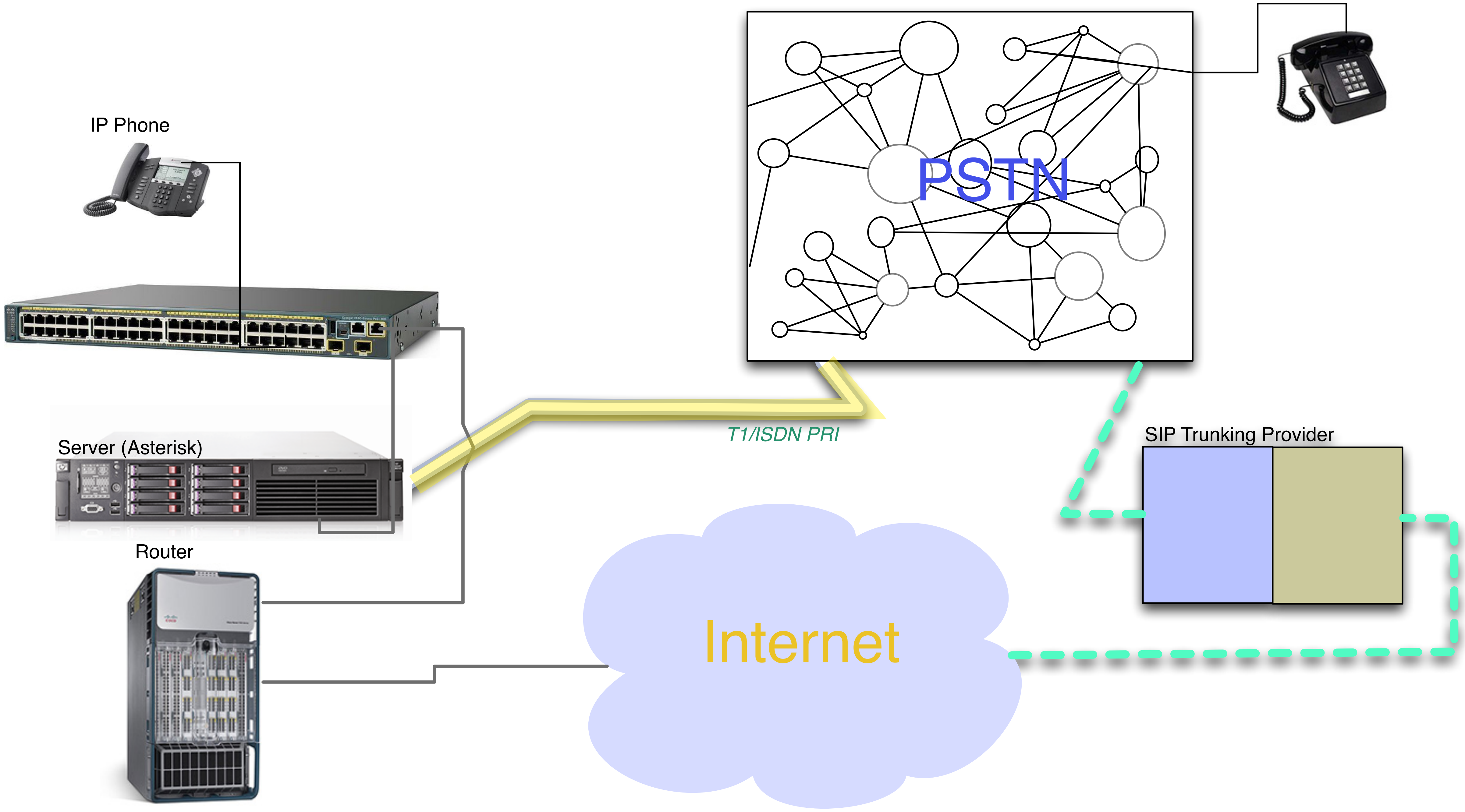

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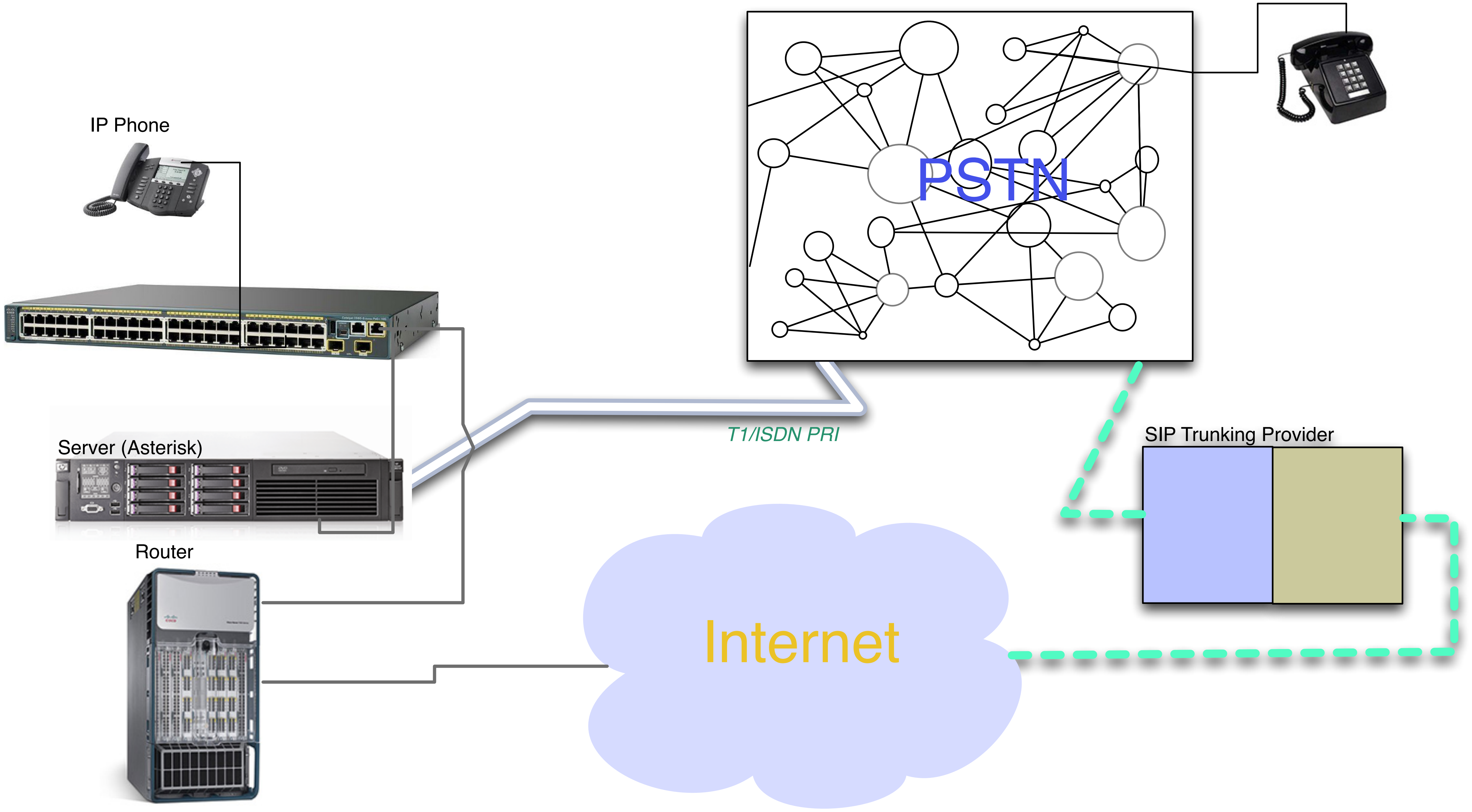

Hardware

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IP Phone (Polycom Soundpoint IP550)
Ethernet 10/100 (+ PoE)
Integrated Switch
Speakerphone

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Microbrowser
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T1/E1 connection
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Cisco SPA112
Analog Telephone Adaptor (ATA)
100 Mbps Ethernet
2 Analog (RJ11) (FXS)

The Protocols

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Once again, countless:

We'll concentrate on SIP and friends.

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Two parts of a phone call

1. Initial signaling and setup
2. Media

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SDP: Session Description Protocol

Supports negotiation of which codecs and media to be used.

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SDP: Session Description Protocol

Supports negotiation of which codecs and media to be used.

RTP: Real Time Protocol

Actual streaming media.

Registration

10.18.14.19
desk1

10.18.14.1
ast

10.18.14.6
desk2

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REGISTER →

From: <sip:*userid*@10.18.14.1>
To: <sip:*userid*@10.18.14.1>
Via: SIP/2.0/UDP 10.18.14.19:5060
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE. REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH

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Allow: INVITE, ACK, CANCEL, OPTIONS, BYE. REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH

OK

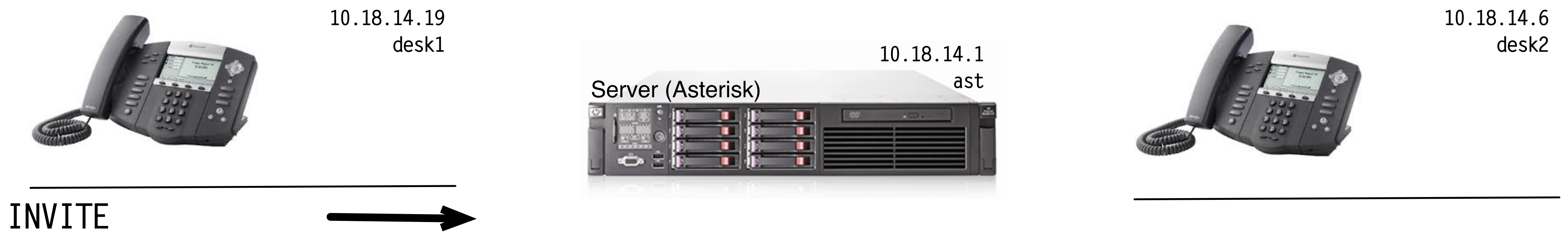
Expires: 300

Registration:
Uses authentication (can be secured)

Response:
Server has a timer to re-register.

Simple Call Setup

Simple Call Setup



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Server (Asterisk)

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INVITE →

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Media Description: audio *port* RTP

Media Attribute: *codec*/8000 ...

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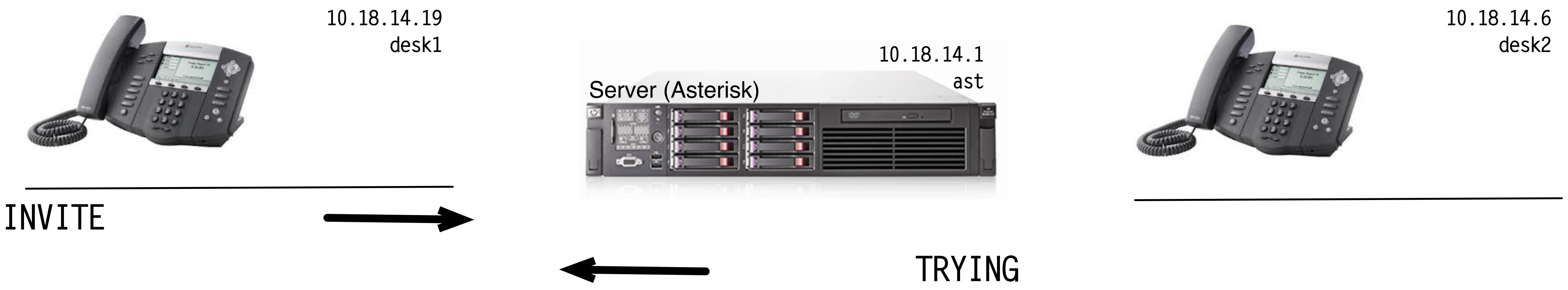

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INVITE

TRYING

INVITE

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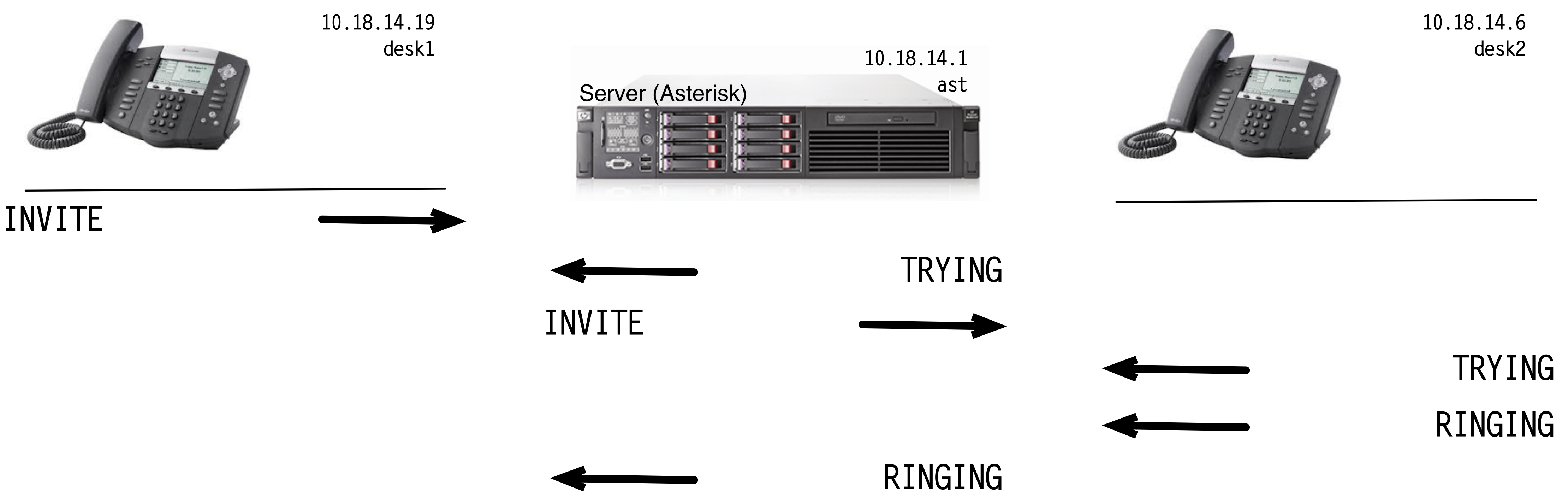

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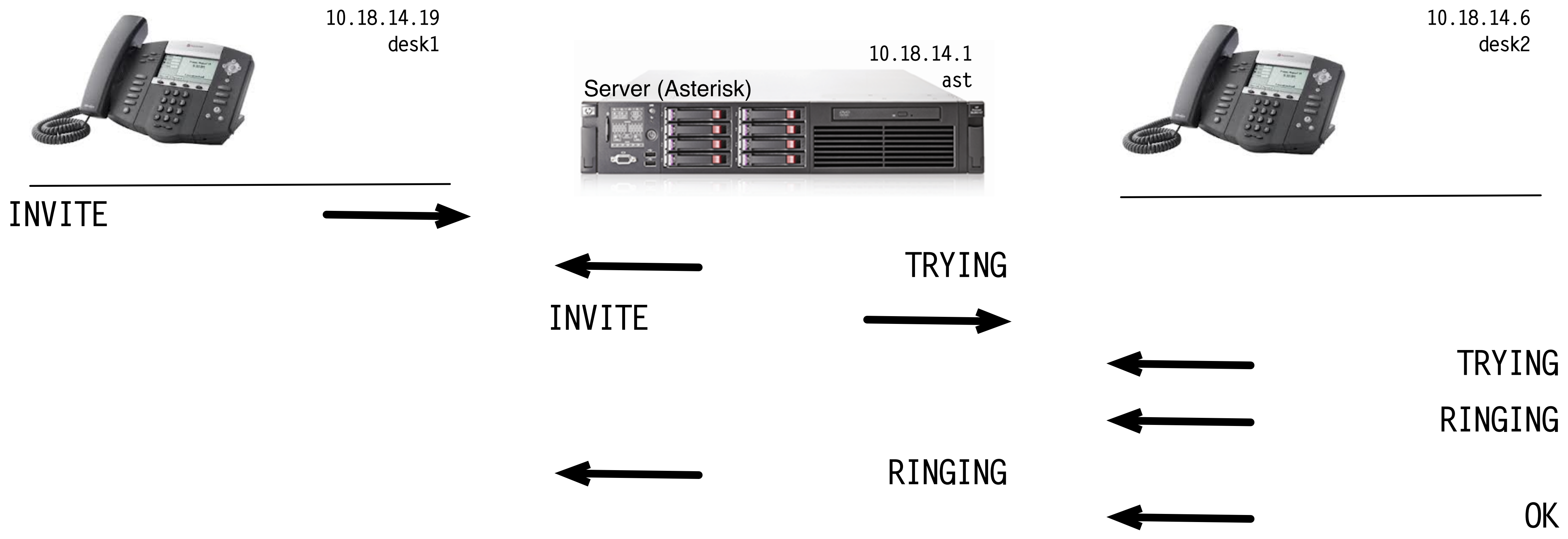

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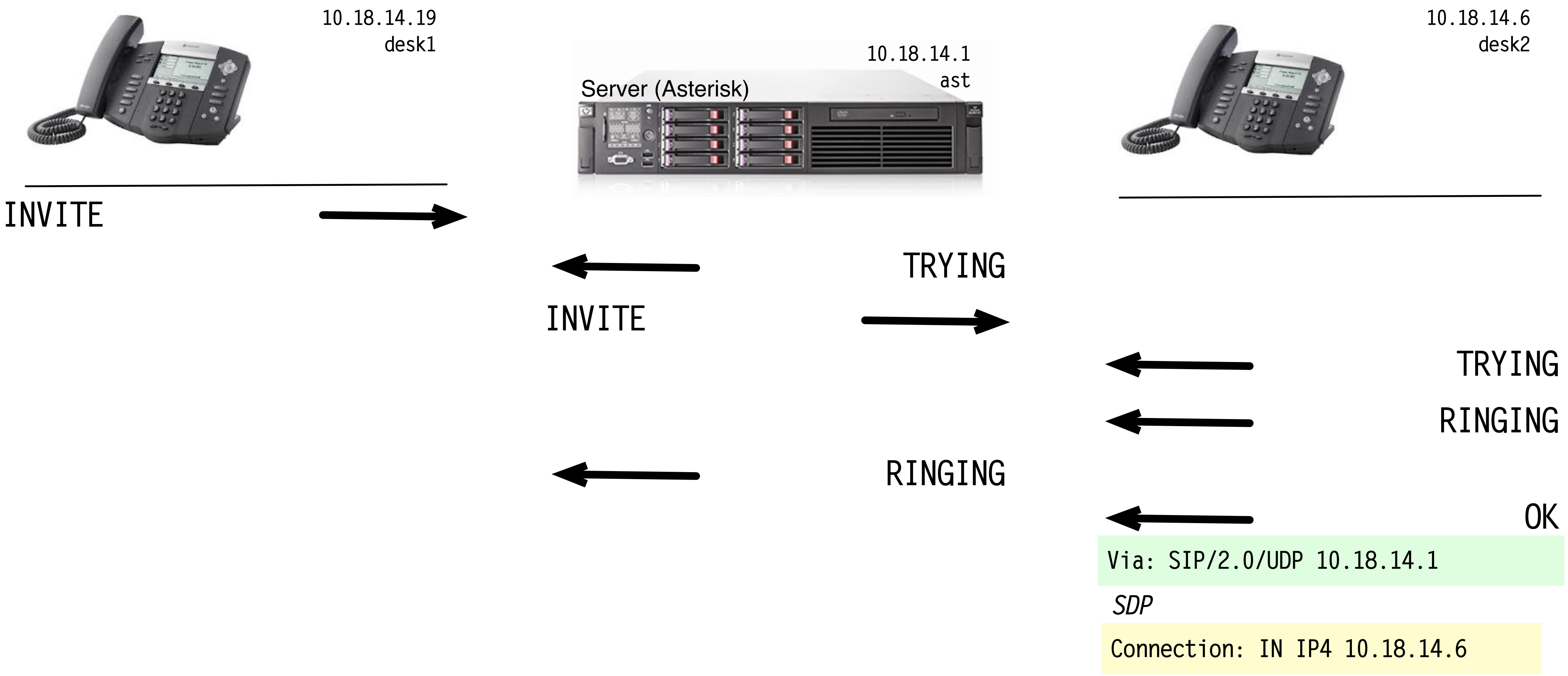

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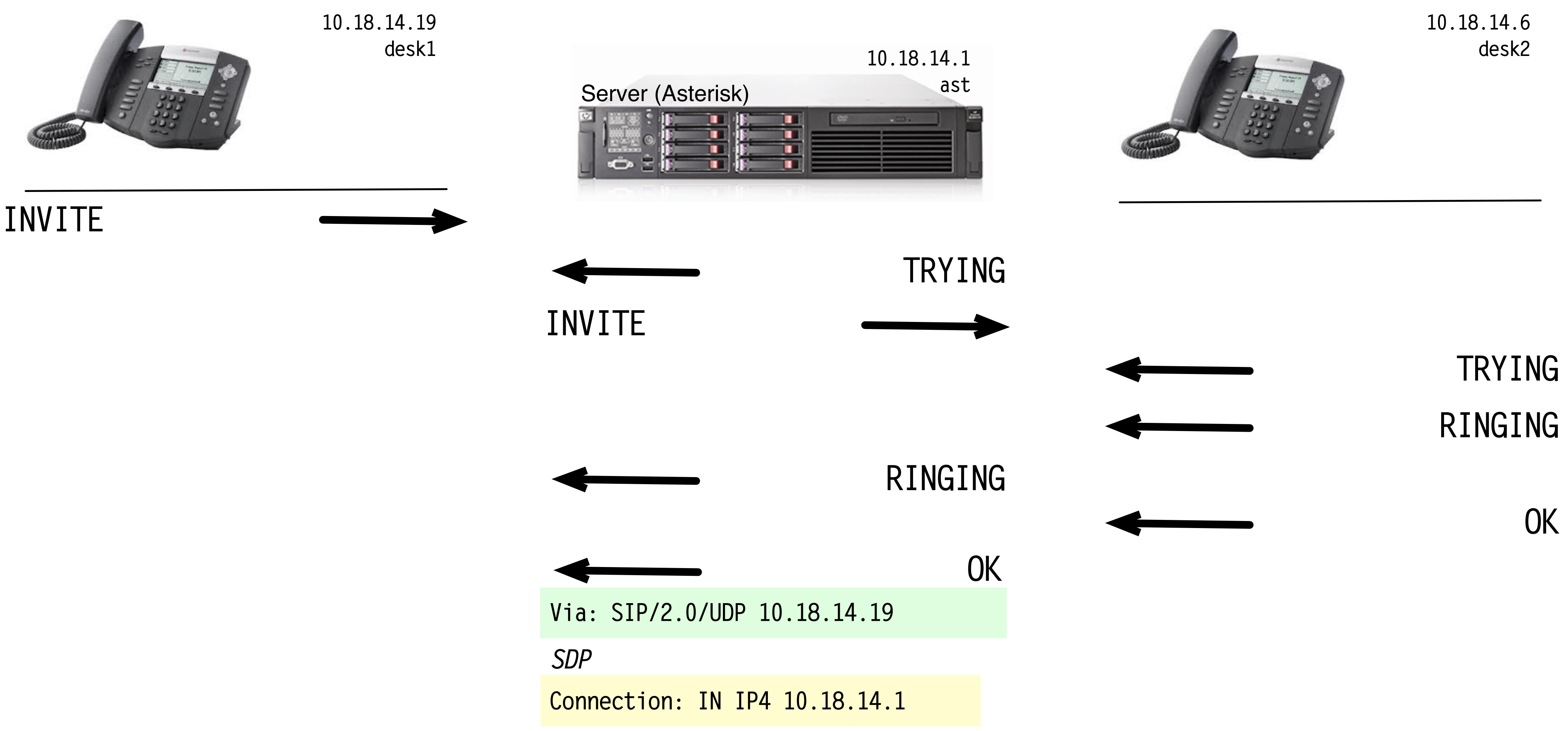

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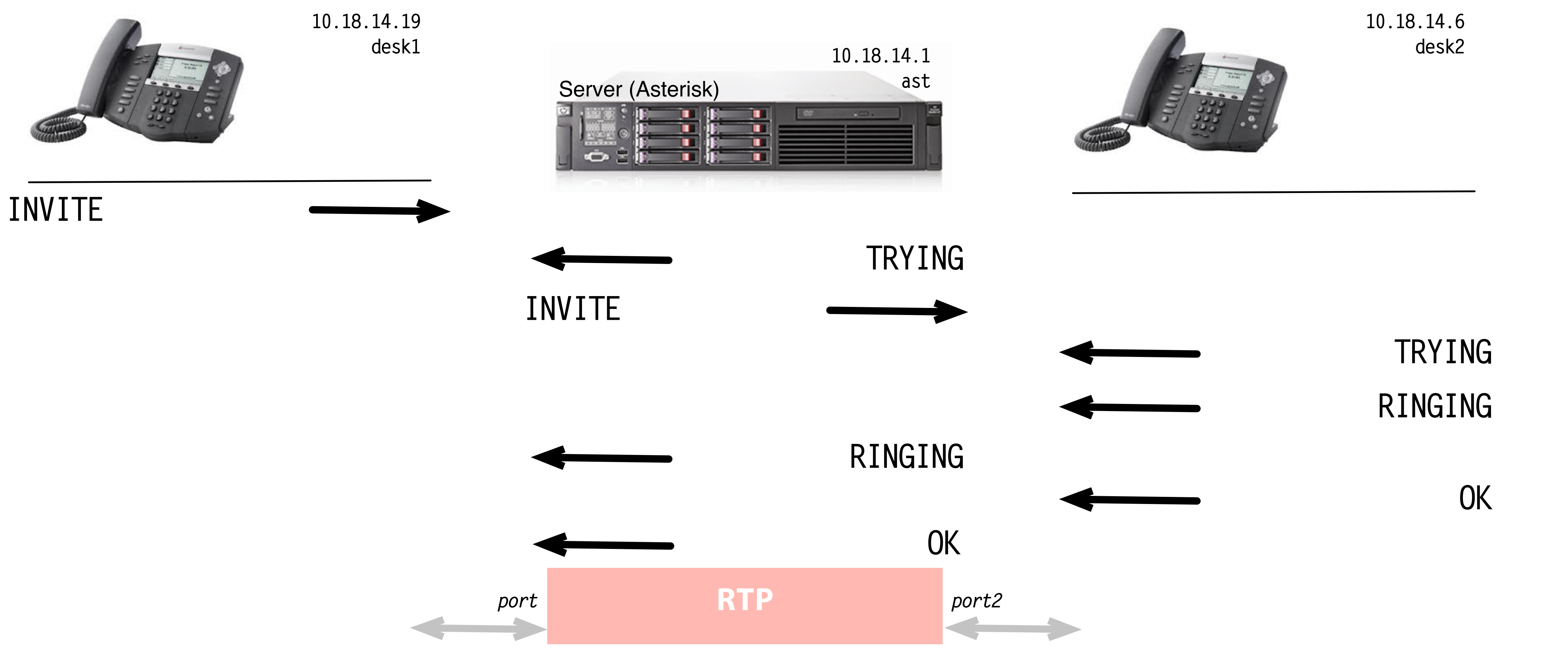

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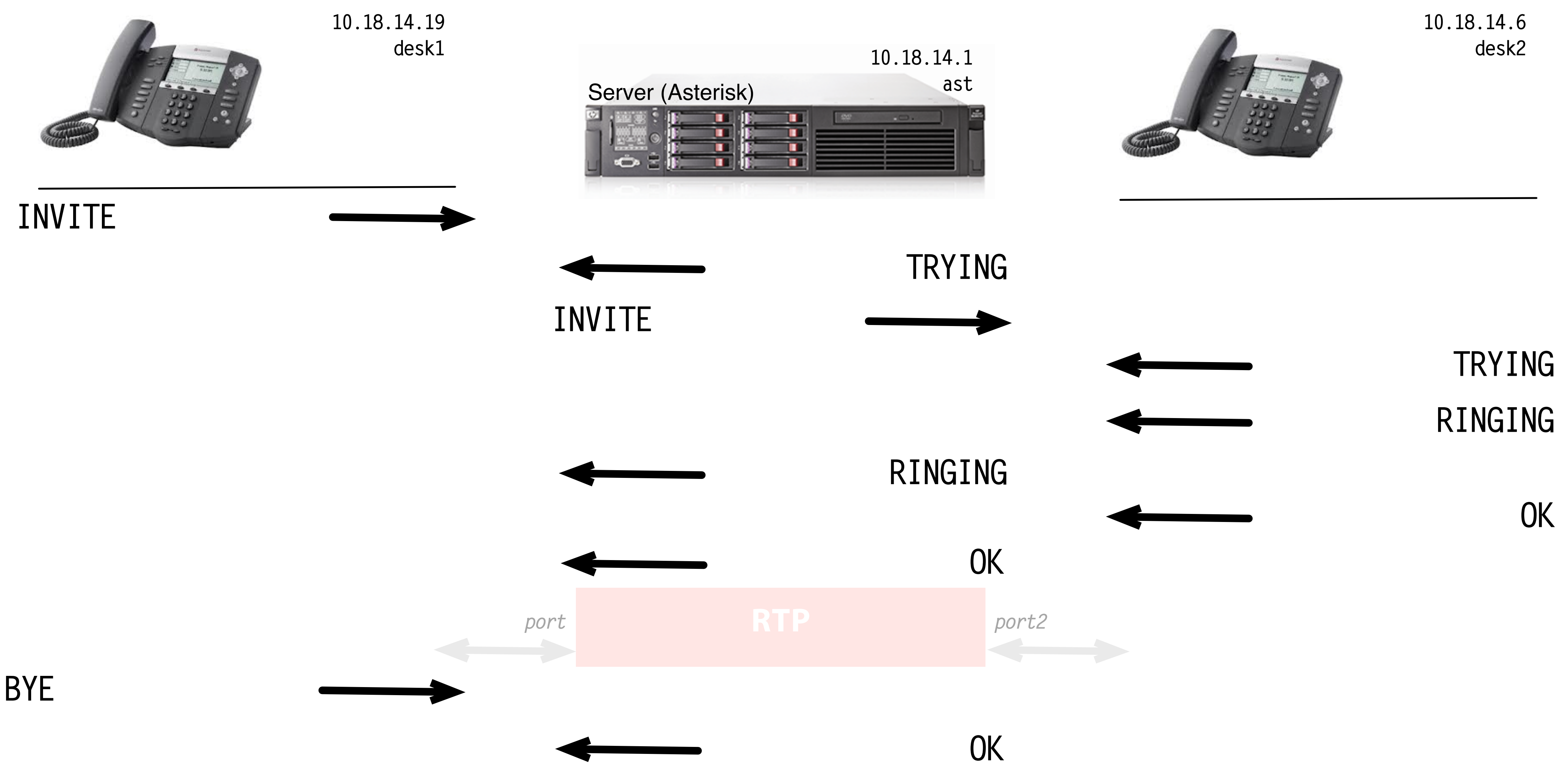

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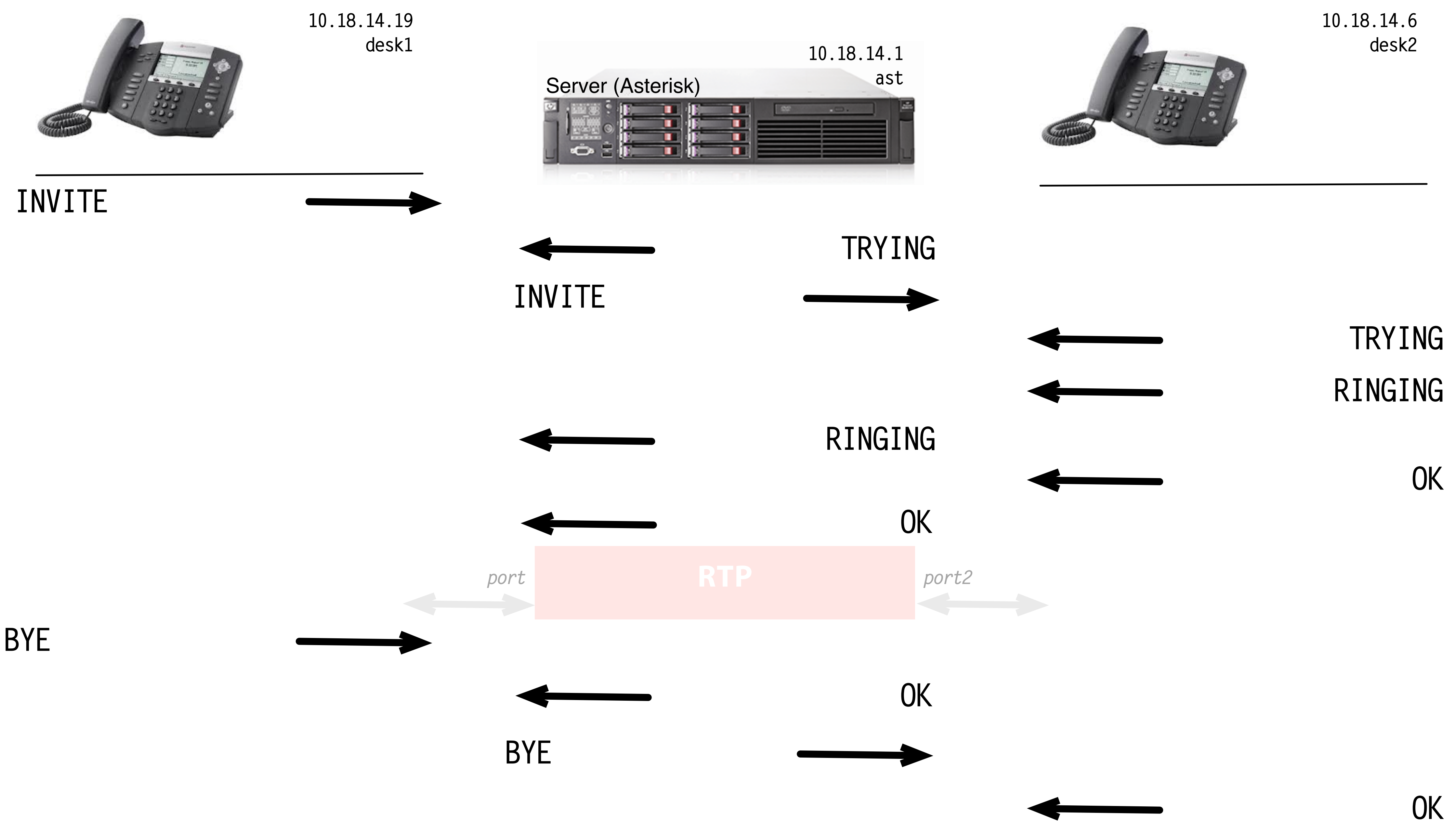

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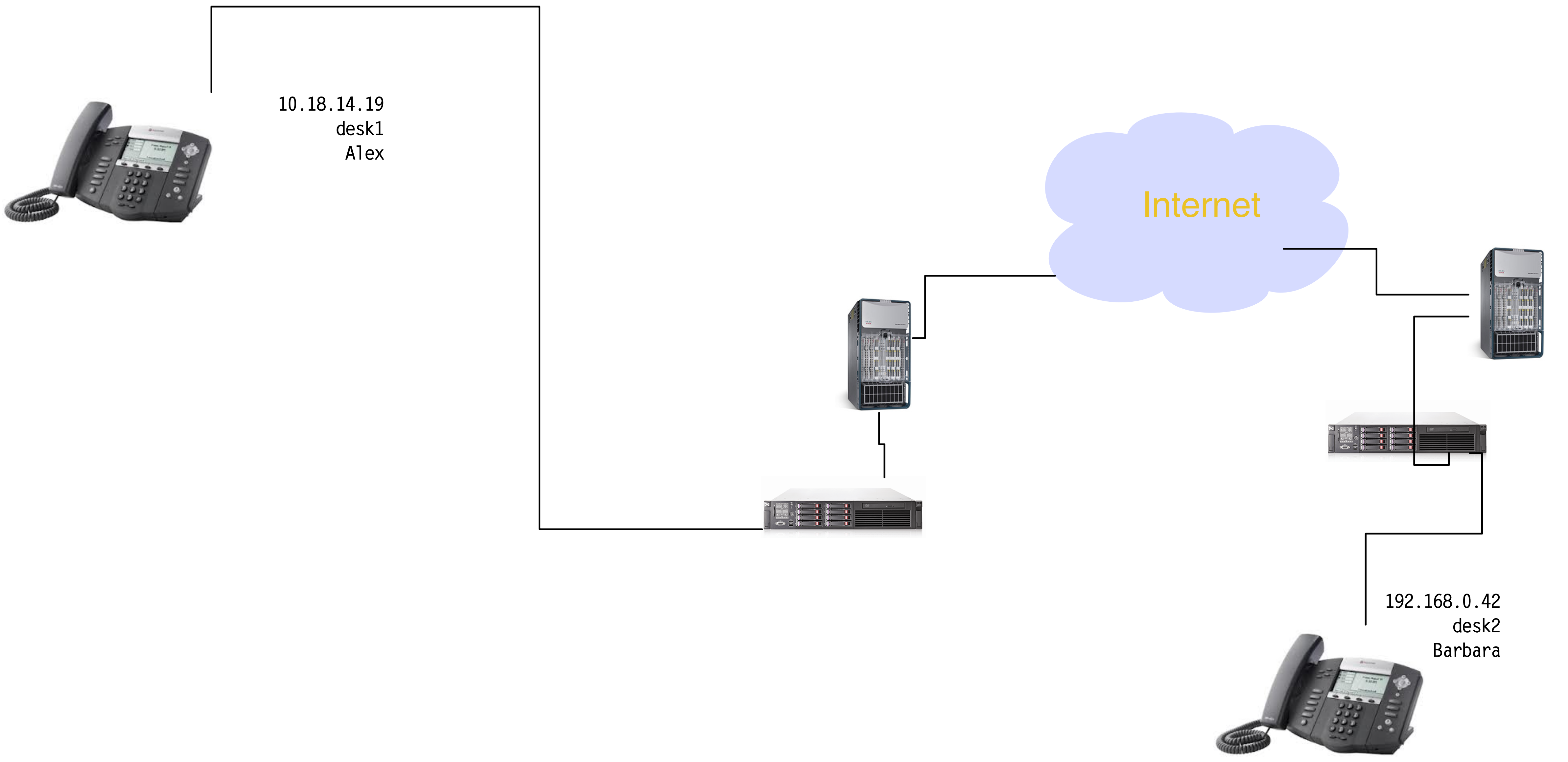

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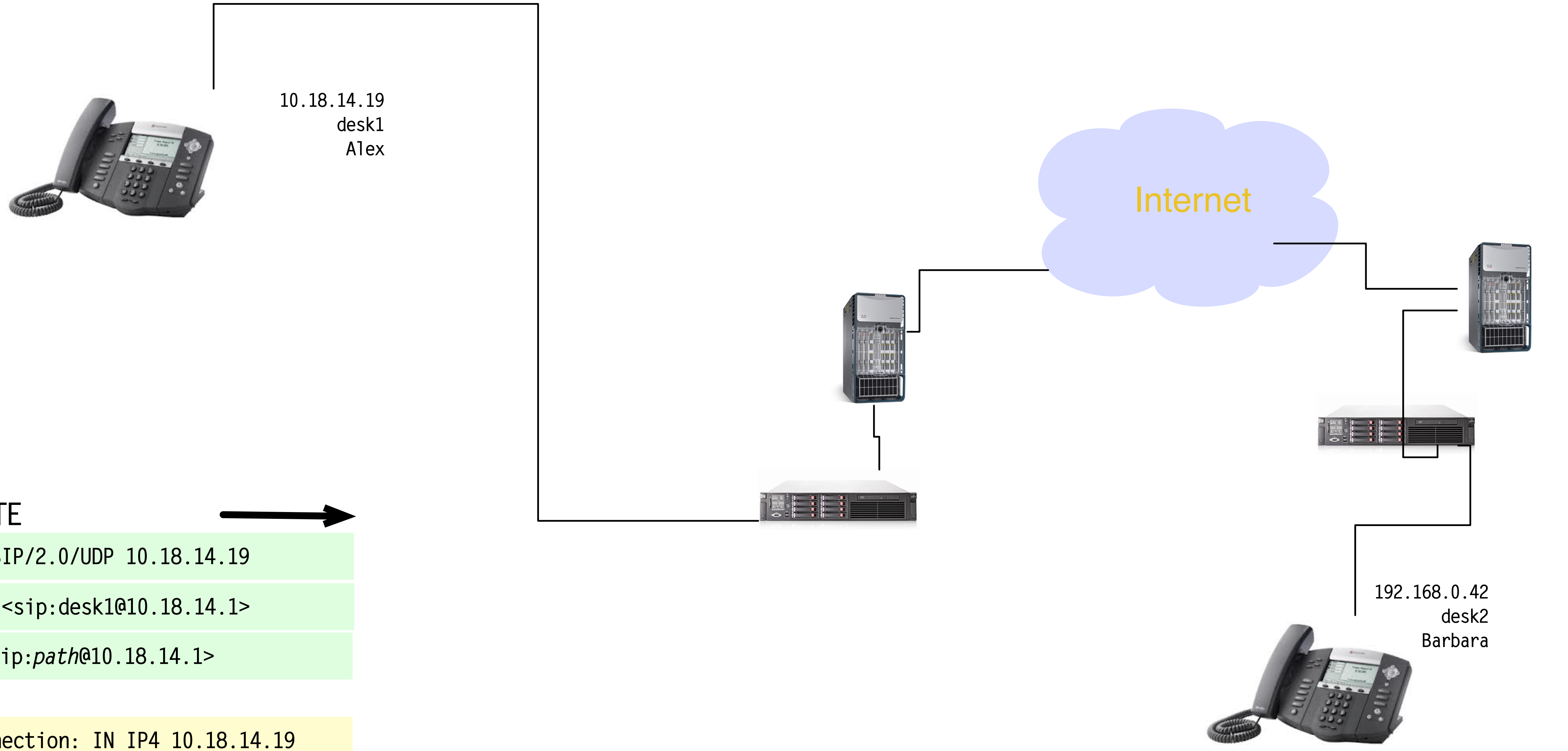

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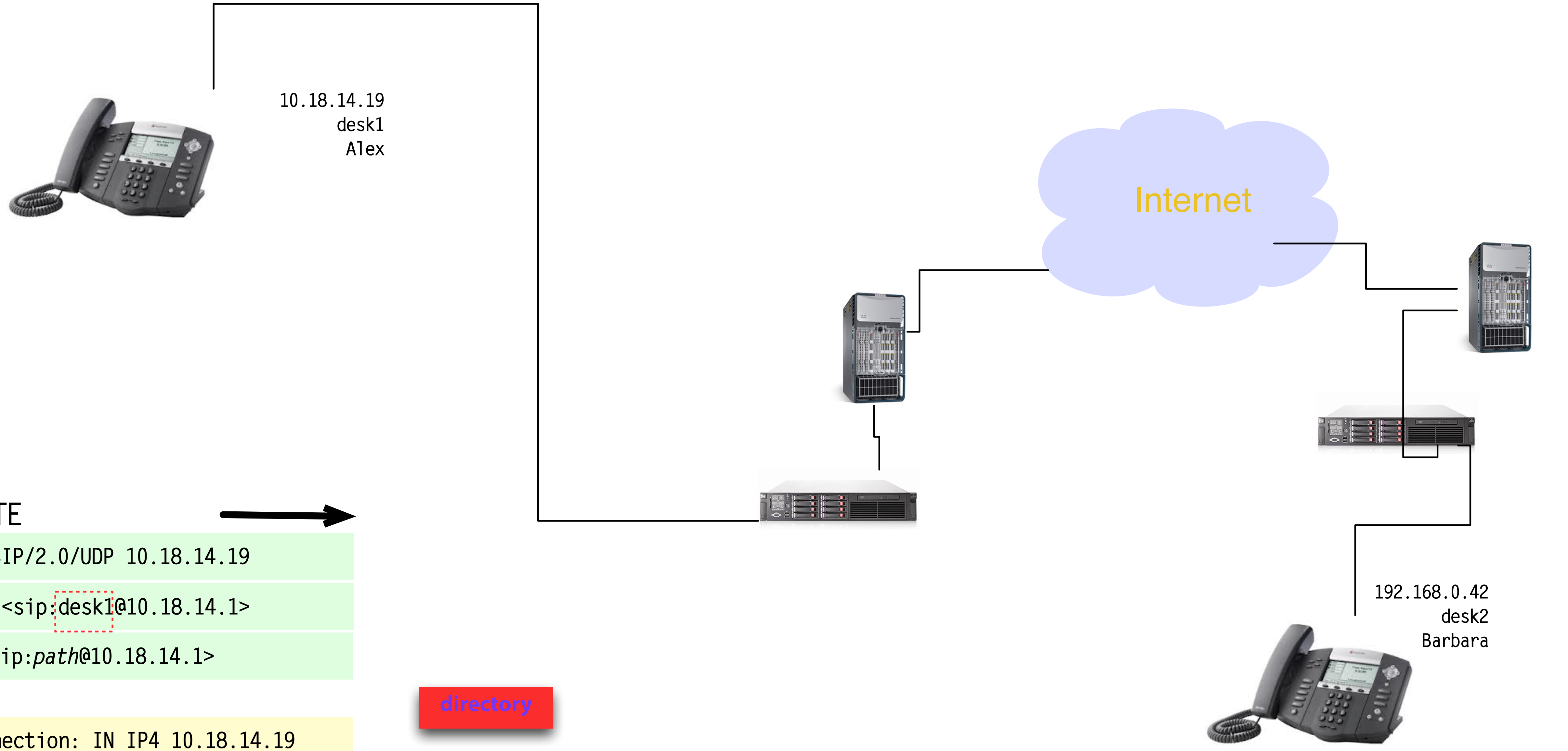

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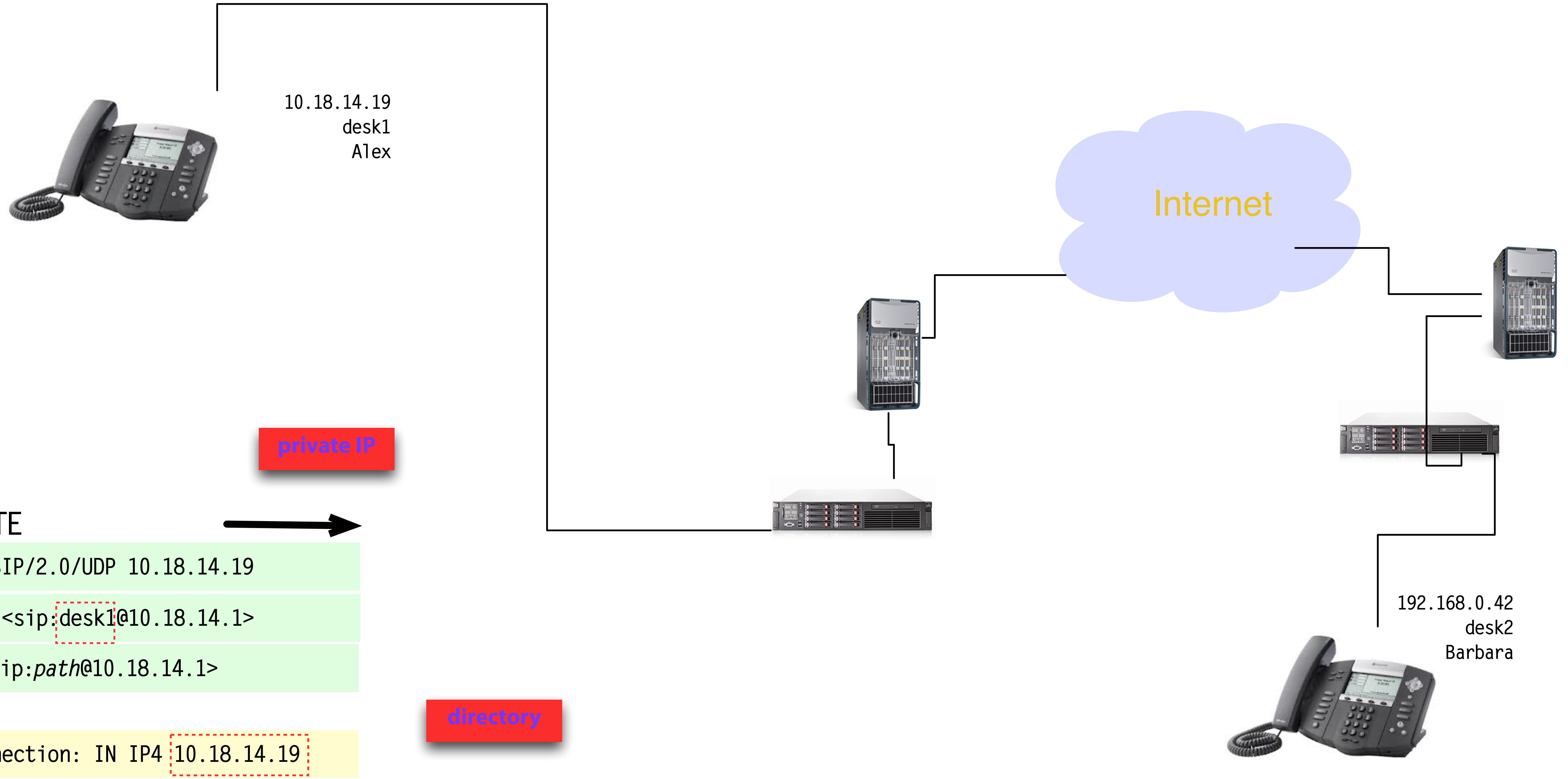

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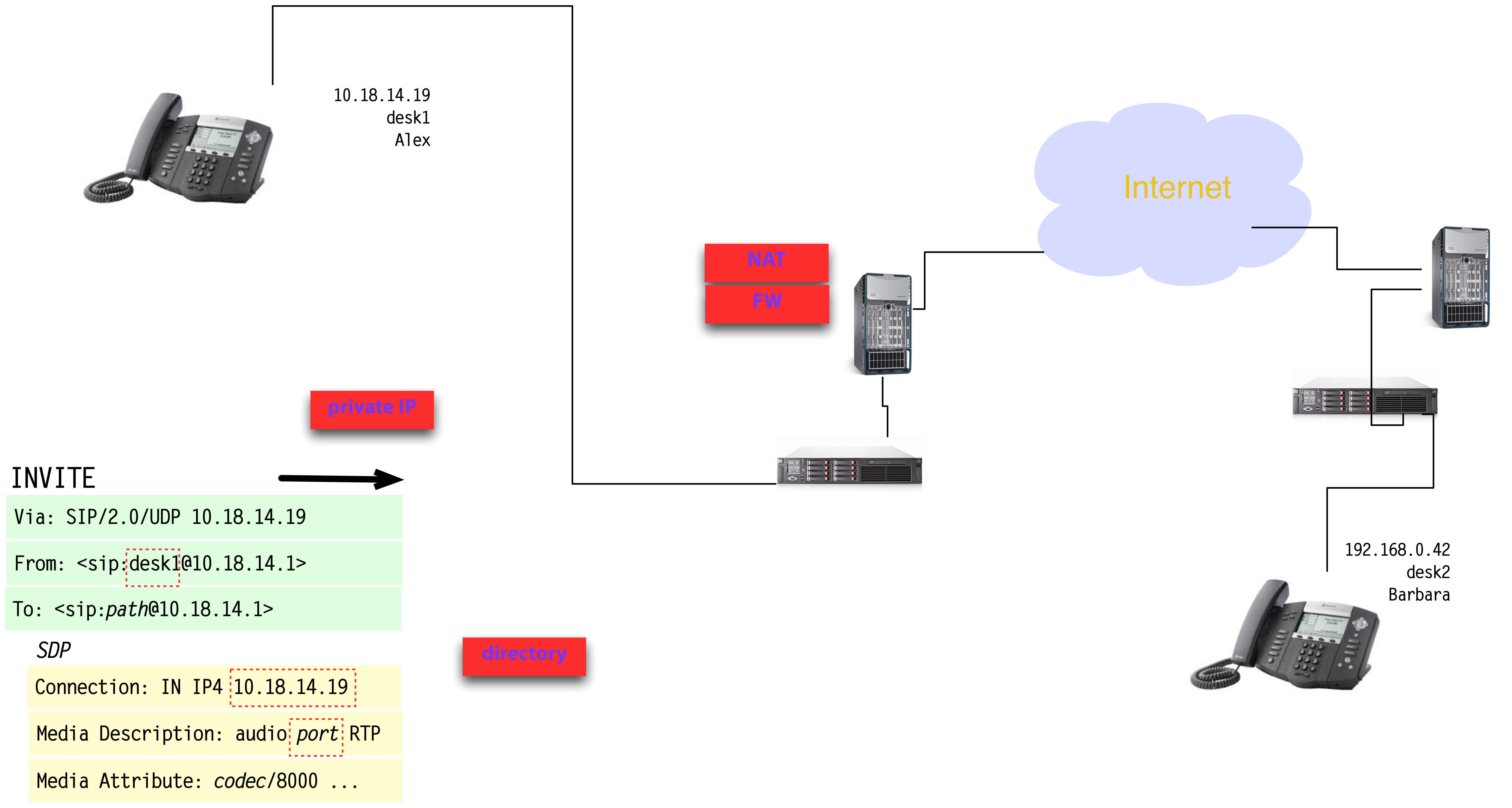

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Not a “reliable” transport.

- *More efficient, lighter-weight*

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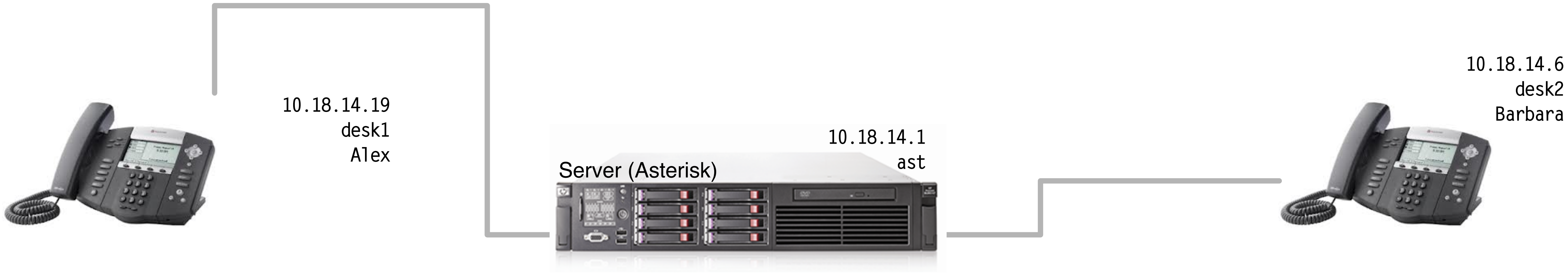
- *Lessons about QoS: traffic shaping, rate limiting, measurement*
- *"Buffer Bloat" and the counterintuitive strategy of speeding up by applying the brakes.*

The Media Stream



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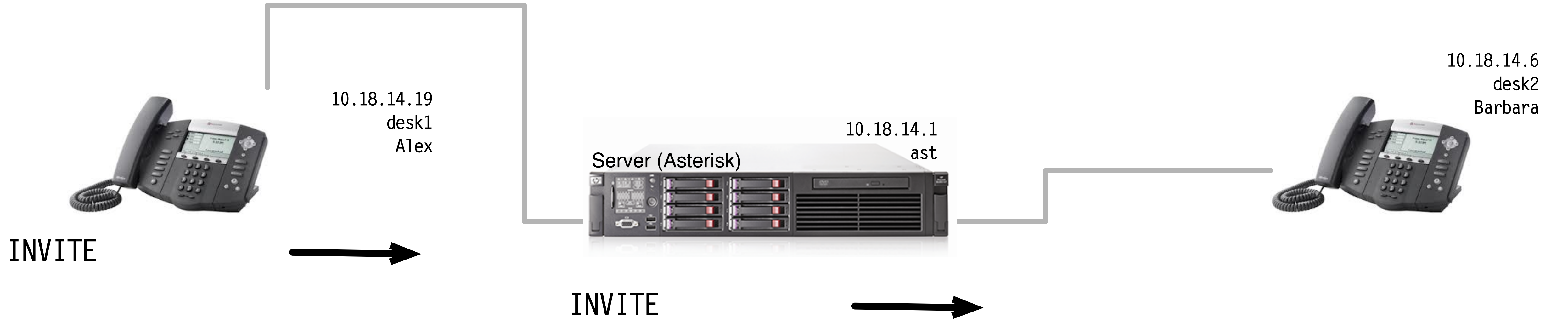

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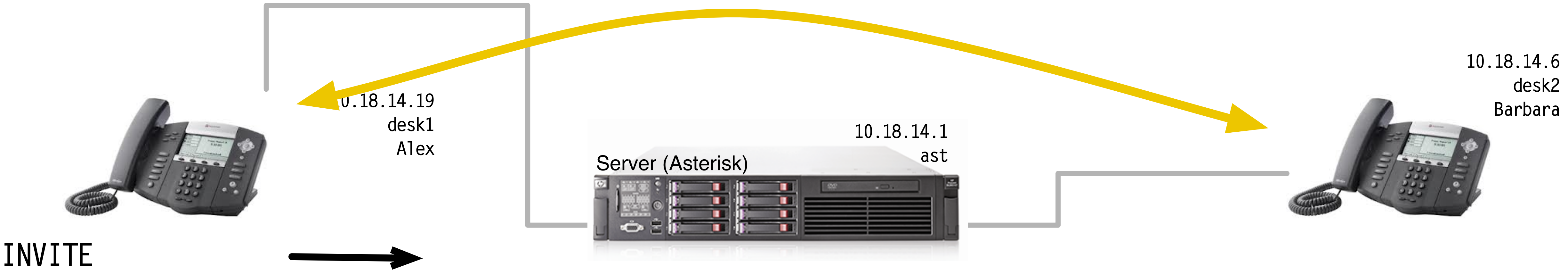

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To: <sip:desk1@10.18.14.1>

SDP

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Media Description: audio *port1* RTP

How Asterisk is organized

Asterisk Versions

Asterisk Versions

2005	**1.2**	DUNDi support; Sound files

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1.2 DUNDi support; Sound files

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1.4 T.38 IP fax; Jabber; SNMP; "Whisper Paging"

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2011 *Dec*

11 Web Sockets SIP;
ICE, STUN, TURN;
WebRTC; Motif

LTS Release

2012 *Oct*

Modules and Functions

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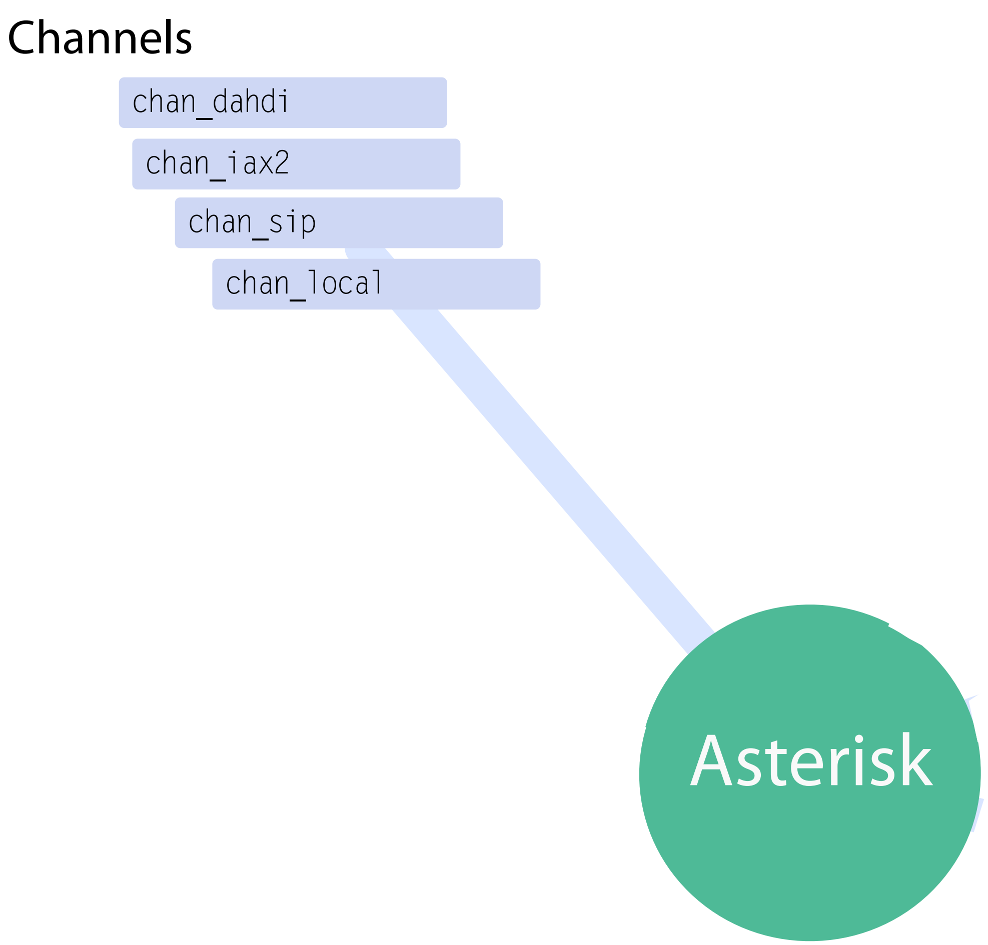
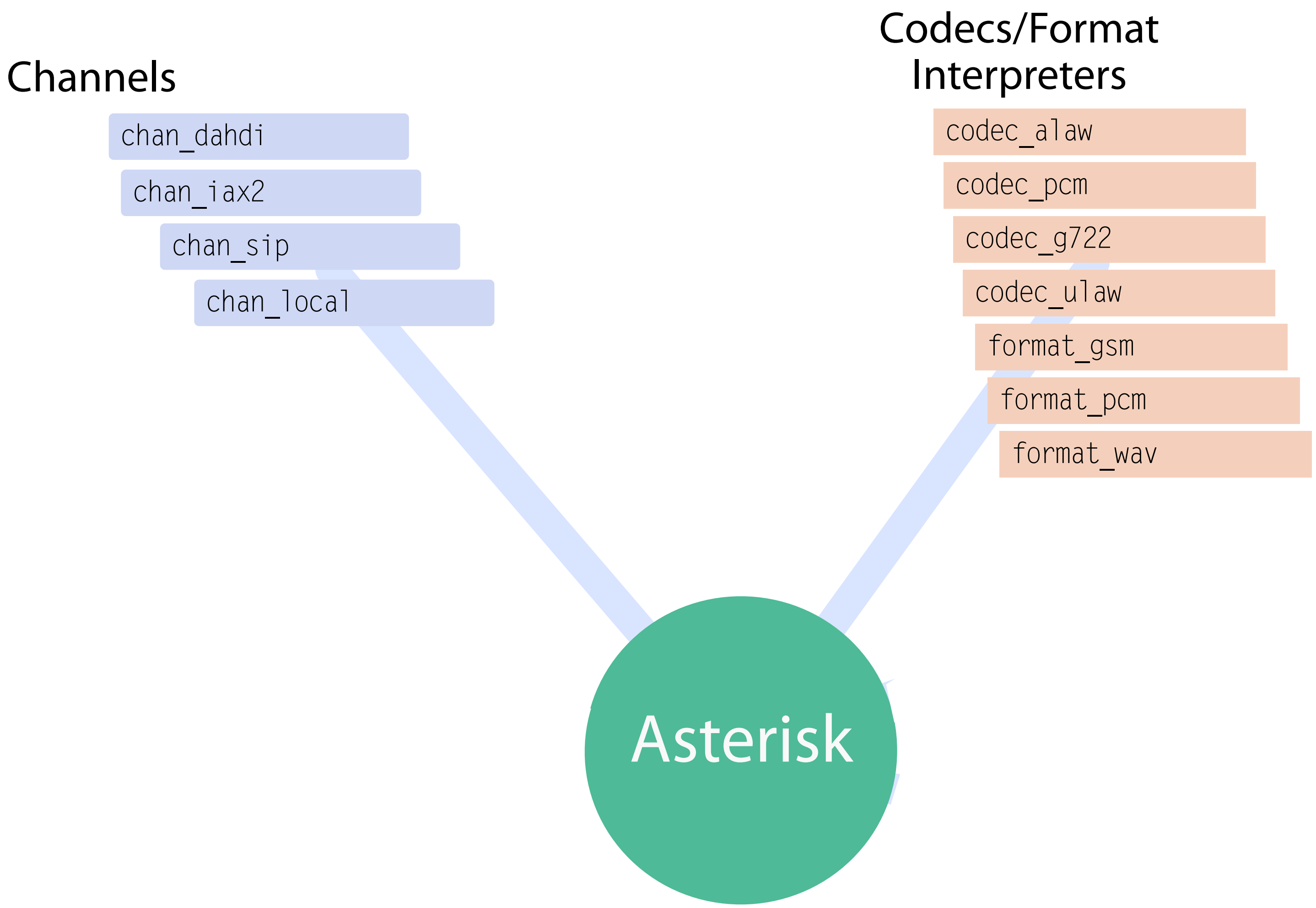


Modules and Functions



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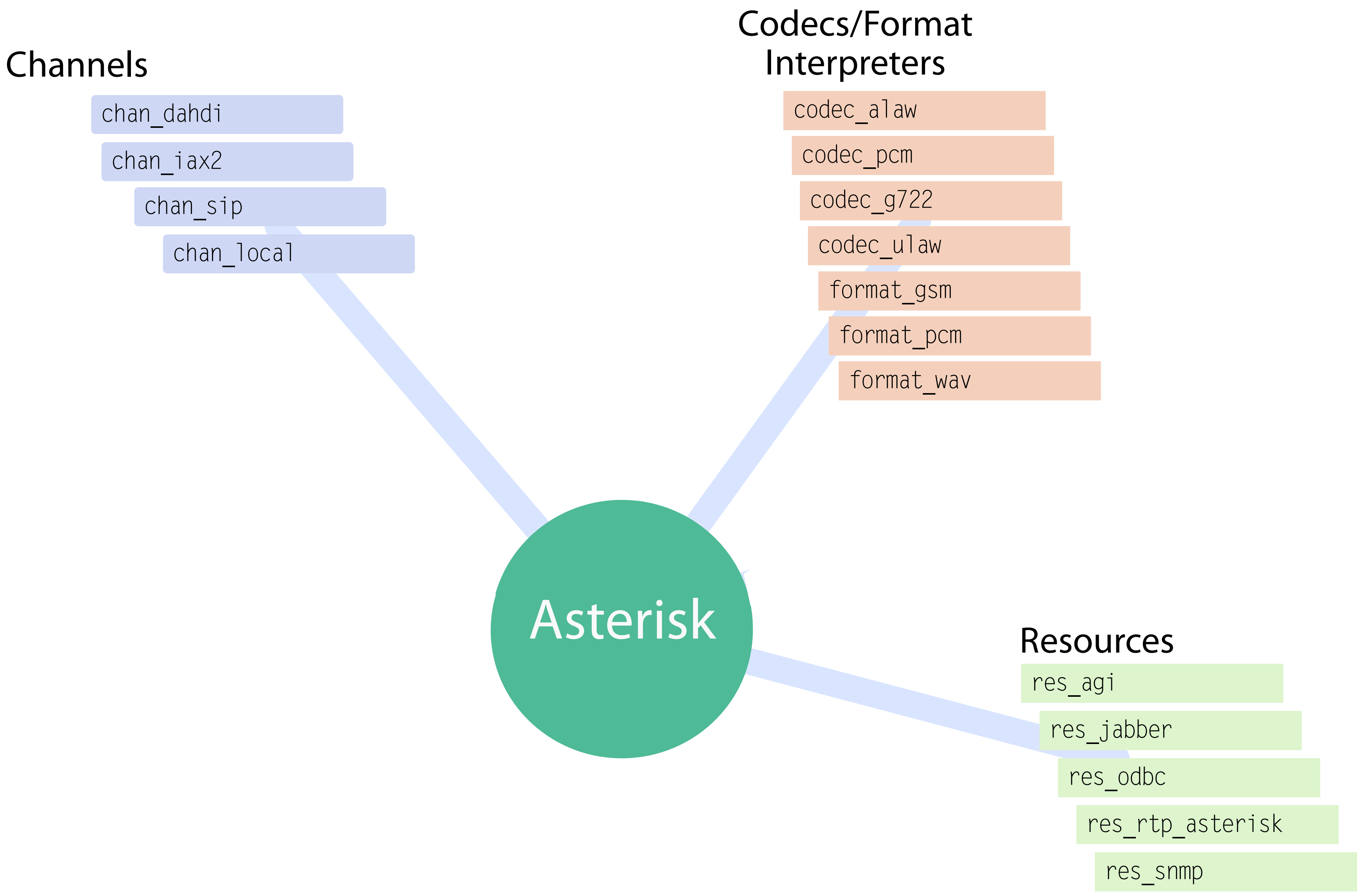

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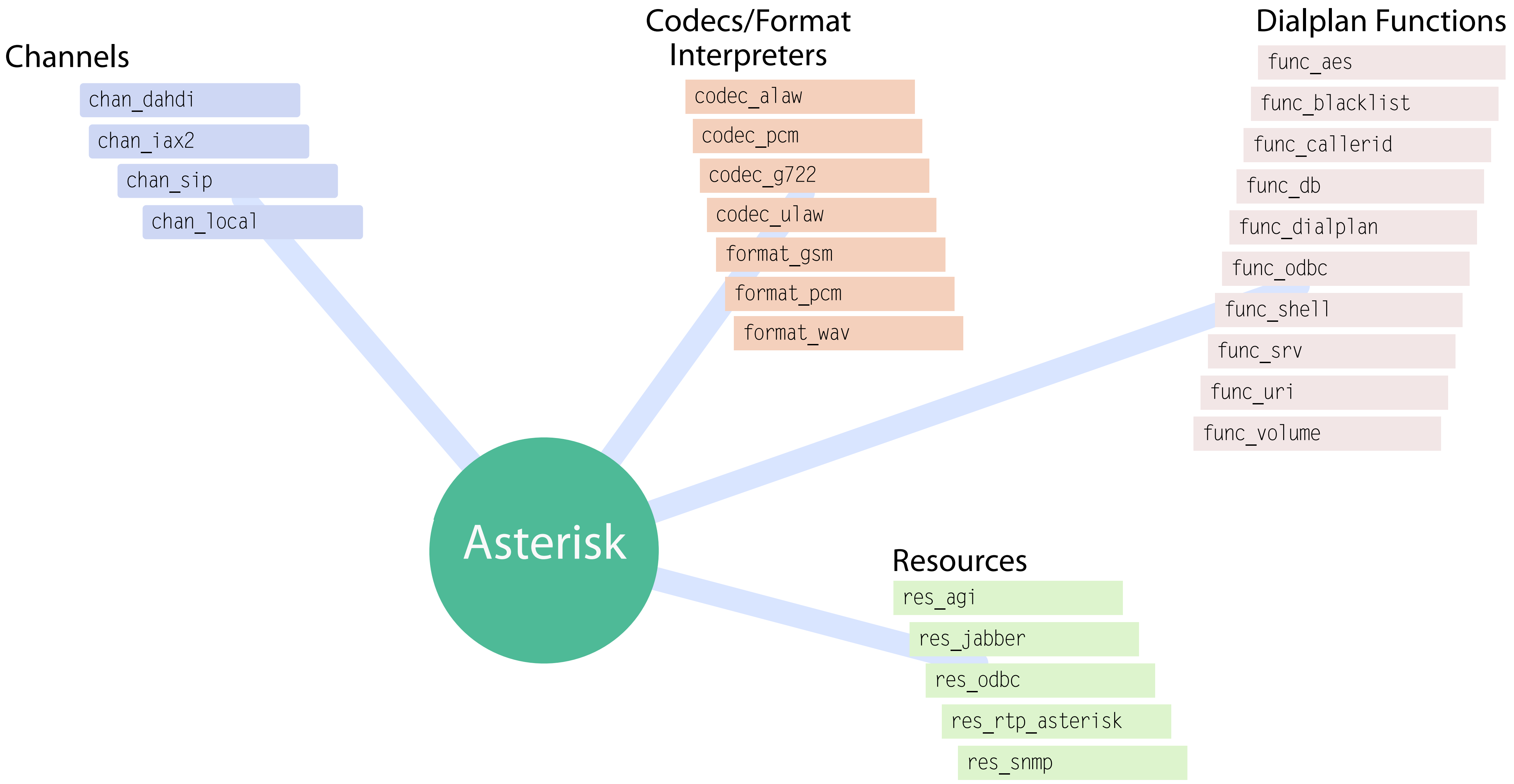

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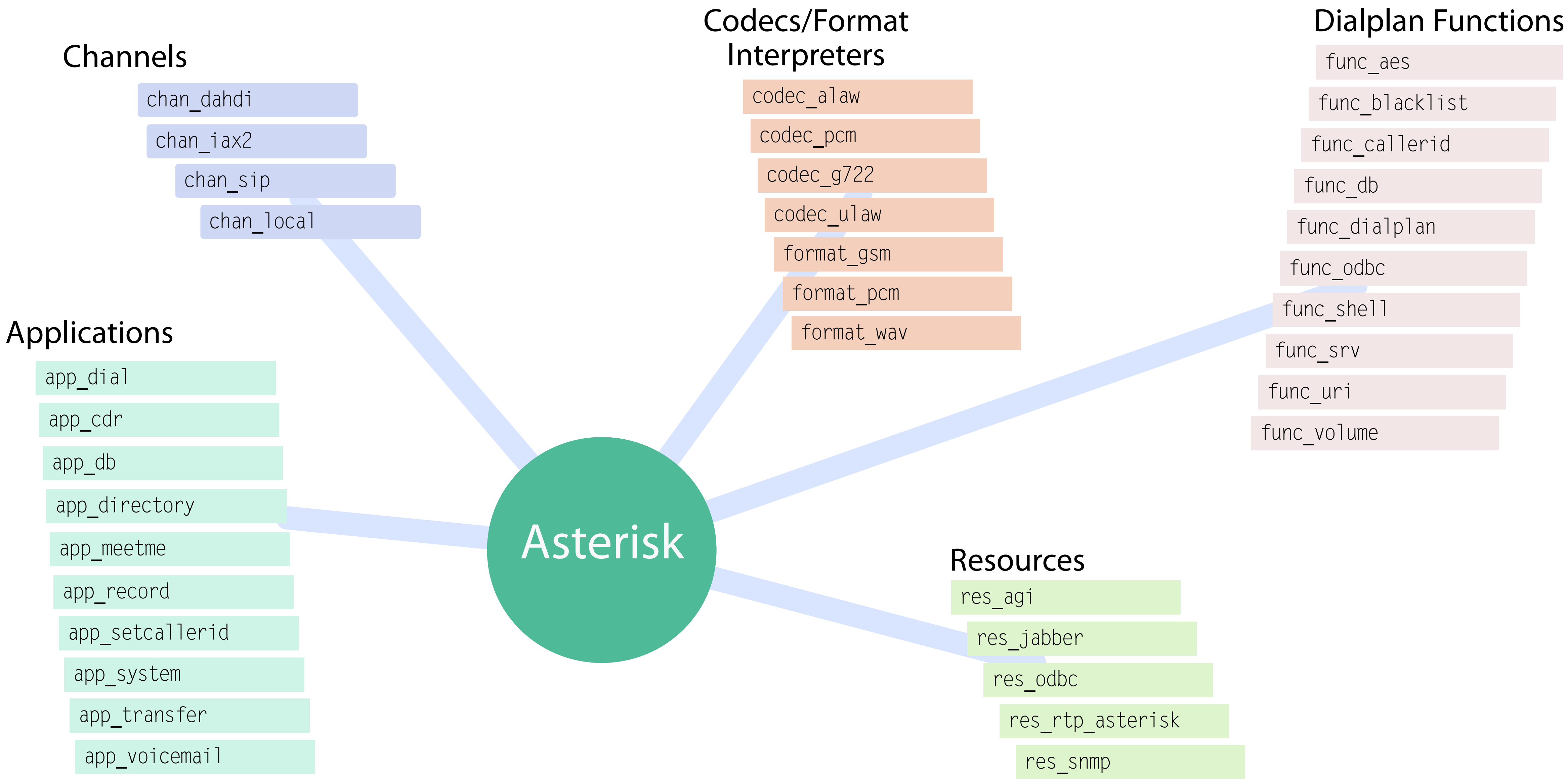

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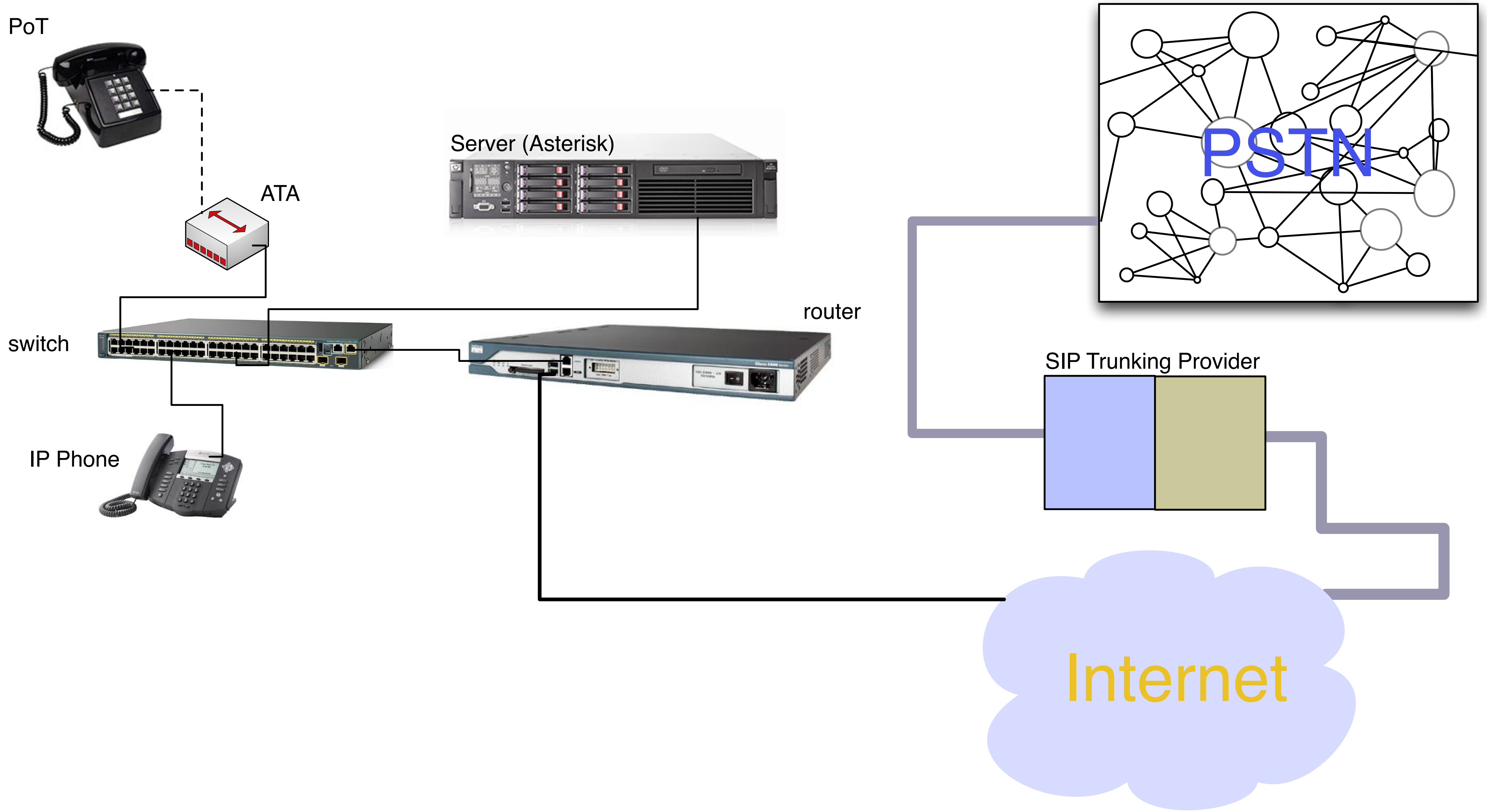

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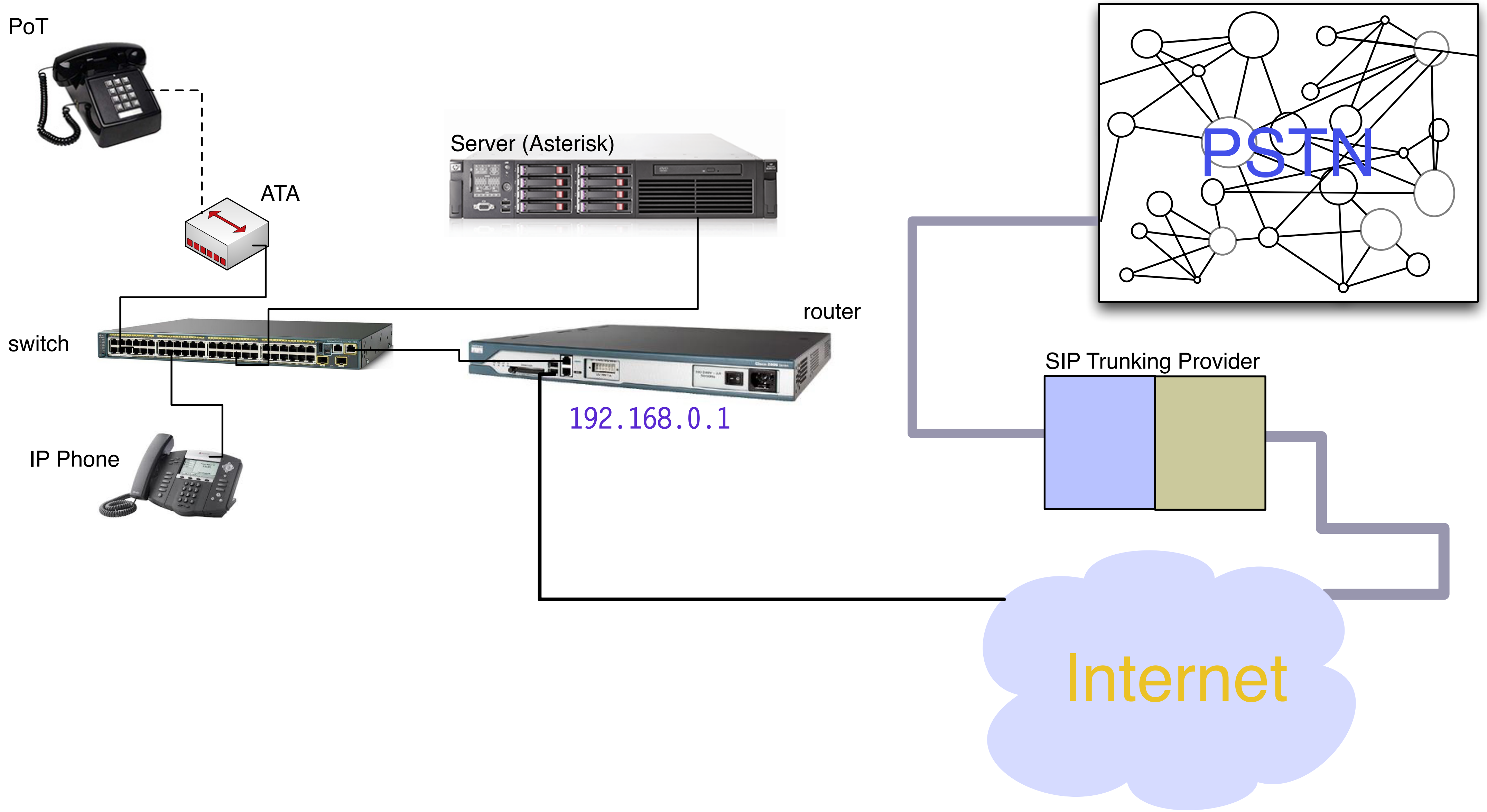

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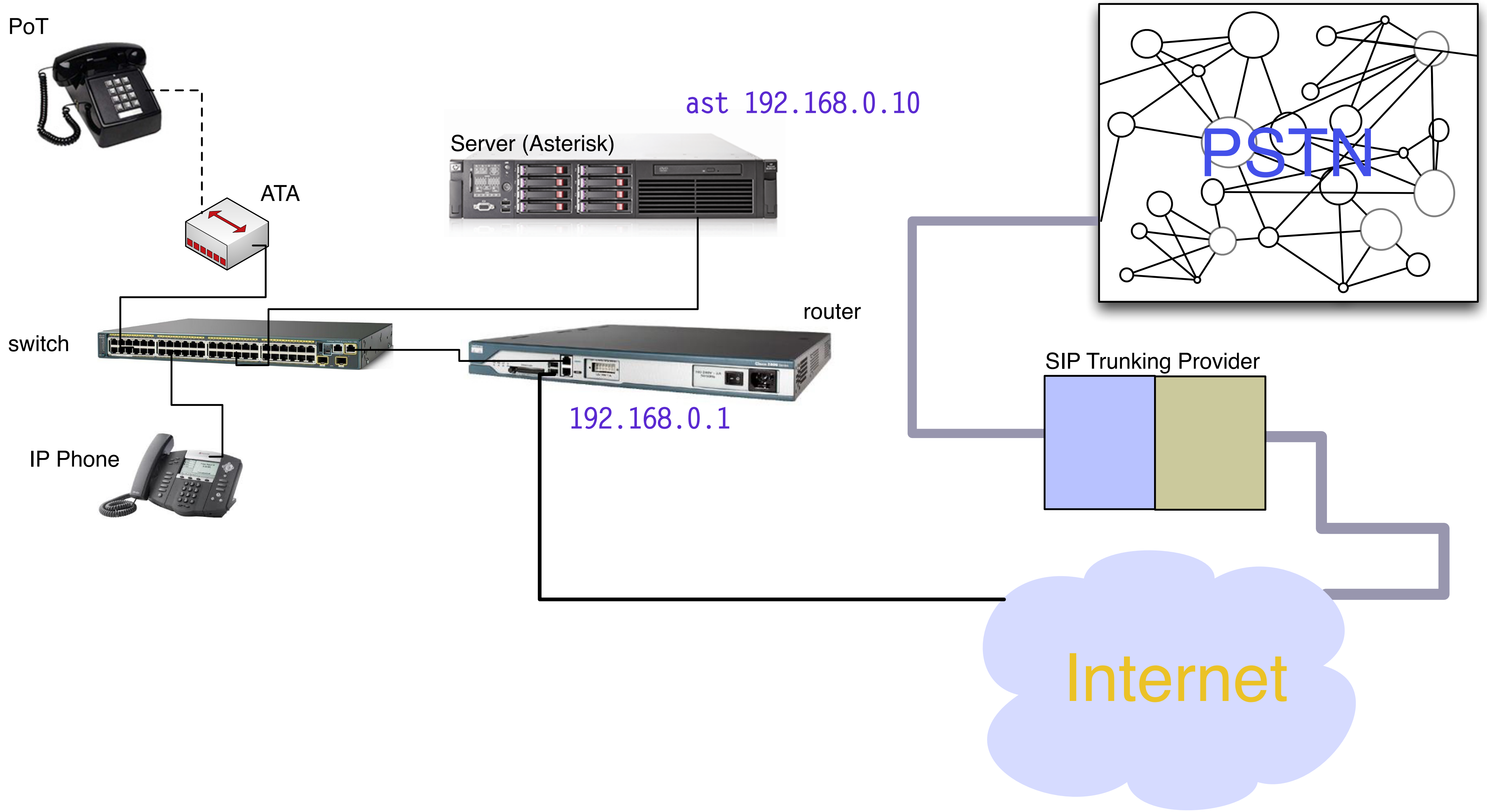

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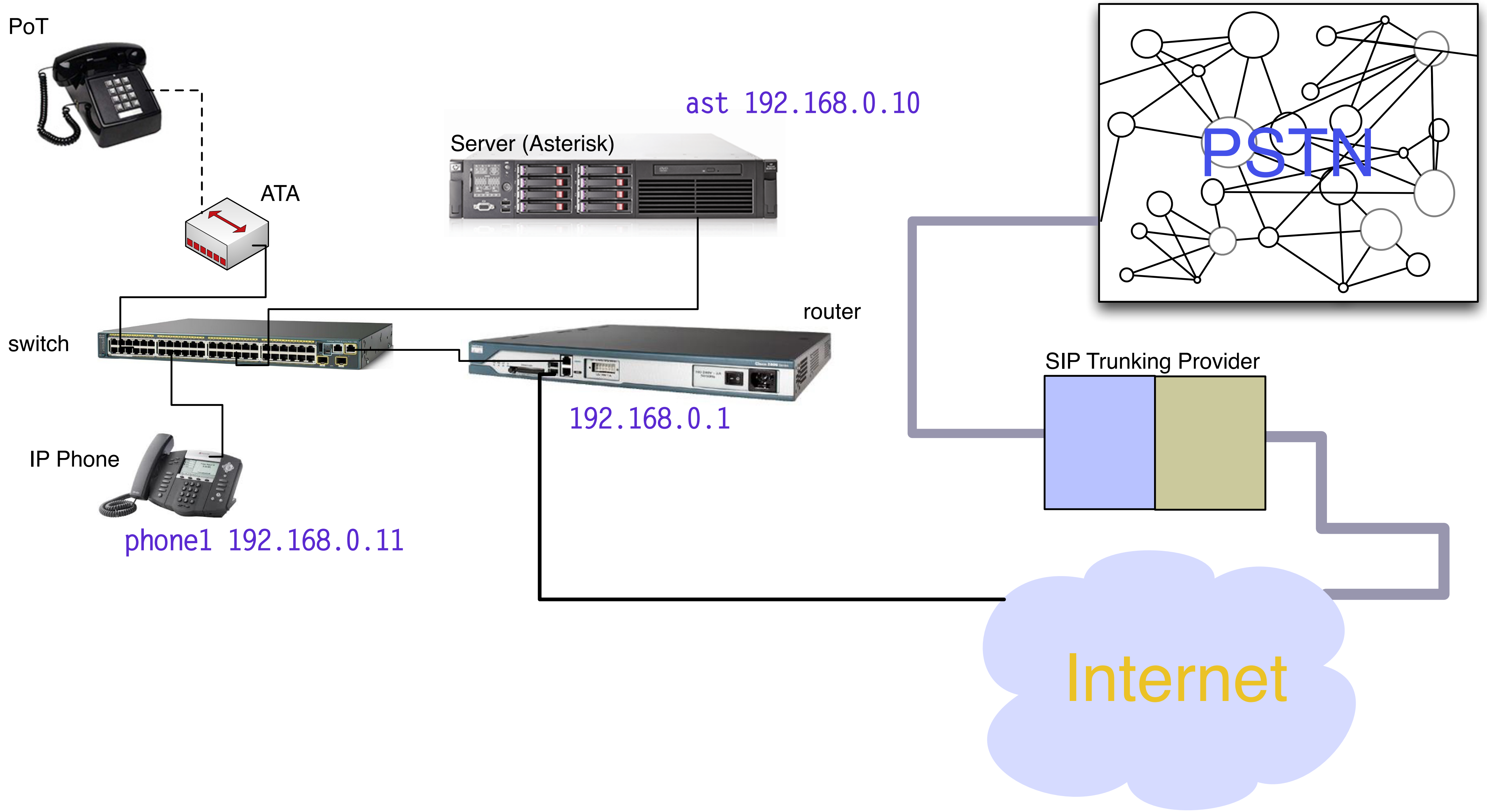

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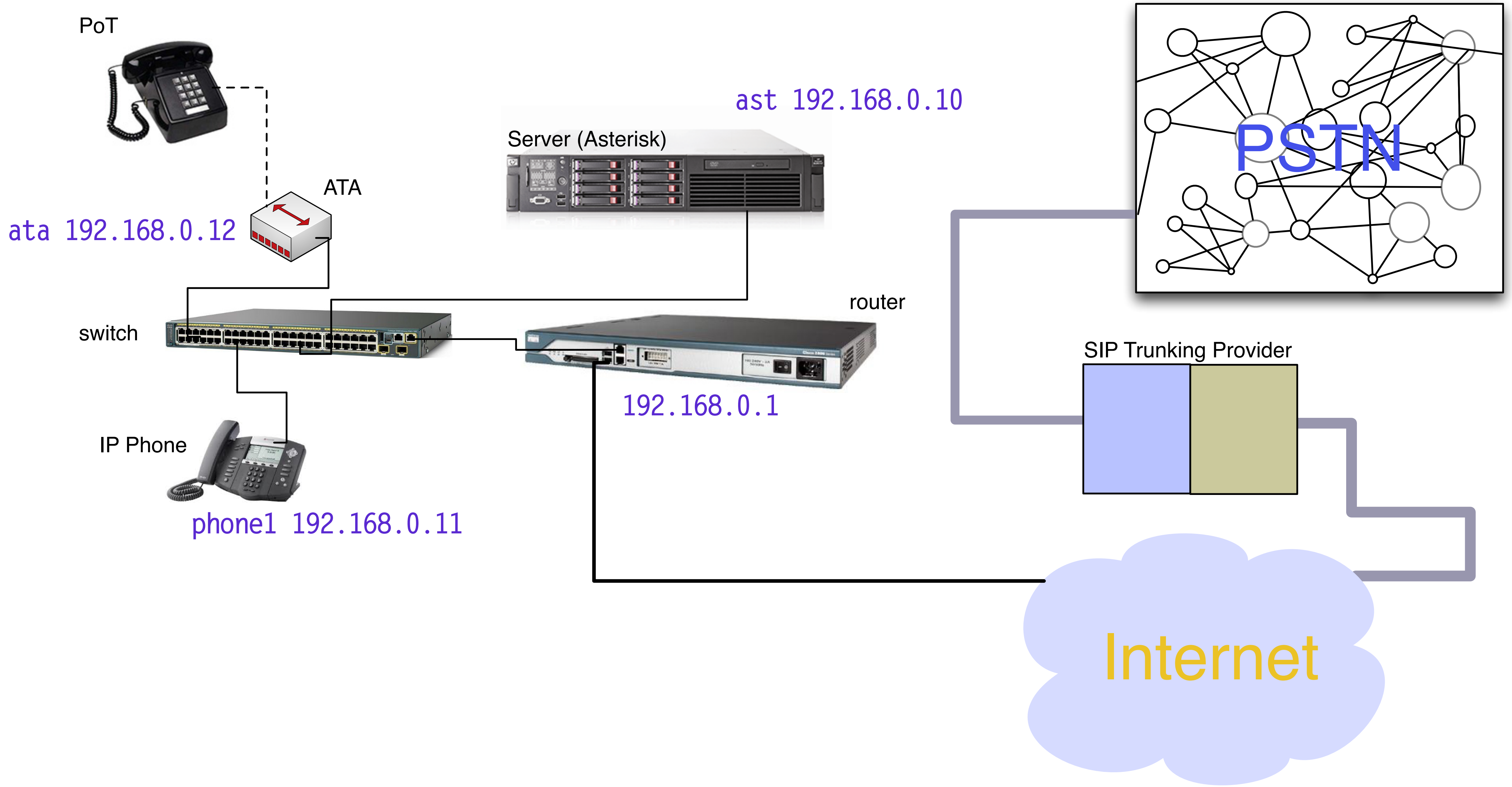

Server Configuration Overview

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Configuration (Files)

Normal installation directories

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Normal installation directories

/etc/asterisk

`asterisk.conf` base config
`voicemail.conf` voicemail
`rtp.conf` RTP configuration
`extensions.conf` Asterisk dialplan
`sip.conf` SIP/device configuration
...

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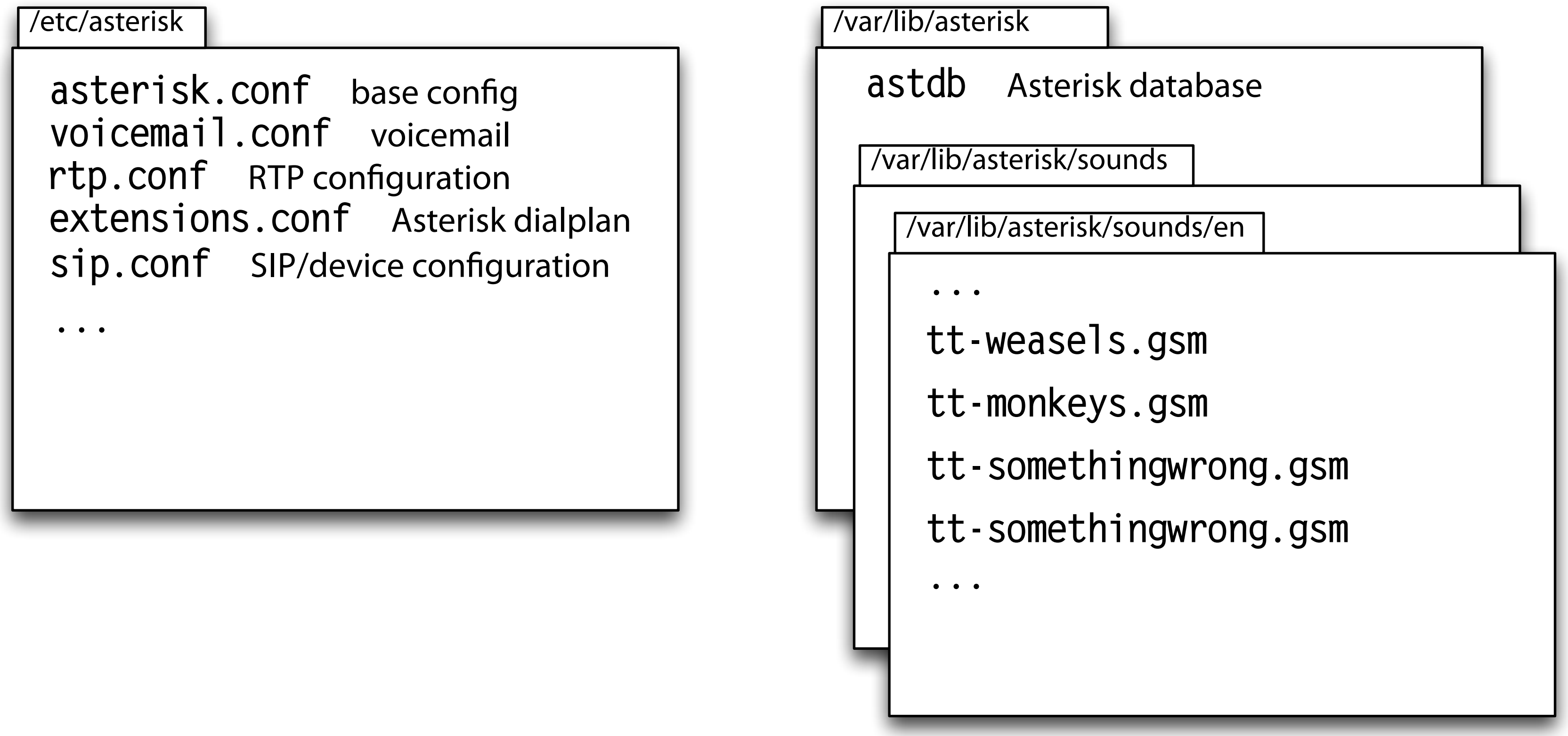

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Normal installation directories

/etc/asterisk

- `asterisk.conf` base config
- `voicemail.conf` voicemail
- `rtp.conf` RTP configuration
- `extensions.conf` Asterisk dialplan
- `sip.conf` SIP/device configuration
- `...`

/var/lib/asterisk

- `astdb` Asterisk database

/var/lib/asterisk/sounds

/var/lib/asterisk/sounds/en

- `...`
- `tt-weasels.gsm`
- `tt-monkeys.gsm`
- `tt-somethingwrong.gsm`
- `tt-somethingwrong.gsm`
- `...`

/var/spool/asterisk

- `dictate/`
- `meetme/`
- `monitor/`

/var/spool/asterisk/voicemail

/var/spool/asterisk/voicemail/default

/var/spool/asterisk/voicemail/default/*extn*

/var/spool/asterisk/voicemail/default/*extn*/en

- `unavail.gsm` unavailable message
- `busy.gsm` busy message

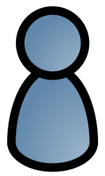
/var/spool/asterisk/voicemail/default/*extn*/INBOX

- `msg0000.wav` (first) message sound
- `msg0000.txt` (first) message metadata
- `...`

Subscribers

Subscribers

Subscriber

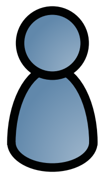


- voicemail

- primary phone (registration)

Subscribers

Subscriber



- voicemail

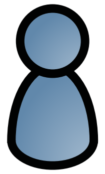
- primary phone (registration)

/etc/asterisk/voicemail.conf

```
…
[voiceMail]
201=>1234,K M Peterson,,,
202=>4567,K M Peterson Test
```

Subscribers

Subscriber



- voicemail

- primary phone (registration)

/etc/asterisk/voicemail.conf

```
…
[voiceMail]
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```
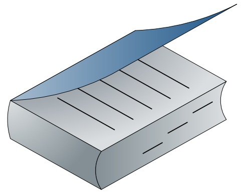
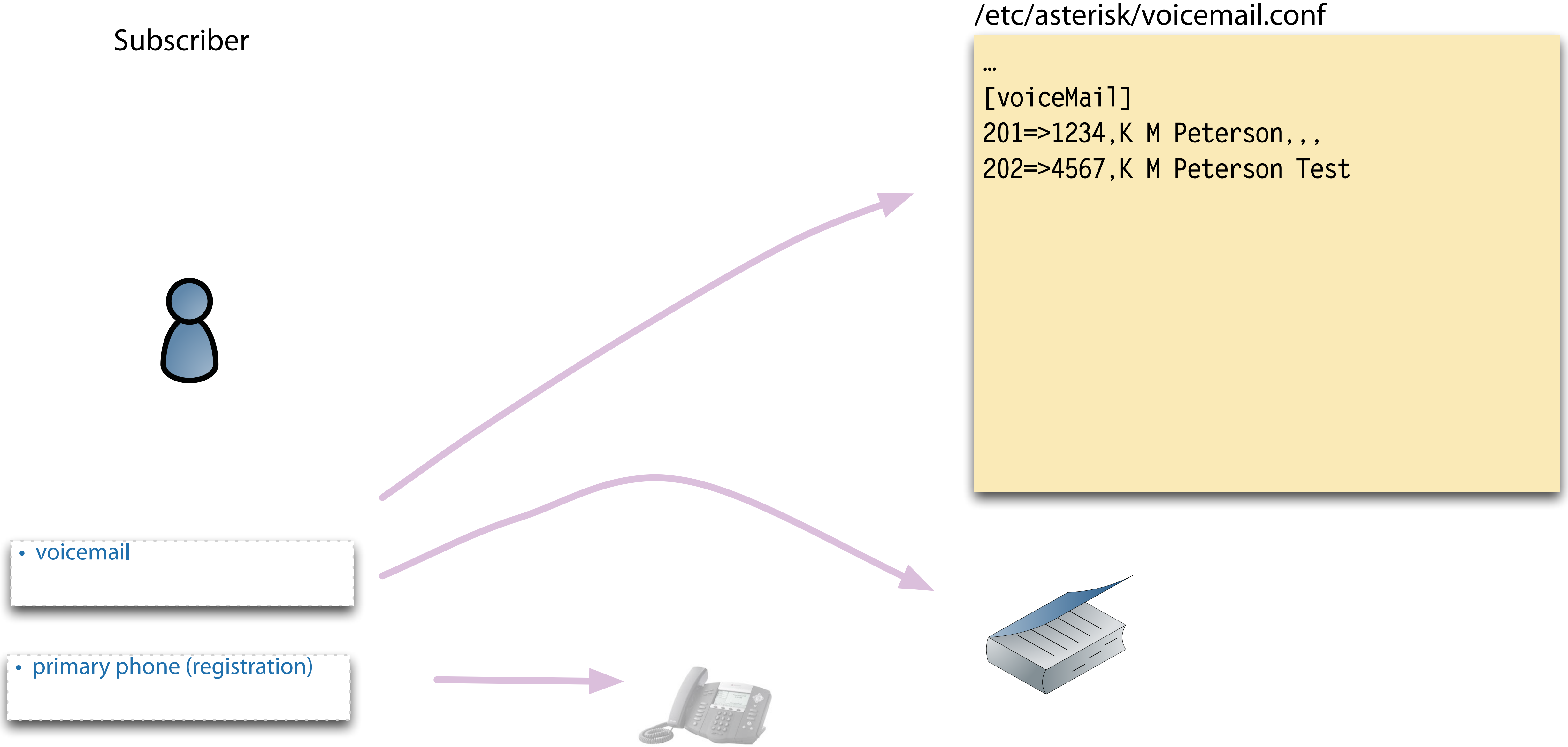



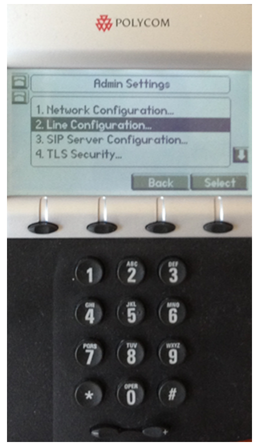
Subscribers



Provisioning

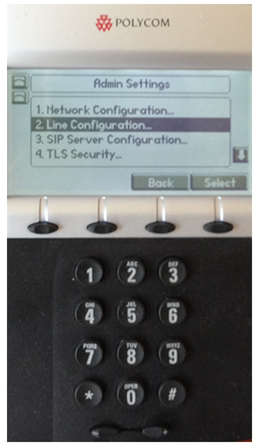
Provisioning

Provisioning



Phone
UI

Provisioning



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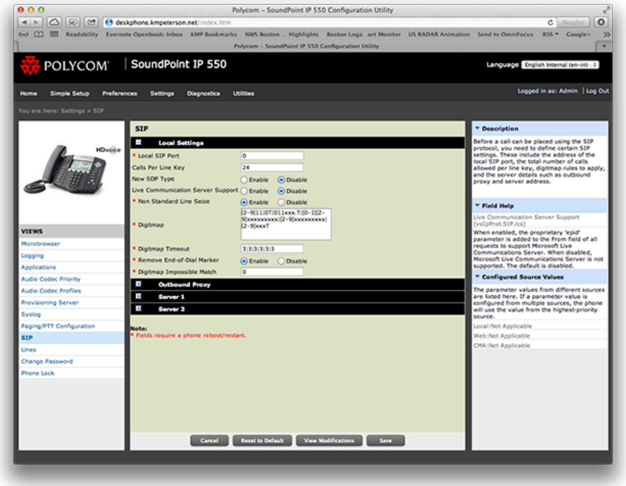
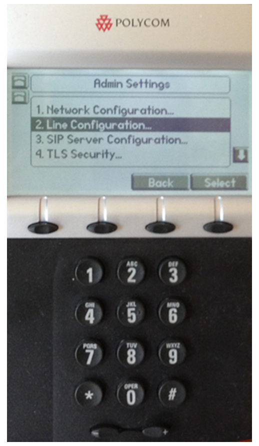
UI

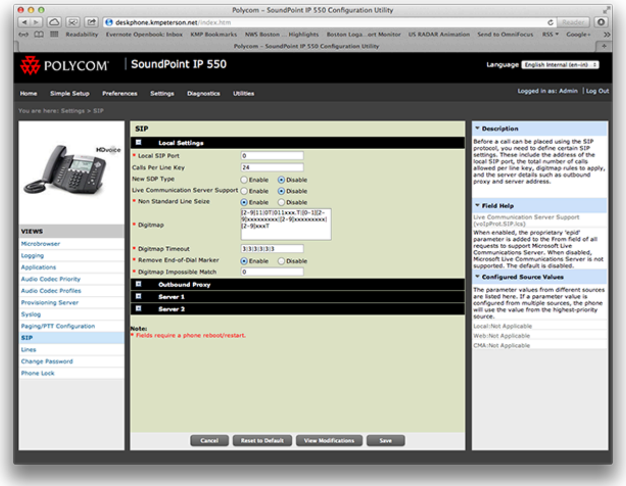


Phone
Internal Web Server

Provisioning



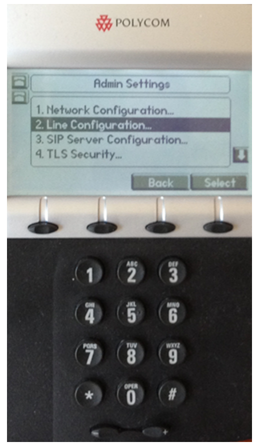
Phone
UI



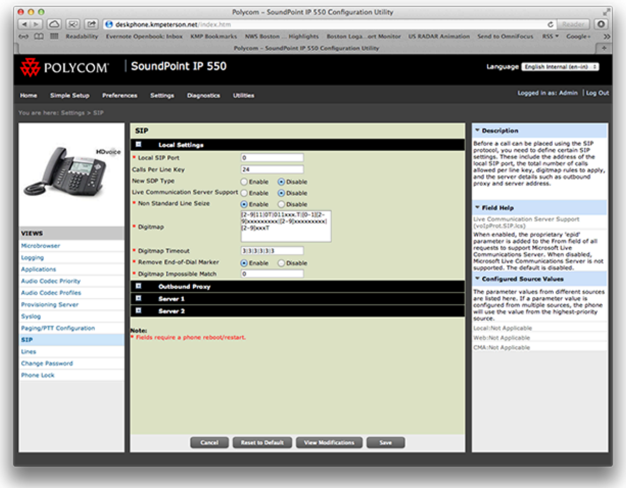
Phone
Internal Web Server

Server

Provisioning



Phone
UI



Phone
Internal Web Server

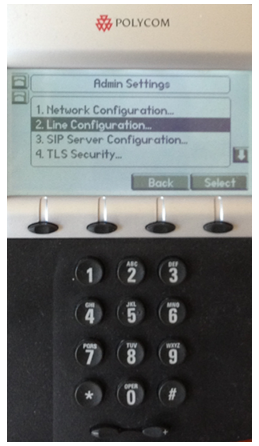
Server-based Provisioning
Via FTP, HTTP, HTTPS

Server

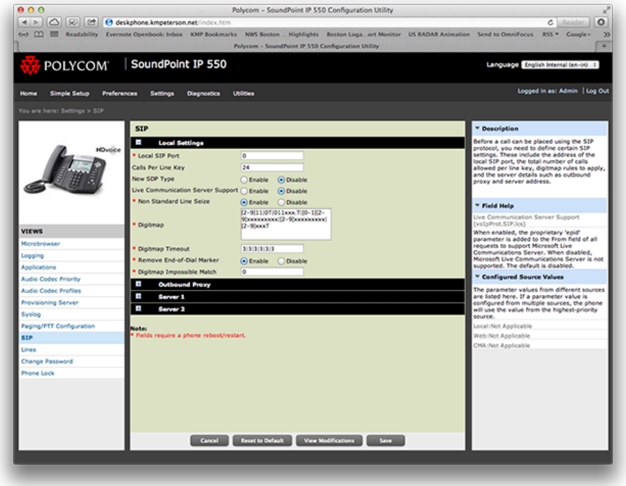
/etc/plcm

`000000000000.cfg`	boot config
`0004f21ef43c.cfg`	overrides
`default1.cfg`	base configuration

Provisioning



Phone
UI



Phone
Internal Web Server

Server-based Provisioning
Via FTP, HTTP, HTTPS

Configurability
- Sounds
- Softkeys
- Microbrowser
- Indicators ...

Server

/etc/plcm

`000000000000.cfg`	boot config
`0004f21ef43c.cfg`	overrides
`default1.cfg`	base configuration

Server Provisioning - SIP

/etc/asterisk/sip.conf

```
...

[phone1]
type=friend
host=dynamic
secret=v6hVzxN7a
context=localHome
permit=192.168.0.0/255.255.255.0
progressinband=yes
dtmfmode=inband
mailbox=201@voiceMail
directmedia=no
callerid="Phone 1" <201>
qualify=50
```

Server Provisioning - SIP

/etc/asterisk/sip.conf

```
...

[phone1]
type=friend
host=dynamic
secret=v6hVzxN7a
context=localHome
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dtmfmode=inband
mailbox=201@voiceMail
directmedia=no
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```

SIP configuration for IP Phone

Server Provisioning - SIP

/etc/asterisk/sip.conf

```
...
[phone1]
type=friend
host=dynamic
secret=v6hVzxN7a
context=localHome
permit=192.168.0.0/255.255.255.0
progressinband=yes
dtmfmode=inband
mailbox=201@voiceMail
directmedia=no
callerid="Phone 1" <201>
qualify=50
```

SIP configuration for IP Phone

`friend` = user and peer (send and receive calls)

`secret`: authentication secret for registration

`context`: dialplan section to process calls

`progressinband`: receive audio from server with progress tones

`mailbox`: voicemail extension and context

`directmedia`: Do not want server to reinvite itself out of media stream

`qualify`: ms timeout to assume phone may not accept calls

Contexts and Extensions

Sections and addresses in dialplan

Server Provisioning - SIP

/etc/asterisk/sip.conf

```
...
[ata]
type=friend
host=dynamic
secret=zNttQR0l
context=localHome
permit=192.168.0.0/255.255.255.0
progressinband=yes
dtmfmode=inband
mailbox=202@voiceMail
directmedia=no
callerid="ATA Phones" <202>
qualify=50
```

Server Provisioning - SIP

/etc/asterisk/sip.conf

```
...

[ata]
type=friend
host=dynamic
secret=zNttQR0l
context=localHome
permit=192.168.0.0/255.255.255.0
progressinband=yes
dtmfmode=inband
mailbox=202@voiceMail
directmedia=no
callerid="ATA Phones" <202>
qualify=50
```

SIP configuration for IP Phone

Server Provisioning - SIP

/etc/asterisk/sip.conf

```
...
[ata]
type=friend
host=dynamic
secret=zNttQR0l
context=localHome
permit=192.168.0.0/255.255.255.0
progressinband=yes
dtmfmode=inband
mailbox=202@voiceMail
directmedia=no
callerid="ATA Phones" <202>
qualify=50
```

SIP configuration for IP Phone

`friend` = user and peer (send and receive calls)

`context`: dialplan section to process calls

`progressinband`: receive audio from server with progress tones

`mailbox`: voicemail extension and context

`directmedia`: Do not want server to reinvite itself out of media stream

`qualify`: ms timeout to assume phone may not accept calls

Server Provisioning - SIP

/etc/asterisk/sip.conf

...

```
[general]
allowguest=no
register=>kmpsipbos:tRQoNl1520@primary.siprovider.com
…
[trunkingProvider]
type=peer
context=incomingTrunk
host=primary.siprovider.com
username=kmpsipbos
secret=tRQoNl1520
qualify=yes
allow=all
directmedia=yes
insecure=port,invite
trustrpid=yes
```

Server Provisioning - SIP

/etc/asterisk/sip.conf

...

```
[general]
allowguest=no
register=>kmpsipbos:tRQoNl1520@primary.siprovider.com
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type=peer
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secret=tRQoNl1520
qualify=yes
allow=all
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insecure=port,invite
trustrpid=yes
```

Configuration for SIP provider

Server Provisioning - SIP

/etc/asterisk/sip.conf

```
                                                  ...
[general]
allowguest=no
register=>kmpsipbos:tRQoNl1520@primary.siprovider.com
…
[trunkingProvider]
type=peer
context=incomingTrunk
host=primary.siprovider.com
username=kmpsipbos
secret=tRQoNl1520
qualify=yes
allow=all
directmedia=yes
insecure=port,invite
trustrpid=yes
```

Configuration for SIP provider

`allowguest`: accept incoming unauthenticated calls?

`register`: command to cause Asterisk server to register to SIP provider

`type` = `peer` (means this definition enables outgoing calls)

`context`: incoming calls pass to this dialplan section

`host`: predefined provider IP address

The Dialplan

The Dialplan

Asterisk "Programming"

(Can be replaced by other features)

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Asterisk "Programming"

(Can be replaced by other features)

Not the easiest syntax to deal with.

Comments follow semicolon `;`

Context names like configuration file sections: `[context]`

Lines begin with `exten=>`*extension*,*priority*,...

Continue in extension next line with `same=>n`,...

Extensions may be numeric, or alphanumeric.

The Dialplan

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(Can be replaced by other features)

Not the easiest syntax to deal with.

Comments follow semicolon `;`

Context names like configuration file sections: `[context]`

Lines begin with `exten=>`*extension*,*priority*,...

Continue in extension next line with `same=>n`,...

Extensions may be numeric, or alphanumeric.

- *Careful with call flow through contexts*

The Dialplan: Internal Extensions

/etc/asterisk/extensions.conf

```
...

[localPlan] ;internal local extensions

exten=>201,1,Dial(SIP/phone1,20)
same=>n,VoiceMail(201@voiceMail,u)

exten=>202,1,Dial(SIP/ata,20)
same=>n,VoiceMail(202@voiceMail,u)

exten=>500,1,answer(500)
same=>n,VoiceMailMain(@voiceMail)

; Misc and Demos

exten=>811,1,Answer(500)
exten=>811,n,Playback(tt-monkeys)
same=>n,Hangup()
```

The Dialplan: Internal Extensions

/etc/asterisk/extensions.conf

```
...

[localPlan] ;internal local extensions

exten=>201,1,Dial(SIP/phone1,20)
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Local extensions that can be dialed by our phones

The Dialplan: Internal Extensions

/etc/asterisk/extensions.conf

```
...

[localPlan] ;internal local extensions

exten=>201,1,Dial(SIP/phone1,20)
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; Misc and Demos

exten=>811,1,Answer(500)
exten=>811,n,Playback(tt-monkeys)
same=>n,Hangup()
```

Local extensions that can be dialed by our phones

x201 - dials the IP Phone
x202 - dials the ATA
x500 - dials Voicemail
x811 - dials the Monkeys

two voicemail boxes defined

Answer: parameter is ms to pause

Hangup: "click".

The Dialplan: Dialing out through SIP provider

/etc/asterisk/extensions.conf

```
...

[localDialOut] ; External dialing

exten=>_NXXNXXXXXX,1,Log(NOTICE,"Internal extension dialed out ${EXTEN}")
same=>n,Set(CALLERID(num)=6175551212)
same=>n,Set(CALLERID(name)="K M PETERSON")
same=>n,Dial(SIP/1${EXTEN}@trunkingProvider,75,r)

[speedDial] ; Speed Dial Codes
exten=>3,1,Set(CALLERID(num)=6175551212)
same=>n,Set(CALLERID(name)="KMP Home")
same=>n,Dial(SIP/16179361234@trunkingProvider,75,r)
```

The Dialplan: Dialing out through SIP provider

/etc/asterisk/extensions.conf

```
. . .

[localDialOut] ; External dialing

exten=>_NXXNXXXXXX,1,Log(NOTICE,"Internal extension dialed out ${EXTEN}")
same=>n,Set(CALLERID(num)=6175551212)
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```

Dialing out, in the US

The Dialplan: Dialing out through SIP provider

/etc/asterisk/extensions.conf

```
...

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exten=>_NXXNXXXXXX,1,Log(NOTICE,"Internal extension dialed out ${EXTEN}")
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same=>n,Set(CALLERID(name)="KMP Home")
same=>n,Dial(SIP/16179361234@trunkingProvider,75,r)
```

Dialing out, in the US

_NXXNXXXXXX - dial string to match
leading underscore matches start of digits received
N=[2..9], X=[0..9]

We must set CID ourselves

Dial takes provided extension EXTEN and sends to our provider.

`[speedDial]` context intercepts single entered digit (3), then sends predetermined number through SIP provider.

The Dialplan: Starting: Local and Outbound

/etc/asterisk/extensions.conf

```
...

[localHome]
include=>speedDial
include=>localPlan
include=>localDialOut

[incomingTrunk]

exten=>16175551212,1,Log(NOTICE,"Incoming, Id=${CALLERID(num)}/${SIPCALLID}")
exten=>16175551212,2,Goto(incomingTrunk,start,1)

; In case call came in without our number, log it…
same=>n,Log(NOTICE,"Incoming call, Src=Uncaught, Id=${CALLERID(num)}")
same=>n,Hangup()

exten=>start,1,Goto(publicIncoming,start,1)
```

The Dialplan: Starting: Local and Outbound

/etc/asterisk/extensions.conf

```
. . .

[localHome]
include=>speedDial
include=>localPlan
include=>localDialOut

[incomingTrunk]

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Target contexts for our local phones and incoming

The Dialplan: Starting: Local and Outbound

/etc/asterisk/extensions.conf

```
...

[localHome]
include=>speedDial
include=>localPlan
include=>localDialOut

[incomingTrunk]

exten=>16175551212,1,Log(NOTICE,"Incoming, Id=${CALLERID(num)}/${SIPCALLID}")
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same=>n,Hangup()

exten=>start,1,Goto(publicIncoming,start,1)
```

Target contexts for our local phones and incoming

The `localHome` context has these others included in it.

The `incomingTrunk` context receives calls from our SIP provider. The extension that we receive to match on is the number that was called - it should be ours.

If we receive an extension in this context we don't recognize, it's an error and we hang up on it.

The Dialplan: Incoming Calls Menu

/etc/asterisk/extensions.conf

```
...

[publicIncoming]
; Main menu for public incoming calls
exten=>start,1,Background(local/welcome)
same=>n,Log(NOTICE,"Playing welcome message.")

exten=>retry,1,Background(local/msg-genMenu)
same=>n,WaitExten(5)
exten=>1,1,Goto(ringDeskPublic,start,1)
exten=>8,1,VoiceMail(201@voiceMail,u)
exten=>0,1,Goto(youHaveTheWrongGuy,start,1)

exten=>i,1,Log(NOTICE,"Invalid response to main menu.")
exten=>t,1,Log(NOTICE,"Timeout in main menu.")
same=>n,Playback(invalid)
same=>n,Hangup()
```

The Dialplan: Incoming Calls Menu

/etc/asterisk/extensions.conf

```
. . .
[publicIncoming]
; Main menu for public incoming calls
exten=>start,1,Background(local/welcome)
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Menu our incoming calls go into

The Dialplan: Incoming Calls Menu

/etc/asterisk/extensions.conf

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exten=>i,1,Log(NOTICE,"Invalid response to main menu,")
exten=>t,1,Log(NOTICE,"Timeout in main menu,")
same=>n,Playback(invalid)
same=>n,Hangup()
```

Menu our incoming calls go into

Play the initial welcome message

Play the general menu message. Wait 5 seconds for a response.

1 is for "complete your call and ring the phone.
8 is for "leave a message".
0 is if you're not sure of who you've reached.

i is a code for invalid response provided.
t is a code indicating no response received.

The Dialplan: Ringing the phone

/etc/asterisk/extensions.conf

```
...

[ringDeskPublic]
; Calls from public go here to ring desk/house phones
exten=>start,1,Dial(SIP/phone1&SIP/ata,30)
same=>n,Background(local/msg-voiceMail)
same=>n,WaitExten(5)

exten=>8,1,VoiceMail(201@voiceMail)
exten=>i,1,VoiceMail(201@voiceMail)
exten=>t,1,VoiceMail(201@voiceMail)
```

The Dialplan: Ringing the phone

/etc/asterisk/extensions.conf

```
...

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; Calls from public go here to ring desk/house phones
exten=>start,1,Dial(SIP/phone1&SIP/ata,30)
same=>n,Background(local/msg-voiceMail)
same=>n,WaitExten(5)

exten=>8,1,VoiceMail(201@voiceMail)
exten=>i,1,VoiceMail(201@voiceMail)
exten=>t,1,VoiceMail(201@voiceMail)
```

Actually making the phone ring

The Dialplan: Ringing the phone

/etc/asterisk/extensions.conf

```
...

[ringDeskPublic]
; Calls from public go here to ring desk/house phones
exten=>start,1,Dial(SIP/phone1&SIP/ata,30)
same=>n,Background(local/msg-voiceMail)
same=>n,WaitExten(5)

exten=>8,1,VoiceMail(201@voiceMail)
exten=>i,1,VoiceMail(201@voiceMail)
exten=>t,1,VoiceMail(201@voiceMail)
```

Actually making the phone ring

Ring the phones for 30 seconds, and then play the voicemail message.

Play the general menu message. Wait 5 seconds for a response.

All options go to voicemail, but entering 8 will bypass remaining part of the message.

Issues and Lessons

- Operations

  the asterisk console: `/usr/sbin/asterisk -r`

  - **backup services**
  - **Support for E911 from commercial providers**
  - **SELinux**
  - **Network Monitoring**

- Security
  - **beware of misuse**
  - **Consider fail2ban**
  - **Security monitoring (consider volume)**

Issues and Lessons

- Voice Quality
  - **Users don’t understand the Internet**
  - **Monitoring strategies**
  - **Still not sure about the media stream**
- Building Asterisk
  - **Not too bad under CentOS**
  - **Needed additional support for ODBC, SNMP**

Issues and Lessons

- Digression: stumped by this problem:
  1. **Hadn’t really use ATA intensively**
  2. **Found out at some point, couldn’t initiate calls and get audio**
  3. **Spent a lot of time looking at packets on wire, reading up about NAT-related issues and SIP protocol.**
  4. **No serious differences between configurations for ATA and Phone.**
  5. **Tried defining port ranges - but they matched between ATA and Phone, and still no sound.**
  6. **Noticed “host unreachable replies” only for ATA voice traffic; then noticed that Asterisk always talked to Phone earlier in session than it did to ATA.**
  7. **Turned of host firewall - iptables - and sound worked. But why...?**

Issues and Lessons: Projects

- Anniversary project
  - **Set up mailbox for friends of in-laws to send their wishes**
- Softphones
  - **Have several configured. Great way to learn about implementations.**
- Whitelist
  - **Add friends to whitelist to avoid initial menu, also have more options if not answered.**
- Follow Me - Mobile
  - **Implemented option to send call to Mobile.**
  - **Mobile programed to send call back if no answer there.**
  - **Asterisk "remembers" numbers if they get sent back.**

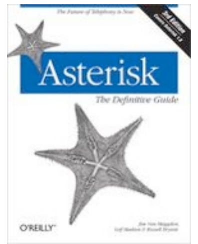
Issues and Lessons: Projects

- Emergency Inbound Calls (“Reverse-911”)
  - **If CID is local emergency number, play short message (“your call is being recorded”), route call to speaker on phone**
  - **Audio copied to sound file; file emailed to me and SMS page sent.**

Asterisk Resources

- Asterisk site: http://www.asterisk.org downloads and info
- Asterisk Wiki: https://wiki.asterisk.org/wiki/display/AST/Home
- VoIP Info: http://www.voip-info.org/
- Hardware: http://www.voipsupply.com
- Books:

Asterisk™: The Definitive Guide
By: Leif Madsen; Jim Van Meggelen; Russell Bryant
Publisher: O'Reilly Media, Inc.
Pub. Date: April 28, 2011 Most Recent Edition
Print ISBN-13: 978-0-596-51734-2



Thanks! Questions?

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http://kmpeterson.com/special/bblisa-asterisk