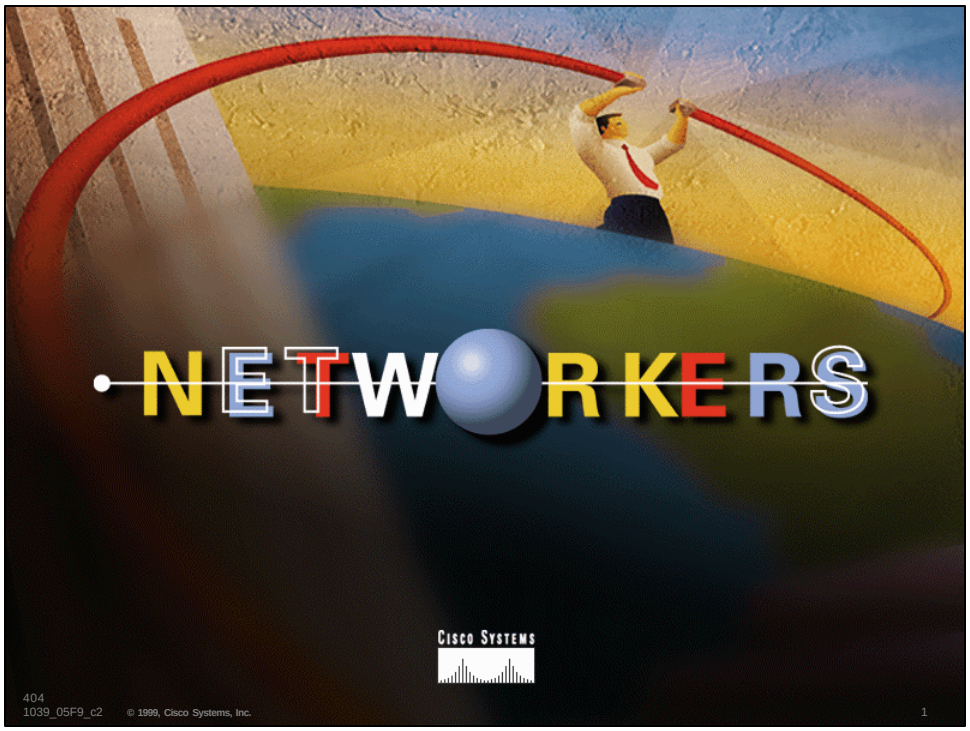

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Understand the Components—Native IP Telephony

- **Phones**
  - "Functional" = full H.323 or SIP, ex: symbol
  - "Stimulus" = Skinny Station or like, ex: Cisco

- **Gateways**
  - PBX and PSTN connectivity

- **Applications and call processing**
  - CallManager, voice mail, IVR, etc.

- **Network infrastructure**
  - Routers, switches, wire, WAN services

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Agenda

- **Prepare**
- **Install**
  - CallManager and IP Phones
  - Gateways
  - Voice Mail and Other Applications
- **Manage and Secure**

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Preparing for Installation

- Realistic expectations
- IP addressing plan
- Power planning
- Dial plan
- Licensing issues
- Allocating bandwidth
- Good network design
- Centralized vs. distributed call processing
- Redundancy
- Selecting an appropriate gateway

Realistic Expectations

**Yank Out My PBX,
This IP Phone System Is
Practically FREE**

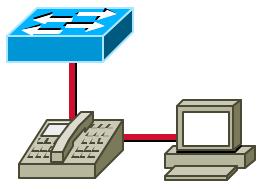
- Nothing comes for free, still need good planning and design
- Beware caveats of CM 2.2

IP Address Plan

- IP phones need addresses too!
  - Configure phones statically or use DHCP
- Address space options:
  - Double current address space
  - Phones on separate subnets
  - Secondary addressing per subnet
- Phones don't work across NAT/PAT/ firewall boundaries today

Share Same Drop— Double Address Space

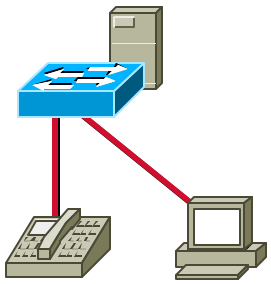
- Phones have 10 BaseT hub, PC limited to 10 Mb
- Future phones have 10/100 switch
- Phone and PC share same VLAN
- If phones lose power, net connection to PC is lost too
- Address statically or use any DHCP server



Separate Subnets

- Separate desktop jacks
- PCs can connect at 100 Mb
- Address statically or use any DHCP server
- Still get easy moves/adds/changes with DHCP

Optional DHCP



10.21.45.12 197.21.45.12

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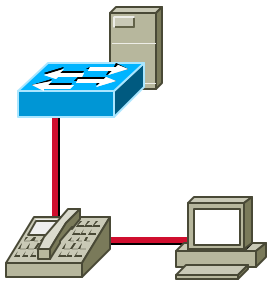
Share Drop—Split Addressing per Port

- VLAN router uses secondary addressing
- Assign static addresses

or

- Use Cisco Network Registrar DHCP to assign addresses from different pools

Cisco Network Registrar



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Script tags phones by MAC header

See talk # 806 for more info

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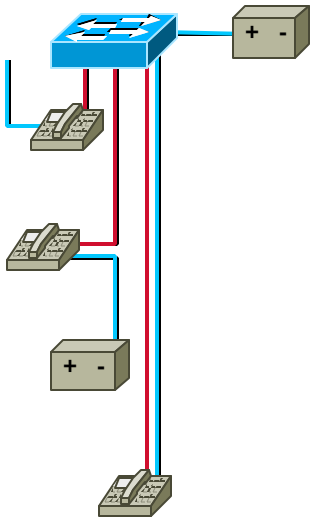
Power Planning

- Local power
  - Optional local UPS for backup
- Power on pins seven and eight from wiring closet
  - For backup use—48 V batteries in wiring closet
- Power from the switch (future)



Licenses

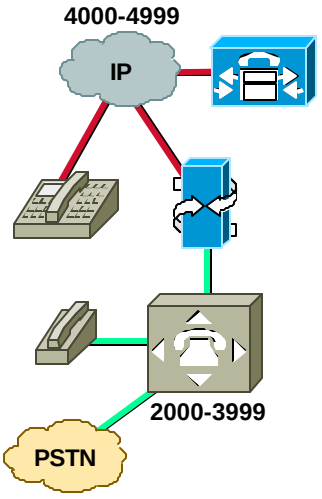
- Don’t forget the licenses!
- Each IP phone or virtual phone needs a license (per seat)
- Used when the phone registers
- Reuse the “primary” licenses on backup call managers
- No licenses needed for gateways, H.323 devices, ConfBridge, TAPI devices, etc..

Dial Plans

- How many digits?
- Do extensions map to DIDs?
- Interworking with a PBX?
- What codes for attendant, outside line?
- Restrictions on outgoing calls?
- Miscellany

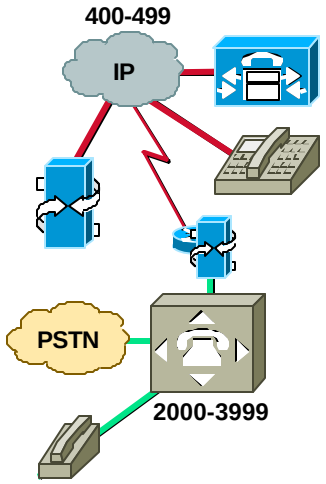
Dial Plan Examples: Migration

- Consistent dial plan between systems
- Both ranges map to DIDs
- Set codes for
  - Outside line (via PBX)
  - Attendant



Dial Plan Examples: Branch

- Separate dial plans and DID ranges
- Use trunk codes or tail-end hop-off to reach main site
- IP system uses local gateway and attendant



Call Restrictions

- Global call restrictions in 2.2

  Example: No calls to Nigeria, Yemen, or US 900 numbers

- Restrictions by group in 3.0

  Lobby phones can dial local and toll-free calls

  Executives can dial internationally

Dial Plan Miscellany

- **Reserve ranges (internal) for:**
  - **Voice mail**
  - **Meet-me conferencing**
  - **TAPI devices**
  - **Dialable devices (paging gateways, etc.)**
  - **Call Park**

How Much Bandwidth?

G.711	64k + Header	79.5k (on Ethernet)
G.729(a)	8k + Header	20k/10k (on PPP)
G.723.1	6.3k/5.3kk + Header	18k/8k (on PPP)

*Current Phones Use These Codecs

- **Compress RTP over low-bandwidth WAN**
- **WARNING! Process switched on the routers**
- **Administrator defines which CODEC in "regions"**
  - **Currently no low-bitrate conferencing**

What Happens During Congestion/Delay?

- **Dropped packets**
  - G.711 tolerates 10% drop rate
  - G.723 tolerates only a few in a row
- **Delay, jitter, and echo**
  - Jitter (variation of delay) is the true enemy
  - See talk number 402 for more information

QoS Solutions

- **Throw bandwidth at the problem**

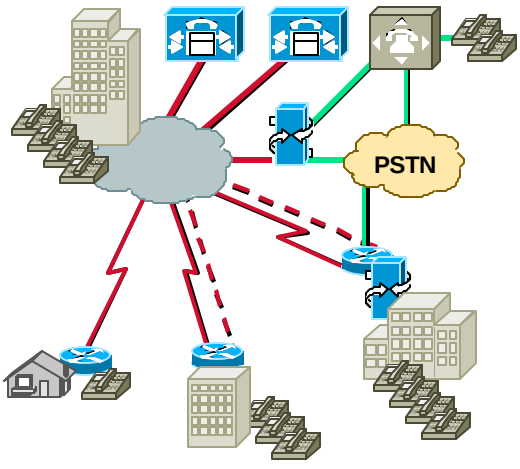
  or
- **Use advanced queuing and reservation**
  - WFQ with IP precedence
  - "Hard QoS"—ATM
  - Queuing on LAN switches
  - RSVP with admission control (future)
- **Use a reasonable CIR on frame links**
  - See talk number 407 for more information

Good Network Design

- Switched Ethernet to the desktop
- Redundant fast core
- Sufficient WAN bandwidth
- QoS
- Resources near main users
- IP multicast

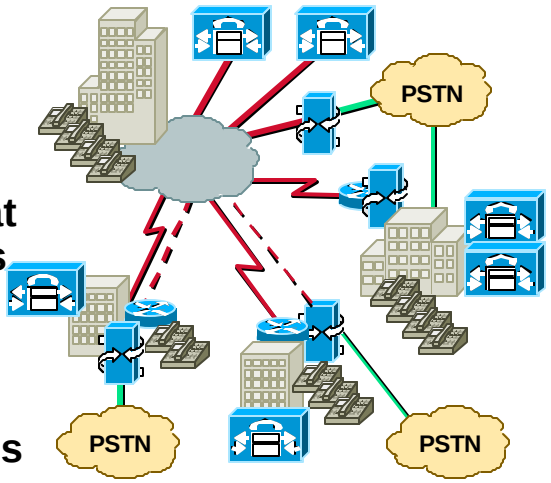
Centralized Design Example

- Redundant centralized call managers
- Dial backup
- Local gateways at large sites
- Link to PBX at central site (optional)



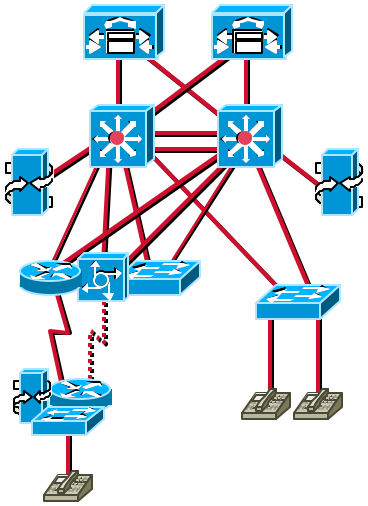
Distributed Design Example

- CallManager at each site
- Gateway(s) at each site
- Redundant CMs at large sites, others use central backup
- No dial backup at sites with two CMs



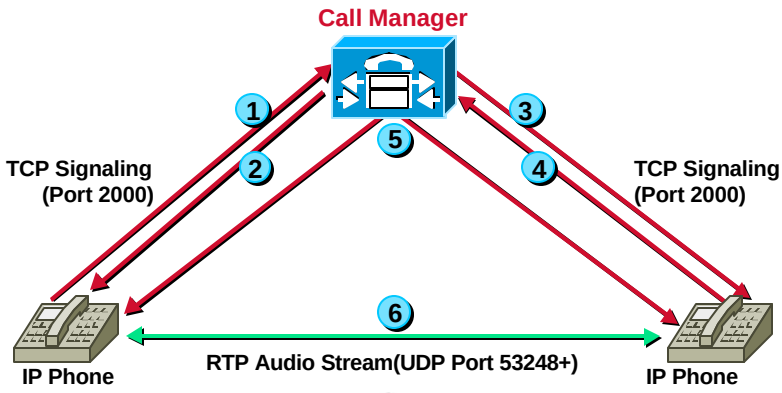
Network Redundancy

- Redundant switch uplinks
- Redundant core at Layer 2 and 3
- Redundant NICs
- Redundant gateways
- Backup WAN links (example: dial backup)
- See talk number 505



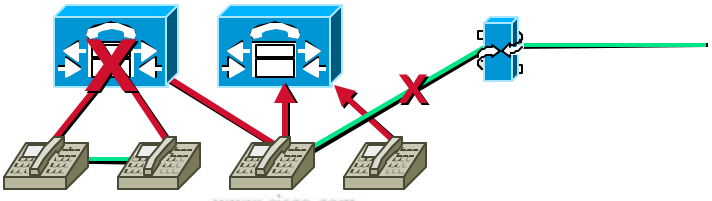
Making a Call

1. Off-Hook and Digit Stimulus
2. Play Tone Commands
3. Ring Command
4. Off-Hook Stimulus
5. Setup Media Stream Command
6. Audio Stream Established



CallManager Redundancy

- 2.2 failover
  - Phone-to-phone calls in-progress continue
  - Skinny gateway calls eventually fail
  - CDR stop record is lost
  - No transfer, conf, hold, or call waiting
  - Idle phones re-home to next server
- 3.0 redundancy and load sharing



Select Appropriate Gateways

- T1/E1 PRI or T1 CAS
  - DT-24+ or DE-30+
  - Cisco AS5300, Cisco 7200/7500, 2600, and 3600
- E1 R2
  - Cisco AS5300 only
- BRI or analog E&M
  - Cisco 1750, 2600, 3600
- Analog FXO or FXS
  - AT/AS
  - Cisco 1750, 2600, 3600
- "Skinny" gateways access supplementary services via Skinny Gateway
- H.323 gateways need to use a media termination point or SGCP in future
- G.723.1 in Cisco IOS™ 12.0(5)T

Agenda

- Prepare
- Install
  - CallManager and IP Phones
  - Gateways
  - Voice Mail and Other Applications
- Manage and Secure

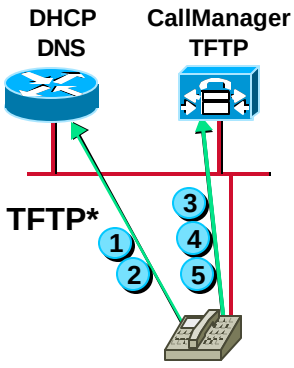
Installing—CallManager and IP Phones

- Addressing
- Phone keyboard templates
- Feature access
- Conferencing
- Missing features
- Managing bandwidth with regions
- CallManager redundancy configuration

Phone's Actions on Startup

1. Get IP address, mask, DNS, etc.
   - Static or DHCP
2. Get TFTP server address
   - Static address
   - Option 150 (single IP address)
   - Option 66 (first IP address or DNS name)
   - Lookup SelsiusCM1.your.domain
3. Get configuration from CallManager TFTP*
   - List of up to five CallManagers
   - Region info and keyboard template
   - Version of code to run
4. Get new code (one time only)
5. Register with CallManager

*Use configuration in flash after timeout

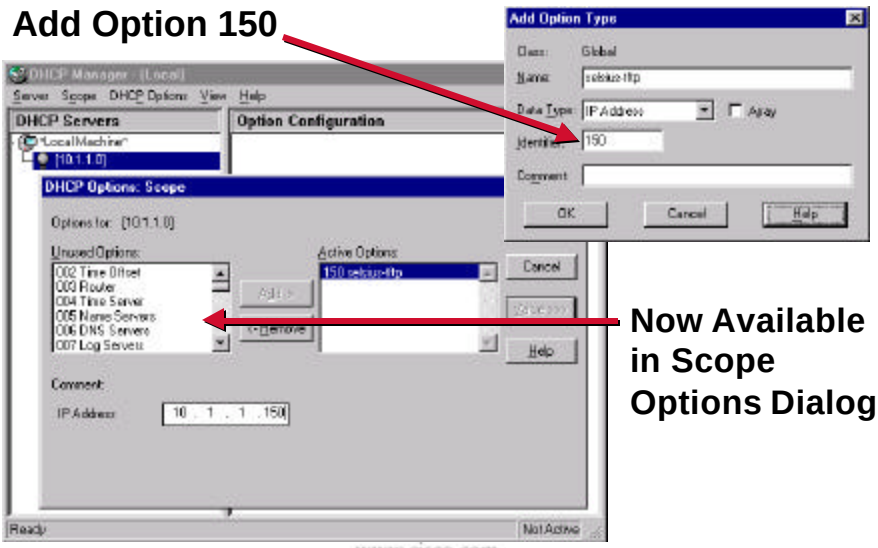


IP Addressing

- **Addressing gateways**
  - **"Symphony" gateway—static**
  - **AT/AS (Vega) and DT/DE+ (Titan II) use DHCP by default**
  - **Use Bootp application to push a static address into flash (DHCP will always overide flash)**
- **Addressing phones**
  - *** * # 1 to manually address**
  - **# skips current entry**
  - *** commits changes and acts like a .**
  - **Use 0*0*0*0 * to re-enable DHCP/reset settings**

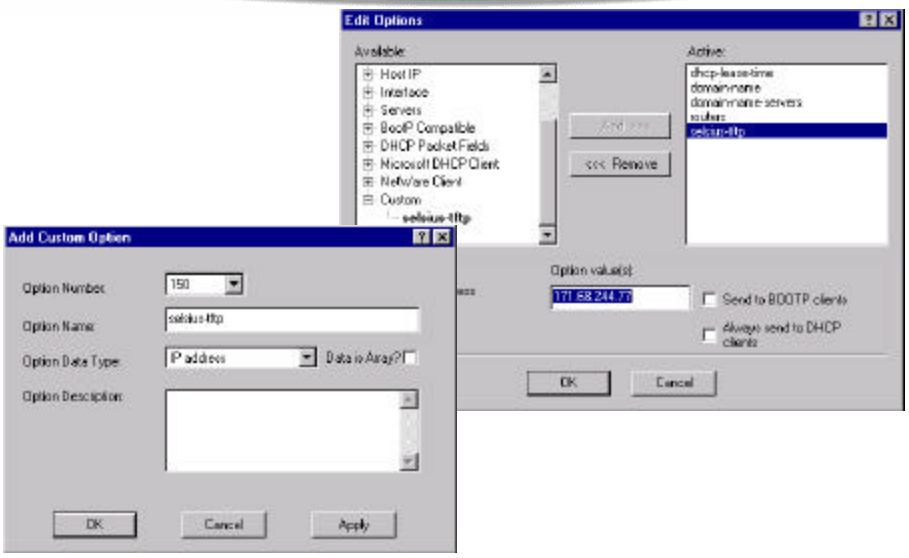
Option 150 for MS DHCP

Add Option 150



Now Available in Scope Options Dialog

Option 150 for CNR



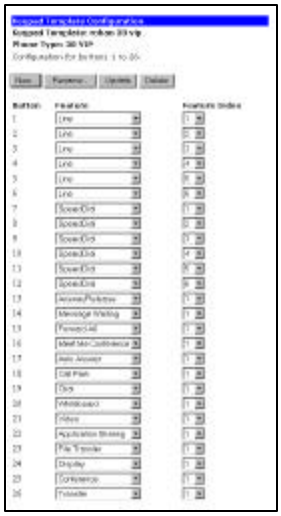
Autoregistration

- If enabled, new phones get a DN which follows them
- New entry appears in database by MAC address
- Manually add additional lines, change prime DN, change key template, etc.

Fig. 4: Auto Registration

Keypad Template

- Assign buttons to instances of lines, features, and speed dials
- No feature access codes in CallManager

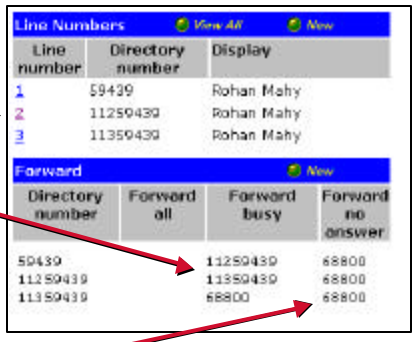


Line Behavior

- Primary DN
  - Call-waiting behavior
- Other copies of prime DN
  - Must forward-busy to other numbers
- Bridged DNs
  - Privacy is inherent
  - First user to pickup gets line

More Buttons for Your Prime Directory Number

- Create extra DNs
- Forward busy through this list
- Turn off call waiting
- Forward no answer to VM



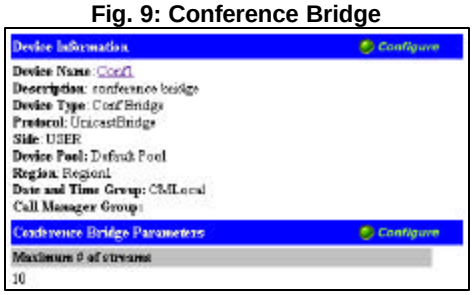
Feature Access

- Transfer
- Conference
- Hold
- Forward all
- Park
- Message waiting
- Collaboration
  - Application sharing
  - File transfer
  - Whiteboard
  - Chat
  - Video
- Answer/release
- Auto answer

Conferencing

- Ad-hoc conferencing
  - Use conference key
  - Install conference bridge software
- Meet-me conferencing
  - Assign range of DNs
  - Dial into conference
- G.711 ONLY

Fig. 9: Conference Bridge

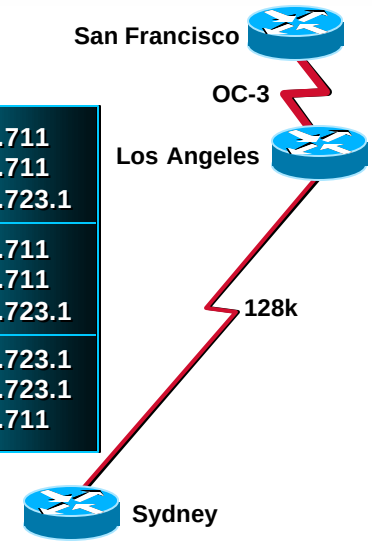


Missing Features

- Emergency dispatch (911, 112, 000)
- Music on-hold
- Call coverage
- …

Bandwidth Usage—Regions

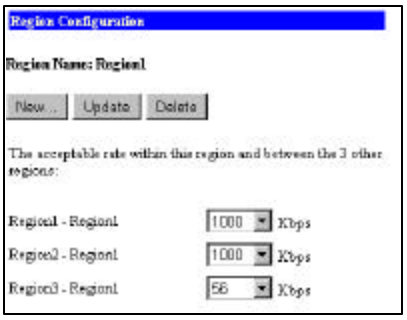
Los Angeles Los Angeles Los Angeles	Los Angeles San Francisco Sydney	G.711 G.711 G.723.1
San Francisco San Francisco San Francisco	Los Angeles San Francisco Sydney	G.711 G.711 G.723.1
Sydney Sydney Sydney	Los Angeles San Francisco Sydney	G.723.1 G.723.1 G.711



Configuring Regions

- **Configure table for each region**
  - **1000k—G.711**
  - **56k—G.723.1**
- **Currently no admission control**

Fig. 3: Codec Configuration



Setting up Redundancy

- TFTP server rebuilds configuration files with ordered list of servers
- Licenses are shared with backup system

CallManager Groups — Configure

CallManager Group	CallManager	Default Group
sns-lab-22-group	171.68.244.77	Yes
	171.68.244.72	

CallManagers — Configure

171.68.244.72
171.68.244.77

More CallManager Redundancy

- Use a script to sync database from primary to backups

  RoboCopy from NT Resource Kit

- PRI signaling is handled by CM on Skinny gateways
- No redundancy yet for H.323

Agenda

- Prepare
- Install
  - CallManager and IP Phones
  - Gateways
  - Voice Mail and Other Applications
- Manage and Secure

Gateway Configuration

- Route patterns
- Route points/groups
- Stripping/adding digits
- Dial restrictions
- PBX interworking
- H.323 devices
- Media termination point

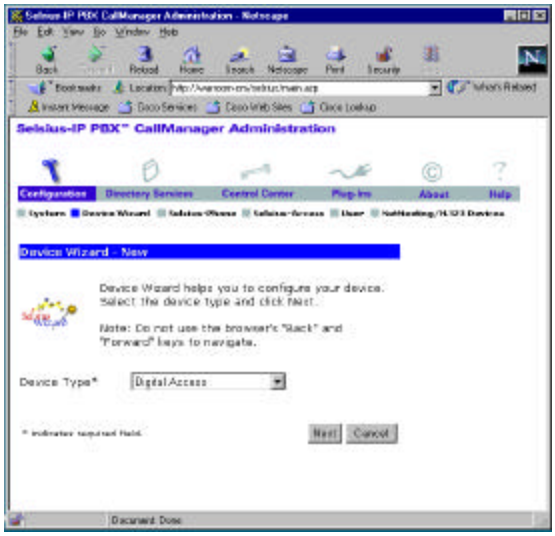
Route Pattern

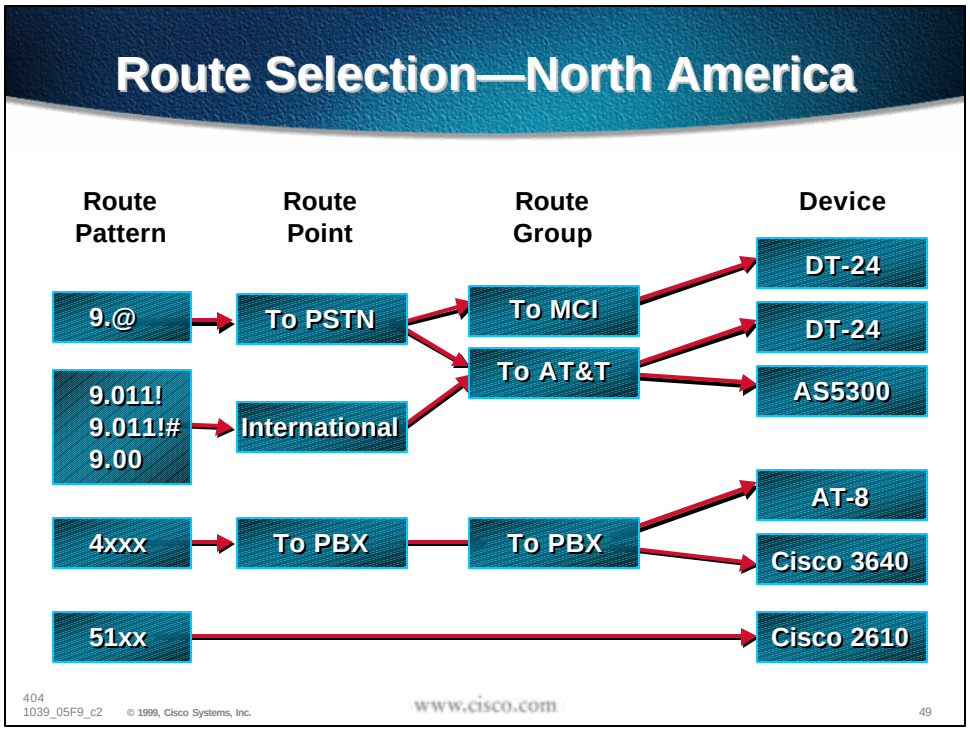
- **CallManager matches most specific pattern**
- **Wildcards**

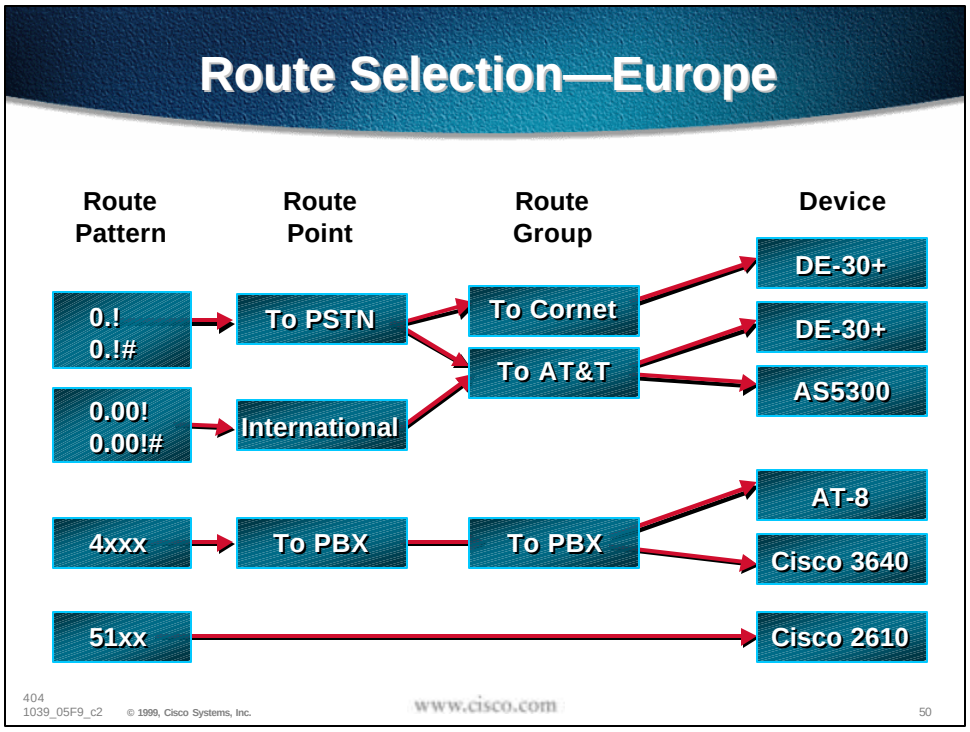
X	Single digit (0-9)
N	Single digit (2-9)
@	North American Numbering Plan
!	One or more digits (0-9)
.	Terminates access code

Add Gateway Device

- **Create skinny gateway by MAC address**
- **Create H.323 gateway by IP address or DNS name**







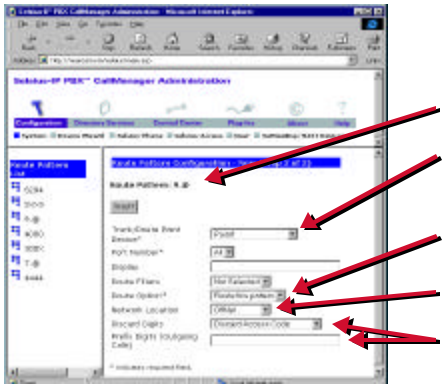
Add Route Group

- Choose the order of devices
- Choose the port number(s) for AT/AS devices
- ALL ports on digital and H.323 gateways

Add Route Point

- Create route point device in Device Wizard
- Choose region
- Choose order

Add Route Pattern



- Create route pattern
- Select which gateway or route point device
- Route or Block pattern
- OffNet = 2nd dial tone
- Add or Strip digits

Add or Strip Digits

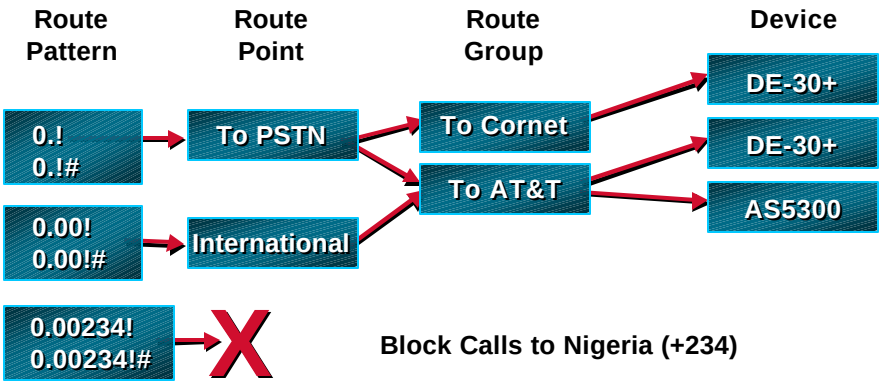
- **Outgoing calls**
  - Strip digits or add access codes to call the PSTN
  - Strip or add digits sent to a PBX
- **Incoming calls**
  - Strip extra digits coming from PSTN
  - Strip or add digits coming from a PBX
- **Examples**
  - Add 1010288
  - Add 9 to get out of Centrex
  - Add trunk access code 847
- **More examples**
  - Keep only last four digits of DID
  - Strip off trunk access codes
  - Strip off extra digits

Route Filters—North America

- Filter patterns
  - Country code
  - Area code
  - Exchange
  - Subscriber
  - Transit network
    - (1010288, etc.)
  - Service code
    - (911, 411, etc.)
- Ex: 1-900 calls

Route Filters—Anywhere

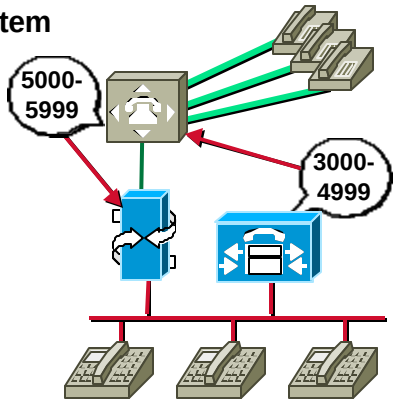
- Create route patterns to block



Block Calls to Nigeria (+234)

Interworking with a PBX

- Use a digital gateway for called/calling party ID
  - T1/E1 or BRI
- Reserve DN range for each system
- Add/strip digits into a consistent dial plan
- Keep features
  - Dial
  - Transfer
  - Conference
  - Hold
  - Forward
  - Calling party name/number
  - Pickup parked call



Configuring H.323 Gateways

- Add a device (Device Wizard)
- Use IP address or DNS name

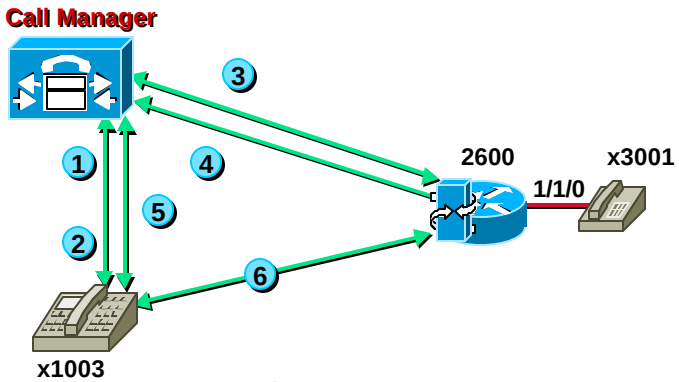
  (The loopback address for a router)
- Select network side in most cases

Cisco IOS Gateway Configuration

```
!
dial-peer voice 1000 voip
 destination-pattern 1…
 codec g711ulaw
 session target ipv4:172.16.45.10          <----  Call Manager's IP
!
dial-peer voice 3001 pots
 destination-pattern 3001
 port 1/1/0
!
interface Loopback0
 ip address 172.16.1.100 255.255.255.255
!
interface Ethernet0/0
ip helper-address 172.16.45.12             < -------  DHCP server's  IP
!
```

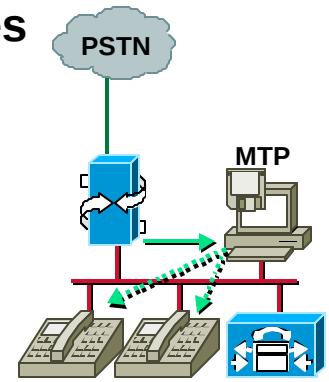

H.323 Gateway Signaling

1. Off-Hook and Digit Stimulus
2. Play Tone Commands
3. H.323 SETUP
4. H.323 CONNECT
5. Setup Media Stream Command
6. Audio Stream Established



Media Termination Point

- Provide supp services to H.323 devices
- NT software (on CM server or elsewhere)
- Acts as skinny client to redirect calls
- G.711 only (A-law or u-law)



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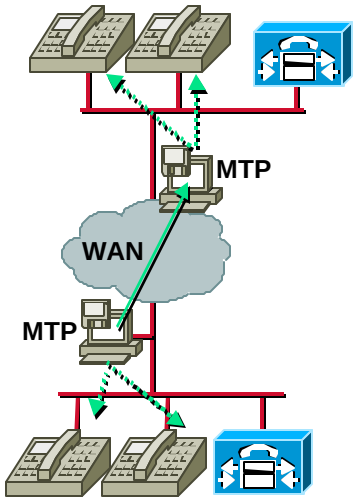
MTP between CallManagers

- CallManagers can communicate with H.323 also
- Pair of MTPs enable supp services from either network



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Agenda

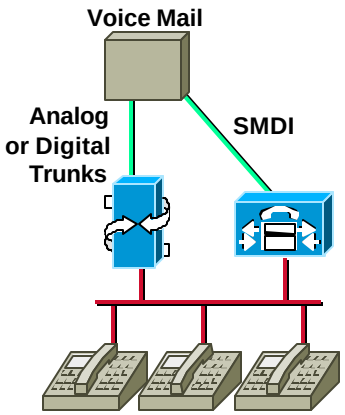
- Prepare
- Install
  - CallManager and IP Phones
  - Gateways
  - Voice Mail and Other Applications
- Manage and Secure

Voice Mail and Other Applications

- Legacy voice mail integration issues
- SMDI
- TAPI
- Amteva UMS
- Attendant console
- ACD, IVR, AutoAttendant, TAPI dialer

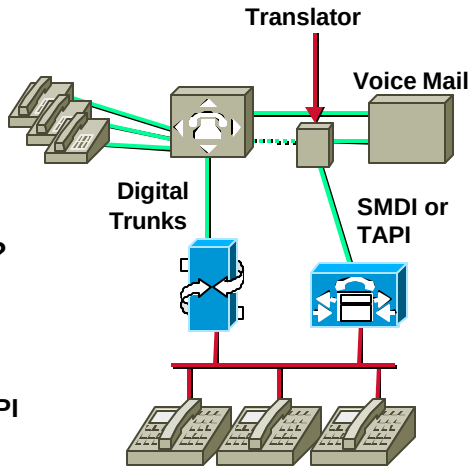
Legacy Integration: No PBX

- Need SMDI to deliver
  - Called party
  - Calling party
  - Call reason
    - Forward all
    - Forward busy
    - Forward no answer
    - Direct call
  - Message-waiting information



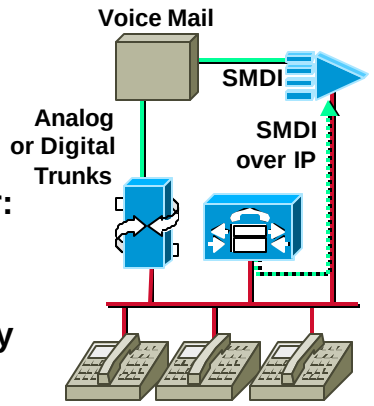
Legacy Voice Mail: with PBX Migration

- What integration is used for PBX?
  - Digital set
  - Analog + SMDI
  - Analog + tones
- Multisystem support?
- Multi-integration support?
- Three options:
  - Dual SMDI integration
  - Translate current integration to SMDI or TAPI
  - Live without message waiting



SMDI

- Install SUMI plug-in
- Connect null-modem serial cable to SMDI port on voice mail
- Or install COM port redirector: connect terminal server or aux port to voice mail
- One SMDI interface only today
- Not redundant until 3.0

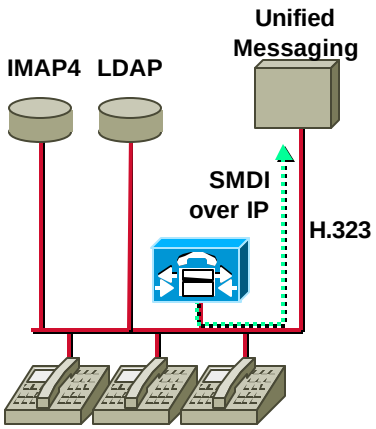


TAPI Interface

- TAPI 2.1 drivers (1st and 3rd party)
  - 1st party participates in call (voice mail)
  - 3rd part monitors calls (TAPI dialer)
- Access to partner applications: Voice mail, IVR, dialer, recording, etc…
  - ex: Active voice, Telekol, etc…
- Provides called/calling party called reason and message waiting info
- TAPI 3.0 and CSCOAPI in the future

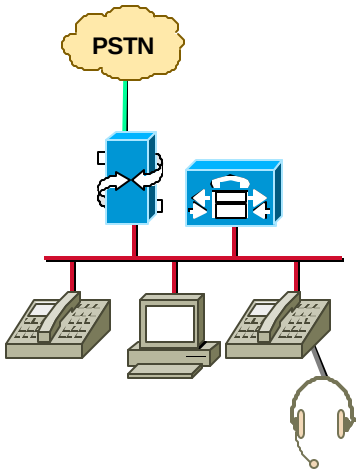
Amteva UMS

- Unified messaging on UNIX and NT
- Configuration stored in LDAP
- Messages stored in IMAP4
- Use H.323 and SMDI over IP for IP-only solution
- Forward busy/no answer to H.323 device



Manual Attendant Console

- Web-based console
- Shows status of lines
- Install TCD and LSS on CallManager
- Install client
- Associate with and IP phone



Coming Soon!

- **Other Cisco applications:**
  - **TAPI dialer (SoftPhone)**
  - **IVR**
  - **Auto-attendant**
  - **Low-end call center**

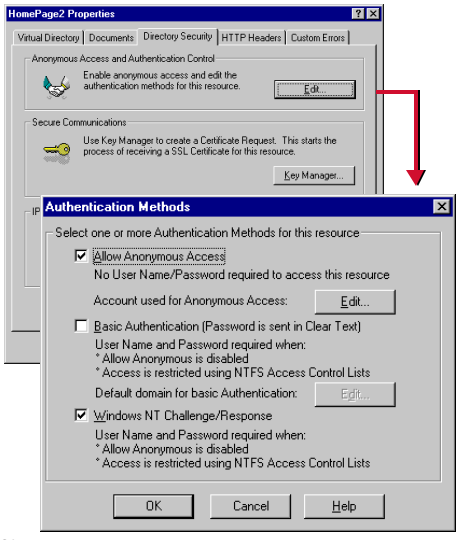
Agenda

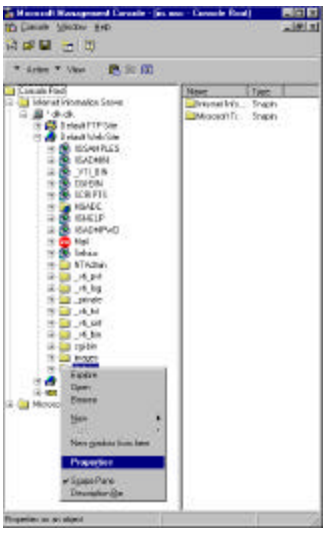
- **Prepare**
- **Install**
  - **CallManager and IP Phones**
  - **Gateways**
  - **Voice Mail and Other Applications**
- **Manage and Secure**

Management and Security

- Securing web admin
- Remote NT admin
- Using another database
- Using call detail records
- Managing reloads
- Turning on trace
- Printing paper labels
- Troubleshooting phones

Web Admin Password Access

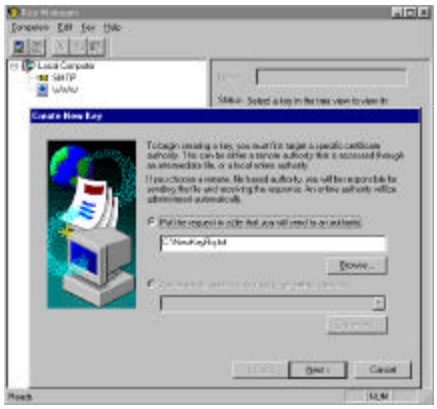




Web Admin SSL Encryption

- Run Key Wizard and get a certificate

  ex: http://www.thawte.com



Tools for Remote NT Admin

- Web interface
- Telnet, rsh, and rcp services

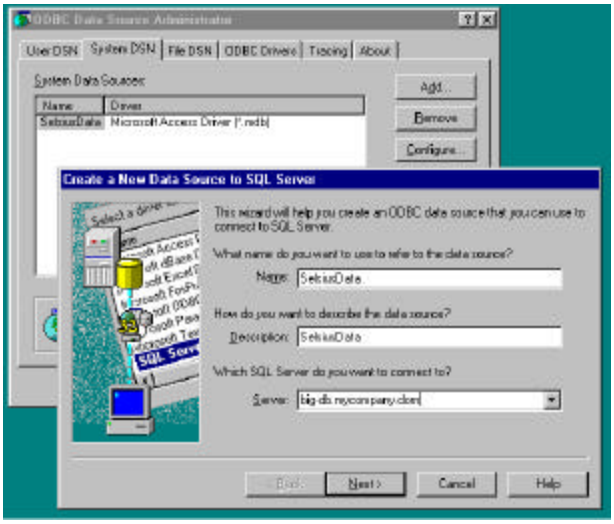
  start and stop services

  tracert, ping, netstat
- Remote NT GUI

  DHCP Admin, User Admin, Performance Monitor, Event Viewer
- SNMP

Using Another Database

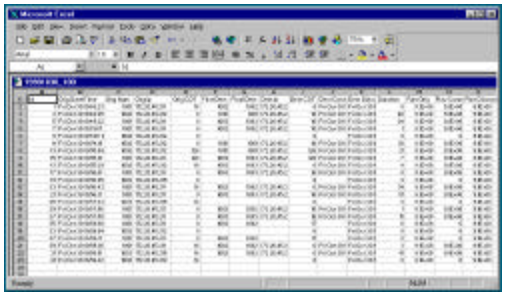
- CM uses access database by default
- Use ODBC to use another database



Using Call Detail Records

- Enable in SCM.INI in\<CM root>\bin
- Puts CDR in Access or ODBC Database

Fig. 14: Excel Spreadsheet of CDR Records



CallManager Reloads

- Reload CallManager after change to the route patterns
- No reload for new phones
- Reloads will disconnect calls in progress
- To fall over cleanly, unplug primary CM's Ethernet, then stop and start

Turning on Trace

- Turn on Trace in SCM.INI

  [SDI]
  SDIFile=1
  …
  [Trace]
  TraceFlag=1
- SelsiusTrace works like "tail -f <tracefile>"
- Watch for

  Device/pattern registrations
  Stimulus messages from phones
  Matched patterns
  H.323 (Q.931-like) messages

Printing Paper Labels

- Print from CallManager
- Use MS Word template

http://www.selsius.com/documentation/v22/12STemplate(Word97).doc
http://www.selsius.com/documentation/v22/30VIPTemplate(Word97).doc

Troubleshooting IP phones

- Got an IP address?
- Found a TFTP server?
- Got it's configuration?
- Got it's load?
- Registered with a CallManager?
- DB error or no licenses?