PCM, DS0s, Audio Quality, and Old Stuff
8000 samples a sec with -127 to -127 for voltage (8 bits binary- PCM scale)
8 x 8000 = 64k/ sec which is where that comes from.

Compression comes from dropping unneeded repeated samples
- also chopping unused frequencies (lowers bits per sample) like the Nyquist method:
Ear = 20-20khz, speech= 200-9khz, so nyquist range = 300-4khz
Nyquist theory (rate and frequency) is outside the scope of this.
https://en.wikipedia.org/wiki/Nyquist_frequency

Steps: sample analog (8000 x sec again), number each sample (quantify), convert to binary (PCM), compress ulaw or alaw

Compression codecs: for all below, 33/50 PPS and 33/20 sample size (with default compression)

G.711	64k	4.3 MOS	uncompressed	160/240 bytes
G.729	8k	3.92	compressed	20/30
ILBC	15.2k	4.14	open source comp	38/50

Due to differing compression, EUR VoIP has to be converted to US VoIP (ulaw and alaw)

MOS is Mean Opinion Score for quality evaluation- is now computed, but originally rated by a human listener hearing phrase "nowadays, a chicken leg is a rare dish"
5 is best and 1 is poor, PESQ Perceptual Eval of Sound Quality, PEAQ is Audio Quality

Back to PCM - 64k is DS0, 24 in a t1- clear correlation between common BW and sample rates.
It is literally a Time Division Multiplexing(TDM), since we are measuring separate waveforms (voice channels/ phone calls) of voltage over time.

Flow control on a DS0, CCS and CAS - Common Channel and Channel-Associated Signaling
CAS lower available bandwidth - 64 kbit/s DS0 to 56 kbit/s - robbed-bit signaling for control info.
CCS (1960s) more secure and effective- dedicated separate control channel to govern traffic
for multiple data channels. AKA common-channel interoffice signaling (CCIS). Most common CCS signaling methods on public switched telephone network (PSTN) are Integrated Services Digital Network (ISDN) and Signaling System #7 (SS7)
- ISDN used primarily on trunks connecting end-user private branch exchange (PBX) systems to a CO, SS7 is primarily used within the PSTN between COs (sort of like BGP is for data nets).
- Greater trunking efficiency due to the quicker set up and clear down, thereby reducing traffic on the network, more info along with the signaling traffic for features such as caller ID, mid-call signaling. In CSS, Central Offices comprise Common Channel Signaling Switching Offices (CCSSOs), database and operator systems

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Ground start

From Wikipedia, the free encyclopedia

In telephony, a ground start or GST is a method of signaling from a terminal or subscriber local loop to a telephone exchange, in which method a cable pair is temporarily grounded to request dial tone. Most middle 20th-century American payphones used "coin first" ground start lines, with the starting ground passing through the coin itself.

Ground start trunk

Local telephone companies typically provide two types of dial tone switched trunks – ground start and loop start. Private branch exchanges (PBX) work best on ground start trunks because those trunks can give them an on hook signal allowing for timely clearing. Many will work – albeit intermittently – on both types.

GLARE in Loop Start

Normal single line phones and key systems typically work on loop start lines. The issue with loop start lines is that the PBX and central office can seize the line simultaneously; since neither gets the response it is expecting, the call is not initiated. The resulting condition is called **glare** or also known as call collision.

In an idle circuit, the central office supplies -48V (nominally) on the ring and open on the tip. A ground start PBX initiates an outgoing trunk seizure on an idle circuit by connecting of the ring lead to ground (maximum local resistance of 550 ohms). The central office senses this condition and grounds the tip lead. When the PBX senses this, it goes off-hook, then removes the ground on ring. The central office sends dial tone and the rest of the call proceeds normally.

In ground start signaling, the central office initiates a call by grounding tip and putting the ringing signal on the line. The PBX has 100ms to sense this condition. The PBX goes off hook; if it had been trying to seize the line by grounding ring, it releases ring from ground and the call proceeds normally.

At the end of either an incoming or outgoing call, the PBX initiates disconnect by going on hook, or the central office initiates disconnect by opening tip. When the other end detects the loss of loop current, it also goes on hook and the call clears normally.

A PBX user must be careful to order the correct type of trunk line from the local phone company and correctly install the telephone system at the PBX end – so that they match. Line equipment in most 20th-century central office switches had to be specially rewired to create a ground start DDCO line. Crossbar switch did it with a paper sleeve on the Vertical Off Normal contact, 5ESS switch by translation, and DMS-100 by a slide switch on the line card, all according to what the customer ordered.

Loop start
From Wikipedia, the free encyclopedia

Loop start is a telecommunications supervisory signal provided by a telephone or private branch exchange (PBX) equipment in response to the establishment of a closed local loop, commonly referred to as the off-hook state. When idle, or on-hook, the loop potential is held a nominal 48V DC, provided by the telephone exchange or a foreign exchange station (FXS) interface. When a terminal initiates use of the line, it causes current to flow by closing the loop, and this signals the FXS end to provide dial tone on the line and to expect dial signals, in form of DTMF digits or dial pulses, or a hook flash. When the loop is opened and current stops flowing, the subscriber equipment signals that it has finished using the line; the telephone exchange resets the line to an idle state. When the FXS rings a telephone, it superimposes an alternating current (AC) signal onto the line. The most common ringing frequency is 20Hz. This amplitude is typically approximately 90V, but regulations prescribe that a minimum of 40V must be recognized. The ringing power was historically produced by a hand-cranked generator in the operator console of the exchange, or attached to a customer's telephone.

An alternative to loop start signaling is ground start signaling.

Other signaling
Modern loop start trunks also have methods of answer supervision and disconnect supervision to alert a foreign exchange office (FXO) interface that the remote party has answered or hung up. Answer **supervision** usually takes the form of the central office reversing the polarity of the line for the duration of the call when it has been answered. This condition is called Reverse Battery or battery reversal. Disconnect supervision can take the form of the polarity reversing back, or removing voltage from the line for a short period of time (battery drop). Disconnect supervision explicitly provided over an interface using battery drop is known in the Asterisk PBX community as Kewlstart.

Another type of loop signaling is the ability for the customer to signal the central office that they would like to make a second simultaneous call, a three-way conference call, or answer a second incoming call. This signal is called flashing or hook flash, and is performed by interrupting the loop for a fraction of a second, usually around 600ms. The flash is longer than a rotary dial pulse, sometimes called a short flash, and is shorter than a hang-up (on-hook).

The term local Loop comes from the voltage loop start at residence, which is not grounded to a PBX directly like a business which instead has ground starts going to the PBX. The residential loop is very much like a loop with each station (phone/number) linked up similar to token ring (one way around)
The diagram presented illustrated SS7 trunks between COs and trunk to a business building private switch labelled simply CCS- this was probably more accurately an

ISDN.

- in PSTN, analog Key Systems link up, while private PBXs are typically digital
- T1 typically means CAS while PRI means CSS (I question this)
- A gateway (Cisco 2921 pictured) can bridge PSTN/ PBX (T1/PRI), analog FXO/FXS [Foreign Exchange Office (PSTN)/ Station(phone)], and VoIP (ethernet) traffic
- Pictured switch has connections out to PBX for lots of POTS lines, individual ports for single FXO/S lines, lots of VoIP ports

Aside on PSTN numbering systems: ITU-E164
1 - 480 - 555 - 1212 --- country (1) Nat'l Dest (480-555) and Subscriber (1212)
Inside a country you can have your own plan like the North Am Numbering Plan (NANP):
[country] [area code] [CO code] [station code]

Why VoIP:
Reduced call costs, cabling costs, no Ana-Dig converters, move/add/change cost, no LD costs
Seamless unified networks and services (voice, fax, email), open standards, more SW and features
Less phone tag with rollover and control capabilities (supposedly)

DSPs:
- Offload processing on voice gateways for coding (Analog to Digi), transcoding (alaw to ulaw)
- provide a Media Termination Point:
    - When someone is put on hold, the station (phone) does hang up - but the MTP keeps the call active to the system, so you can dial out, join calls, etc. same with conference calls- need MTP
    - Conference systems need more DSPs to be more MTPs
- [login required] http://www.cisco.com/web/applicat/dsprecal/dsp_calc.html
- tool tells how many DSPs needed
- DSP modules PVDM1, 2, PVDM3 the latest.
- It's recommended to have plenty provisioned when budget allows, when demand catches up
- DSPs limited in # of calls they can handle
- can be grouped into teams, farms, so they can be directed as such for allocation

RTP, RTCP - Realtime Transport Protocol and Control Protocol
- work on top of UDP, RTP even #'d ports 16384-32787; RTCP odd ports following RTP # ***
- RTP provisions time stamps, sequence numbers to order and remove jitter
- de-jitter buffer watches ahead for delay variation
- RTCP mainly for stats- not control. Reports count, delay, loss, jitter (remember, jitter=packet1delay-packet2delay)

==

*** On PORTS from RFC 3551
https://tools.ietf.org/html/rfc3551
Applications operating under this profile MAY use any such UDP port pair. For example, the port pair MAY be allocated randomly by a session management program. A single fixed port number pair cannot be required because multiple applications using this profile are likely to run on the same host, and there are some operating systems that do not allow multiple processes to use the same UDP port with different multicast addresses.

However, port numbers **5004 and 5005 have been registered** for use with this profile for those applications that choose to use them as the default pair. Applications that operate under multiple profiles MAY use this port pair as an indication to select this profile if they are not subject to the constraint of the previous paragraph
==================

Cisco Unified Communications Manager Express (CUCME)
http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/requirements/guide/cme115spc.html
http://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-express/tsd-products-support-series-home.html
- Was originally a product in the '90s from Selsius (Call Manager)
- Most people still just call it "call manager"

Base system has conferencing, voice mail, etc.
Later became an ISR license upgrade
(Pictured is a Cisco 1861, just had end-of-life notice in May 2016)
Supported 5-450 phones
Use IOS or the CCP GUI to configure
- Interactive voice response (IVR) style phone tree
- Computer Telephony Integration (CTI) allows supported apps to use - such as Salesforce- when customer calls, number brings up customer records automatically.

Cisco Unity Express (CUE) Integration with card installed for VM integration
- Trunk to other phone systems
    - TSP (legacy) - telephony service provider - for T1/FXO ports
    - Internet TSP (ITSP) - where provider gives SIP links so the old lines aren't coming in
    - H323 - trunk to other phone systems or CUCM

Cisco Unified Communications Manager - CUCM
Originally ran on top of NT4 (2000-2003) until Cisco got sick of MS security hell
Now a virtual appliance running on top of a Linux core
Cluster-based design 40k IP phones max per cluster (but 80k possible)
The cluster and cluster-size is more related to the DB
Multiple clusters easily added, LDAP compatible (AD integration); backup software for built-in DR

There are two primary functions operating in CUCM devices:
Database operations: In a cluster, there is one designated DB publisher and up to 19 subscribers
 - Subscribers are read-only unless publisher goes down.
- Only user-facing features [message-waiting indicator (MWI), forwarding, etc] are able to be written to subscribers when a publisher goes down. When publisher comes back online, it synchronizes, and subscribers are again read-only.

Runtime Communications (ICCS) - Actual operations-service routers
 - call processing, active communications
 - There is a limit to 8 call processing nodes (why you will never have the max 19 DB subscribers)
 - Configured on each phone as a primary, another secondary, tertiary, etc

<u>Unity's Evolution:</u>
Unity Call Manager - (legacy, NT4) - 15k limit
Unity Express - 30k users
Unity Connection - 20k users - (newest, and the "real deal")
Unity BE 4000 - Business Edition - 1k users - another sub-version like Express
http://www.cisco.com/c/en/us/products/unified-communications/unity-connection/index.html

Active/Active, Exchange integration, same appliance OS as CUCM
VPIMv2 - Voice Profile for Internet Mail (RFC 3801) subset of MIME for use between voice processing server platforms- extending the familiar myEmailAddress@myDomain.com addressing to voice mail and fax systems.

These systems (except BE) are independent
- Integrated with SCCP and SIP for signaling and with pilot numbers (phone extensions).
- CUCM publisher references DB calls to Unity to handle and Unity signals MWI etc, to destinations.

SCCP aka "Skinny" phones:
https://en.wikipedia.org/wiki/Skinny_Call_Control_Protocol

Cisco UCM IM&P
 - Used to be standalone and now part of CUCM
 - 2 modes: UC Mode (45k users and Standalone (75 max users)
 - Another "mode" now is Skype for Business - was Lync InterOp/MS Call Mgr with 40k users
 - privatized versions of Google Hangouts/ Skype/ WebEx Etc. (security and compliance)
 - Allows connections to outside domains
 - Now best suited for Jabber (which Cisco purchased)

Cisco Jabber
- http://www.cisco.com/web/products/voice/jabber.html
- all-in-one Presence, IM, voice, video, voice messaging, desktop sharing, conferencing

TelePresence Video Communication Server (VCS)
http://www.cisco.com/c/en/us/products/unified-communications/telepresence-video-communication-server-vcs/index.html
- Like CUCM for TP endpoints
- can handle "uncontrolled endpoints" like a WebEx plug-in or other outside video call
- VCS Expressway - most known feature- can traverse firewall through core and edge components
- enables Jabber Client, hyperlinks to start videoconf (with VPIMv2 address)
- incoming and outgoing requests

Cisco TelePresence Management Suite (TMS)
http://www.cisco.com/c/en/us/products/conferencing/telepresence-management-suite-tms/index.html
- centralized management of multiparty conferencing, infrastructure, and endpoints.
- telepresence device provisioning
- easily schedule Outlook and Exchange telepresence
- multisource phonebooks
- management of in-progress calls, conferences
- resource utilization reporting

IP Phone
- needs integrated switch, it's own IP address
- make sure the switch has a power budget built in to handle the PoE you need
- make sure you know the PoE standard it uses! 802.3af (15.4w), 802.3at (25.5w)
- or pre-standard "Cisco Inline Power" (7940/7960 IP phones)
- minor considerations might be desk placement, but that's already an issue for workstations
- some devices (sidecars) might need a power brick
- be sure to audit post-deployment, users too- make sure support was given.

For voice VLAN, port on switch is an access port, not a trunk.
Cisco phones are able to put VLAN tags on and use untagged frames for the attached workstation
If you plug in a noncompatible phone or device it wont see the voice VLAN (security feature)
CUCM obviously lives on the same VLAN as voice traffic (say VLAN 10)

IOS
show power inline -- for power info
show cdp neighbors   --will also show phones, models
show cdp neighbors detail
show lldp neighbors   -- says Cisco is switching to LLDP from CDP (?)
show cdp detail <device>

show cdp entry <item keyword>

Made voice and data vlans, apply the to interface:
int fastEthernet 0/0/30
switchport access vlan 20
switchport voice vlan 10
show vlan
show interfaces fastEthernet 0/0 switchport

Powerup for first time:
1. Switch supplies power, phone boots SW image/ firmware
2 Switch provides VLAN info to phone via CDP. In SIP this is CTL (certificate-based to authenticate legitimate servers and devices)
3. Phone sends DHCP request, where it gets tftp info too (option 66 is DNS, option 150 is IP)
4. Phone gets config file from tftp server (cnf.xml with 3 CUCM servers, right FW vers. and locale)
5. Phone contacts first CUCM server listed in the config (CUCM Group), registers, gets softkey template - establishes SCCP/skinny connection
 With SIP, step 5 downloads dial rules and stuff from tftp

SCCP is like a dumb terminal, SIP is more independent

SCCP/skinny does everything through CUCM except the RTP/RTCP traffic which is endpoint to endpoint. SIP is replacing SCCP/skinny

Regular boot first time: switch detects with CDP, negotiates power needs. Switch gives voice VLAN, phone restarts into right VLAN, does the rest above. Dailtone can happen when CUCM is resolved.

Hard Reset
Unplug ethernet, plug back in holding # key down until speaker light comes on.
On keypad type 123456789*0# (all the keys on keypad in order)

Note: one voice packet contains 20msec
If one drops it fills the space between by stretching the neighboring packets's contents

Audio vs Video requirements:
<30ms jitter
<150ms delay
<1% loss
QOS: DSCP EF (for audio) DSCP AF41 (for video)
Audio little BW, Video lots of BW
What's more important? audio wins out (which is better video but no audio or ?)
Priority: make at most 4 categories of traffic priorities

Types of QoS

"QoS doesn't engage unless there is congestion" (hmm)
- best effort (fifo)
- Integrated Services (intserv) RSVP'ing bandwidth
- Differentiated services (diffserv) - classification and marking

 - Queuing (congestion management)
 - Congestion avoidance - WRED Weighted Random Early Detection. Old crude type is Tail Drop
 - Link efficiency

Queuing strategies
Weighted Fair Queuing (WFQ) - low traffic senders get priority over high senders
Class-based WFQ - defines bandwidth among classes that are defined- jitter on queuing of things!
Low Latency Queuing (LLQ) - Characteristics of the above with a priority queue added - Class-based WFQ with strict priority elements (PQ-CB-WFQ priority queue + the others) - works so you know how big the pipe is for voice data and can figure out how much can fit at a time

"tail drop" is when buffer gets full so everything is dumped out

payload compression (squish the data - codecs)
header compression - squishing the header
LFI - link fragmentation and interleaving - split big packets - big packets get serialization delay- the delay of loading the packet into transport

See videos 16 and 17 for QoS stuff - REVIEW or REVIEW elsewhere
Inbound QoS
Classify, mark, police
Outbound QoS
Queue, mark, avoid, shape, police, compress, LFI

AutoQoS VoIP/Discovery
- Provides a template-based config (run traffic, let it watch) , allows manual tuning
 - reduces deployment, minimizes human error, provides consistency

IOS:
int fa0/1
auto qos voip [cisco-phone I cisco-softphone I trust] --trust is use DSCP/CoS

If you chose cisco-phone it dumps this into the running config:
srr-queue bandwidth share 10 10 60 20
priority-queue out
mls qos trust device cisco-phone
mls qos trust cos
auto qos voip cisco-phone

spanning-tree portfast
service-policy input AutoQoS-Police-CiscoPhone
-- it chose a service policy/ class map setup for us
-- and yeah portfast is fine since we are in a non-BDPU area on this side of the phone

no auto qos voip - to zap what it put in

You can develop CLI modular service class-maps

GNS3
Used c3725
Use Cloud on maps and attach it to your computers eth0 to get out, and DHCP assign

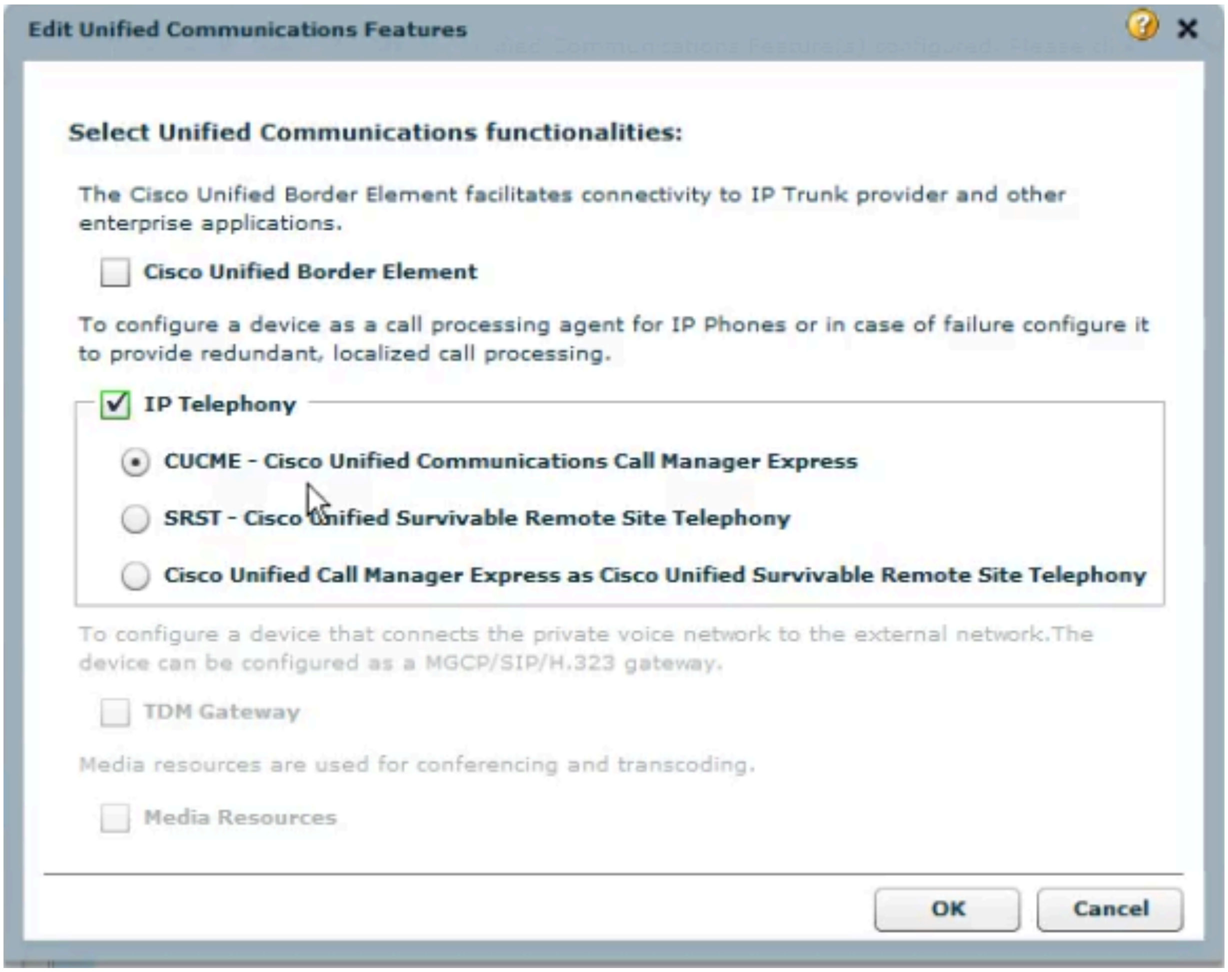
CCP Cisco Configuration Professional
CCP Express for basic LAN/WAN connections, firewall, and NAT, but no UC
- cisco-config-pro-k9-pkg-2_8-en.zip
- Runs on Java in browser window
- Complete core config via GUI
- log into router, set up enable, secret passwords, a username etc, telnet/ssh, and http-server

enable --- when you do this, you are actually entering privilege 15 mode
hostname R1
enable secret cisco
username pascal privilege 15 secret cisco
ip http secure-server (you'll see it reporting key generation, etc 1024 default)
! ip http server is fine but unencrypted
ip http authentication local
line vty 0 15
login local
transport input telnet ssh
! optional dhcp config
ip dhcp pool DHCP_POOL
network 10.0.0.0 /24
default-router 10.0.0.1
exit
ip dhcp-excluded 10.0.0.1 10.0.0.5
end

<u>Unified Communication Configuration</u>
Default welcome page has CCP News items on top, Community Info at the bottom (devices)
Discover, discovery details (more info like IOS and model) and manage devices are options for that.

Telephony settings, extensions, etc won't work in GNS3! Using the 2801 router image helps with this. Says IOS 15.1(4)M9 comes with UC 8.6 built in (R2)
[the other router in example (R1) is IOS 12.4(24)T8 on a 7206VXR]

On top click 'configure', the on the sidebar click unified communications> features, which brings up the
Unified Communications Features "Select UC functionalities" window



Cisco Unified Border Element (CUBE)
Use when you need one device to talk to and be the border agent to a ITSP SIP provider
(DC of Asterisk VoIP servers was example, they need one device to be 'bridge' to ITSP)
(http://www.asterisk.org/downloads)

IP Telephony
CUCME - Cisco UC Call Mgr Express - this is the only one we use here. Turn it on click ok, and other items under Unified Communications configs listing show up

Cisco Unified Survivable Remote Site Telephony (SRST)
You have a remote branch with IP phones working through CUCM devices at another location over t1, fxo, digital to ITSP. If the main connection goes down, the SRST kicks in. License for CUCME+SRST on same device costs more- it's the last choice here-

telephony doesn't just survive outage but has all the features

In CCP UC Features, TDM Gateway and Media Resources will be greyed out if no DSPs are present
Gateway checkbox is just that- a line to outside phone system types. Media resources checkbox allows offering DSP services to other CUCM devices when needed

After config'ing CUCME as above command line preview is given:

```
telephony-service
 no auto-reg-ephone
 max-redirect 5
 web admin system name user secret 0 *****
 exit
```

<u>Items under Unified Communications:</u>
Unified Communications Features
Telephony Settings
Users, Phones and Extensions (folder)
VoIP Settings
Trunks (folder)
Dial Plans (folder)
Telephony Features (folder)
Media Resources (greyed out)
Unified Communications Security Audit

Telephony Settings - shows parameters and config'd value
    (clicking 'Edit'):
    Supported endpoints: SCCP(skinny), SIP(voice register pools) or both
        (these should be configured in IOS)
    Number of phones: (25 or other- lists supported/licensed/allowed number)
    Number of extensions: (what it sound like - "ephone-dn tag <#>" in CLI)
    (date/time format)
    Phone registration Source IP: primary CUCM device IP
    Secondary dial-tone digit: 9
    Softkey functions (optional) - FXO hook flash and hunt group logout (Hlog)
System Config
    Message displayed on phones:
    Directory naming schema: First Name First
    Music on hold option (pulldown)
    Phone default PIN
    Web admin settings for system and customer web admin accounts
        user/pass, enable customer web account user/pass
Timeouts
Dataplan Pattern
Transfer Pattern
Phone URLs

Set SCCP, an ipaddy, 25 users, 50 extensions, only gen settings
click ok/save- get preview commands:
    ---> [you need to enter these manually to "unlock" the other configurations in GNS3]

```
telephony-service
 secondary-dialtone 9
 ip source-address 172.29.100.201
 cnf-file location flash:
 max-dn 50
 max-ephones 25
 exit

telephony-service
 create cnf-files
 reset all
 exit
```

CNF files are XML file holding config info for phones like where DHCP, TFTP servers are, what firmware to use, etc. In this case held or generated from Flash on CUCM device.

3 tiers of users: sysadmin, customer admins (to add, move, change), and phone admin (reg. users)
Cust admins use web-based tool

[sidebar on downloading the tool, CME x.x GUI files for IOS 1X.X releases, tar files; bigger zip available for ringtones, bgs, etc. Says he got files from Cisco and www.firewall.cx]
Check IOS version, CU CME version, CME GUI versions for device and download

```
archive ?
  config        Arching the running config
  log           Manage archive log
  tar           List or extract tar image files
archive tar ?
  /create       Create tar
  /table        List files
  /xtract       Extract files
archive tar /xtract tftp://172.29.100.153/cme-gui-8.5.0.tar flash:
```

    [extracts to flash:]

```
config t
(config)# telephony-service
(config-telephony)# web admin ?
 customer     customer admin
 system       system admin
(config-telephony)# web admin system secret ?
```

0 UNENCRYPTED follows
5 ENCRYPTED follows
(config-telephony)# web admin system secret 0 cisco
-- do same with customer
Web server is already set up
ip http secure-server
ip http path flash:/cms-8.5.0-gui

Fixing GNS3 so it is happy using CUCM stuff
All of that was done on R2. We are switching to R1 again.
Fix it so it runs properly in GNS3. In IOS:
clock set 9:34:00 oct 12 2015

```
R1(config)#telephony-service
R1(config-telephony)#max-ephones ?
  <1-500>  Maximum phones to support
R1(config-telephony)#max-ephones 100
R1(config-telephony)#max-dn ?
  <1-1000>  Maximum directory numbers supported
R1(config-telephony)#max-dn 250
R1(config-telephony)#ip source-address ?
  A.B.C.D  Define IP source address
R1(config-telephony)#ip source-address 172.29.100.200 ?
  any-match     Disable strict IP address checking for registration
  port          Define tcp port for Telephony Service/CM FALLBACK
  secondary     Define secondary IP address
  strict-match  Require strict IP address checking for registration
  <cr>
R1(config-telephony)#ip source-address 172.29.100.200
Updating CNF files
```

This is for SCCP/Skinny
telephony-service --turns it on
ip source-address 172.29.100.201 -where to accept phone registration from
---- In CCP, refresh devices so it has new config.

Adding Phones and Users
Go to Configure>Unified Communications>Users, Phones and Extensions>Extensions>
Click "create"

Primary phone number (ephone dn): 1000 (like an extension)
Secondary phone: 512-555-1212
Name: John Doe

Command line:
ephone-dn 1 dual-line <--without dual-line, no hold, conference, etc features.
numeber 1000

label John Doe
huntstop
exit

Go to Configure>Unified Communications>Users, Phones and Extensions>Phones and Users>
Tabs: Phone, User, Mailbox, Phone Settings
Phone tab, click "create"
Phone model: list is populated by CUCM version's list
MAC: you can get off the phone itself
Lines:
Line type: regular, overlay, monitor, call waiting on overlay
(overlay is like monitoring another phone's extension)

User tab: userid, name, display name, password, pin
Phone settings Night service (forward "unstaffed" extensions to this # after hours)
Enable remote worker: connect to another phone system (needs to use codec)
Speed dial definitions

You will have to enable the firmware on the phone
Configure>Unified Communications>Users, Phones and Extensions>Templates and Firmware>
- GNS3 won't let you do anything like that.

```
ephone 1
 mac-address 0013.1B5D.E63D
 type 7960
 auto-line
 exit
ephone 1
 button 1:1
 restart
 exit
ephone-dn 1
 exit
```
(There was more but it wasn't shown)

verify ephone config with sh ephone, sh run

Analog Connections:
Router running CUCM out to:
Fax/modem (legacy device) - use FXS port, PSTN use FXO (foreign exchange office) port, PBX can use either
FXS (foreign exchange station) sends dial tone (so devices getting hooked to FXS port makes sense), FXO receives dial tone
"Ear and Mouth" E&M - receive and transmit - trunk between voice router and PBX system

Originally made to connect PBX systems from different vendors
During gradual migration, extension's calls are transferred from PBX to new phone system
Digital Voice Interfaces - t1, e1, ISDN (TDM) - multiple calls vs single call analog
WIC/VWIC - (voice) wan interface card for those ports (with DSPs on card's board)

Review of Digital Signaling:

CSS Channel Associated Signaling (CAS) aka Robbed Bit Signaling
Steals bits from the voice lines for signal data stuff
Good is all channels are available for data, but user BW is reduced
Common Channel Signaling (CCS) more common
Dedicated data channel (more features like callerID), untouched user BW, loose a channel

Legs, like hops a call makes. POTS call legs and VoIP call legs
Aka dial peers, each leg has out and in, uses different codecs
Voice Activity Detection - detects silence on call to conserve using line, might be on a dial pair

<u>Old router with 2 analog lines directly connected (FXS):</u>
show voice port summary
voice-ports 1/0/0 and 1/0/1 show up
dial-peer voice 1111 ?

```
 mmoip        Multi Media Over IP
 pots         Telephony
 voatm        Voice over ATM
 vofr         Voice over Frame Relay
 voip         VoIP
R1(config)# dial-peer voice 1111 pots
R1(config-dial-peer)# destination-pattern 1111
R1(config-dial-peer)# port 1/0/0
```

(at this point the other phone can dial this one even though it isn't configured yet)
R1(config-dial-peer)# debug voip dialpeer

Here is the debug information when 2 FXS lines (directly old telephones) have one call going to the other:

```
*Mar  1 00:51:53.275: //-1/5872E2358014/DPM/dpMatchPeersCore:
   Calling Number=, Called Number=1, Peer Info Type=DIALPEER_INFO_SPEECH
*Mar  1 00:51:53.279: //-1/5872E2358014/DPM/dpMatchPeersCore:
   Match Rule=DP_MATCH_DEST; Called Number=1
*Mar  1 00:51:53.279: //-1/5872E2358014/DPM/dpMatchPeersCore:
   Result=Partial Matches(1) after DP_MATCH_DEST
*Mar  1 00:51:53.279: //-1/5872E2358014/DPM/dpMatchPeersMoreArg:
   Result=MORE_DIGITS_NEEDED(1)
*Mar  1 00:51:54.003: //-1/5872E2358014/DPM/dpMatchPeersCore:
   Calling Number=, Called Number=12, Peer Info Type=DIALPEER_INFO_SPEECH
*Mar  1 00:51:54.003: //-1/5872E2358014/DPM/dpMatchPeersCore:
   Match Rule=DP_MATCH_DEST; Called Number=12
*Mar  1 00:51:54.003: //-1/5872E2358014/DPM/dpMatchPeersCore:
   Result=Partial Matches(1) after DP_MATCH_DEST
*Mar  1 00:51:54.003: //-1/5872E2358014/DPM/dpMatchPeersMoreArg:
   Result=MORE_DIGITS_NEEDED(1)
*Mar  1 00:51:54.607: //-1/5872E2358014/DPM/dpMatchPeersCore:
   Calling Numbe
NyroGyros#r=, Called Number=123, Peer Info Type=DIALPEER_INFO_SPEECH
*Mar  1 00:51:54.607: //-1/5872E2358014/DPM/dpMatchPeersCore:
   Match Rule=DP_MATCH_DEST; Called Number=123
*Mar  1 00:51:54.607: //-1/5872E2358014/DPM/dpMatchPeersCore:
   Result=Partial Matches(1) after DP_MATCH_DEST
*Mar  1 00:51:54.607: //-1/5872E2358014/DPM/dpMatchPeersCoreArg:
   Result=MORE_DIGITS_NEEDED(1)
*Mar  1 00:51:55.255: //-1/5872E2358014/DPM/dpMatchPeersCore:
   Calling Number=, Called Number=1234, Peer Info Type=DIALPEER_INFO_SPEECH
*Mar  1 00:51:55.255: //-1/5872E2358014/DPM/dpMatchPeersCore:
   Match Rule=DP_MATCH_DEST; Called Number=1234
*Mar  1 00:51:55.255: //-1/5872E2358014/DPM/dpMatchPeersCore:
   Result=Success(0) after DP_MATCH_DEST
*Mar  1 00:51:55.259: //-1/5872E2358014/DPM/dpMatchPeersMoreArg:
   Result=SUCCESS(0)
   List of Matched Outgoing Dial-peer(s):
     1: Dial-peer Tag=111
*Mar  1 00:51:55.263: //-1/5872E2358014/DPM/dpMatchPeersCore:
   Calling Number=, Called Number=1234, Peer Info Type=DIALPEER_INFO_SPEECH
*Mar  1 00:51:55.267: //-1/5872E2358014/DPM/dpMatchPeersCore:
   Match Rule=DP_MATCH_DEST; Called Number=1234
*Mar  1 00:51:55.267: //-1/5872E2358014/DPM/dpMatchPeersCore:
   Result=Success(0) after DP_MATCH_DEST
*Mar  1 00:51:55.267: //-1/5872E2358014/DPM/dpMatchPeersMoreArg:
   Result=SUCCESS(0)
   List of Matched Outgoing Dial-peer(s):
     1: Dial-peer Tag=111
NyroGyros#
```

Note where it says "called number" it is filling the number in gradually (1, then 12, then 123, etc)

R1(config)# dial-peer voice 3153 pots
R1(config-dial-peer)# destination-pattern 2... (sic- three periods)
R1(config-dial-peer)# port 1/0/1
The 3 dots there- demo debug-dialed 2333 and it worked.

<u>VoIP leg between two routers for the analog phones they have</u>
So for a different setup, we have a phone hooked up to each router (in a two router setup) so there is one leg between them, which needs to be a VoIP leg.

R1(config)# dial-peer voice 101 voip
R1(config-dial-peer)# destination-pattern 101
R1(config-dial-peer)# session target
---- you don't use port on this VoIP line- configure sessions, not ports

```
NyroGyros(config-dial-peer)#session target asdf
Incorrect format for Session Target
Must be of the form     ^((loopback:rtp)|(dns:.*)|(ipv4:[0-9]+\.[0-9]+\.[0-9]+\.
[0-9]+(:[0-9]+)?)|(enum:([1-9]|1[0-5]))|(ras)|(sip-server))$
Or                 ^((settlement)|(settlement:[0-0]+))$
```

R1(config-dial-peer)# session target ipv4:10.1.1.1

<u>Dialpeer Wildcards:</u>
. matches any digit and *
always is "process now" and will process preceding digits
[4-9] match any one digit in this range. If you see two digits [24-6] mentally insert comma [2,4-6]
^9 negation - match anything but 9
9t t means any number of any digits- no terminator until it reaches a interdigit timeout (IDT)
Instead of going 9[1-9][1-9][1-9][1-9][1-9][1-9][1-9][1-9][1-9][1-9][1-9]
does terminate and intl calls have more than 11 digits, but is considered "lazy"
+ is more of a flag indicating E.164 number (PSTN)

Go to Configure>Unified Communications>Dial Plans

VoIP
Priority 0-10 (0 highest priority)
If you have more than one ITSP connection, you can prioritize them, do load balancing or to set one as a backup if the other goes down
Remote site: set IP or SIP Trunk

Destination number: 1[2-9]..[2-9]
Items for incoming calls to this dial peer:
Incoming called number, answer address
What address do you want it to answer for "maybe a certain callerID value or something like that"
Shutdown dial peer - just disables.

Protocol - h.323 or SIP (usually you'll want SIP)
Codec - g711 is common- might be default to a Cisco proprietary, which means talking to another systems codec is going to require some DSP action
VAD - Voice Activity Detection - drop silent voice packets for optimization
If router is super-busy, this can malfunction with unexpected results!

Digit manipulation - modify the calling or called party information
If you are coming from ext 1101 - calling party transformation will make something other

than 1101 show up when it is given to the ITSP
Called party transformation means you dial 911 and it gets intercepted and transferred to an internal extension or Dominos Pizza before it gets to the ITSP

POTS
Two tabs- Create Dial Plans, each one of the choices looks like it will bring up a wizard, but it actually takes you to the second tab with the create window (same thing) for Dial Peers.
This is very similar to the one for VoIP, number, priority preference, dest number/ incoming called #/ answer address. Trunk/ Trunk Group. Also has calling restrictions on incoming and outgoing.

Digit manipulation is sort of the same, but adds digit strip option to strip all digits matching wildcards

Implementing "COR" Call Restrictions (under Dial Plans)
In PBX days class restrictions were CoS (class of service) but in Cisco, that conflicts with QoS acronyms so they call it COR

Tale of exposed SIP to "owned" to make intl calls by outsiders
Call restrictions and allowances: creating and delegating groups

```
9011T    <-- International
91[2-9]..[2-9]......    <-- LD
9[2-9]..[2-9]....   <-- local
[2-9]11  <-- local
```

3 groups, 3 permissions: Intl, LD, and local.

Make these in the tab "Outgoing Call Types"

```
dial-peer cor custom
 name LOCAL
 exit

dial-peer cor custom
 name LD
 exit

dial-peer cor custom
 name INTL
 exit
```

Switch to the Permissions tab add:
LobbyAccess
And you get to select from the call types just set to add to this permission name (local but no ld or intl)

dial-peer cor list LobbyAccess
 member LOCAL
 exit

dial-peer cor list EXEC
 member LOCAL
 member LD
 member INTL
 exit

But guess what, it turns out that if you want to just apply the permission name and thats it, you'd still have to make a Permissions entry for it first, so:
So you can think of these as wrappers. They will then show up in a list of restrictions you can apply to VoIP or POTS dial peers in the previous area for that
dial-peer cor list Local
 member LOCAL
 exit

dial-peer cor list LD
 member LD
 exit

dial-peer cor list INTL
 member INTL
 exit

So the cor list needs to be applied to each dial peer, or they'll be TOTALLY unrestricted and can dial anywhere!

Dial peers under dial plans, apply restriction allowance you want to OUTGOING (LD for LD-ok items)
On Extensions (Under Phones, Users) you want to apply it to INCOMING permissions.

ephone-dn 1 dual-line
 corlist incoming LD
 exit

Create directories - this is done in Phone User when you add first and last names in.
Here is how it goes into the command line:
ephone 1
 username jimmye
 exit
ephone-dn 1
 name Jimmy Easton
 exit

Under "Telephony Features" there is also "Directory Entries" where you can also put in

names
This is a bit different since it isn't really attached to the phone

```
telephony-service
 directory entry 1 23456 name Corp Fax
 exit
```

In General Telephony Settings there is how names in the directory are sorted (First name first, etc)

Forwarding users -
Unity Express module

```
telephony-service
 voicemail 5555

ephone 1
 call-forward {all I busy I etc} 5555
```

CCP under Extensions>Edit> has the form
Fun trick - to thwart if someone grabs a phone, hits CForward and puts in an Intl number to get free calls to it, under Restrictions for extension is "call forward max length" you can set for the number of max digits.
COR list is a better way though
Also consider people forwarding their extension to their mobile- the call then comes in from the PSTN to the internal phone system and back out, eating up 2 lines - trunks to pay for - everyone does this now so it's accepted/normal

Avoiding "Hairpinning"
Say your company has 3 offices each with CME going through MPLS
One is in Texas, Another in Japan, and another in California.
John in Japan (1xxx) is out on vacation and forwards phone to Jill in California (2xxx).
Bill (3xxx) tries to call from Texas, and his call goes to Japan, then California, and QoS gets meaningless.
Here is a way for John's phone to send a redirect instead to the right IP, using H.450.3/SIP redirect:

```
telephony-service
 call-forward pattern 1..
 call-forward pattern 2..
 call-forward pattern 3..

telephony-service
 transfer-system { full blind I full-consult I local-consult }
```

 - full blind is good for phones with a single line - it's the only way they can transfer!
 - full- and local- consult use H.450 to keep from hairpinning (described above) local-consult is Cisco only so all phones need to be Cisco for it to work.

Transfer patterns in "General Telephony" settings
(to transfer to outside lines)

Add transfer pattern like 9........... or 91..........
telephony-service
 transfer-pattern 91..........

BUT - Evil transfer trick
Call receptionist and ask to be transferred to ext 9011 - this is 9 (dial out) and 011+ international calling - so all the caller has to do is put in the international number to make the call
Use 9[2-9] instead

Call Park (putting it in the call center-style queue)
You need park ephone-dns to park calls in:
ephone-dn 40
 number 9998
 name ParkMe
 park-slot

Park-slot without options can leave caller forgotten!
 - directed - park is directed to specific number (Sales)
 - reservation-group
 - reserved-for
 - timeout <seconds> when person who parked it is reminded. Add limit to how many timeout cycles they get to go through before they are unceremoniously DISCONNECTED!

Telephony Features also has call park but this is a bigger op for multiple parking provisioning
It uses a recall item next to CLI "limit" directive to not hang up but be sent straight to person who parked.

Configuring pickup
- when you have multiple groups answering the same "line" like a help desk, can be a pain if not set up right- be sure to not used "directed" since it causes user confusion

Intercom option between two phones

Cisco Unified Communications Manager
- Core Cisco Unified CM Administration Interface (CCM-ADMIN)
        http://ipaddy/ccmadmin/
user: ccmadmin and pass what you set
TONS of setting and areas.
Some settings areas need a separate login, like Cisco Unified Operating

SEE ACCOMPANYING SCREENSHOTS

Core Administration Menus:
System - Global Settings

- Access to CUC Managers and servers - two different items to manage servers
- Access to Cisco Unified CM Groups
- Device Pool, Device Mobility
- DHCP and LDAP, Single Sign On
- Security, Enterprise and Service Parameters
    enterprise parameters allows setting autoreg proto to SIP or skinny/sccp
- SRST

Call Routing
    Dial Rules
    SIP Route Pattern
    Dial Plan Installer
    Dial Plan Replication
    Park, Intercom controls
Media Resources
    Hold Music
    Transcoding
    Mobile Voice Access
"Advanced Features" - name doesn't fit
    Voice Mail
    Directory Lookup and Sync
Device - "this is where you will probably live"
    CTI Route Point
    Gatekeeper
    Gateway
    Phone
    Trunk
    Remote Destination
    Device Settings
Applications
    CUCM Assistant Config Wizard
    Plugins
User Management (sic)
    You have users which are members of groups which have different privileges
    Many pre-made groups
    Go to End user, (click add new, put in basic basic info)
    Go to Usr mgmt>Usr Settings> Access Control Group
    Add new name Tier1, add user to it, go back to list where you'll see new group listed
    Add Roles to the new group, by choosing from all the existing groups provided
    (roles have task privileges assigned, read and update allowed (or not)

"Application user" refers to a certain set of general minor administration tasks/roles you can assign more quickly. App users are one level above end users, who don't get to administer anything- some are what might be doled to an office manager for example

Defined users are added to phones in user-device association, have both password and PIN User Mgmnt> Credential policy lets us place restrictions on

passwords (max 8 chars e.g.). you can have several policies if you want
manual end user creation: using the CUCM GUI one by one. Bat is batch- bulk administration.. find users based on search criteria and update those particular user profiles identically in bulk. You can also define a user template to use when adding a bunch of new users as batch
The system uses a CSV file and has Excel macro compatibility. LDAP is the last way to manage users
LDAP synchro brings in all LDAP users to use. LDAP auth is a different item for SSO- relies on LDAP or AD uptime of course.See system> LDAP> LDAP system> and LDAP directory>

Also in user management>user/phone add is universal device template settings for the CUCM autoreg setting (set device pool, security and profiles, etc) - you can make a profile for yet-to-be-configured phones (that need extensions fixed, other things) and have them with limited call abilities until setup is complete

Bulk Administration

Example- you have 1000 phone and need to change the same thing on all of them

Main Navigation Menu

Cisco Unified Reporting

Cisco Unified CM Administration

Disaster Recovery System (DRS)- SFTP backup only!

Cisco Unified Serviceability - sortof a system control panel for the Admin tools interface

Tools> Service Activation - select server - CM and TFTP min req for phones

Tools> Control Center- Feature Services - select a server, a svc, and tell to restart the svc

Cisco Unified OS Administration

- will ask for another user/password
- meant to not be given to day to day admins

Cisco Unified IM and Presence Reporting

Cisco Unified Communications Manager
CUCM cmd line - simple menu-like text interface
prompt is "admin: "
? gives options
useful sections: system, utils

Then there is a separate
Cisco Unity Connection Administration - uses the same "OS" as CUCM (above)
Same main navigation choices as Cisco Unified Connections Manager

Sidebar Menu is a tree format with little boxes with a plus sign in them to collapse/expand
Users>
Class of Service
Templates
Contacts
Distribution Lists
Call Management
Message Storage
Networking
Unified Messaging
Video
Dial Plan
System Settings
Telephony Integrations
Tools

Device > Phone
Enter device protocol and select phone model from pulldown; a form shows up below to put in info
Many options, what do you need?
Many have asterisks, will bug you if you left anything undone
MAC Addy, description, device profile

** some parts of this interface, clicking save once just puts a note at the top of the screen and you have to hit SAVE AGAIN. Watch out.
-- When you click save, it presents the info with a sidebar called Association with hyperlinks including add new directory number- to modify the phone's button items
-- When done with everything- hit "Apply changes" and the phone setup should be made available
Auto-registration - when phone queries CUCM Publisher the first time and it doesn't match the MAC up, the phone will get an xmldefault to run off of (just to get to the server) - it will dole out a default and an extension in a range (maybe not the one you wanted) but it registers the phone in CUCM so you don't have to fill out all the info in the CUCM Admin

Note- Cisco IP Communicator is the desktop app for Windows which masquerades as a real phone

In Unified CM Administration, CM Group Administration is where you set up CM Groups which are your CM servers (where the 3 for each phone come from)

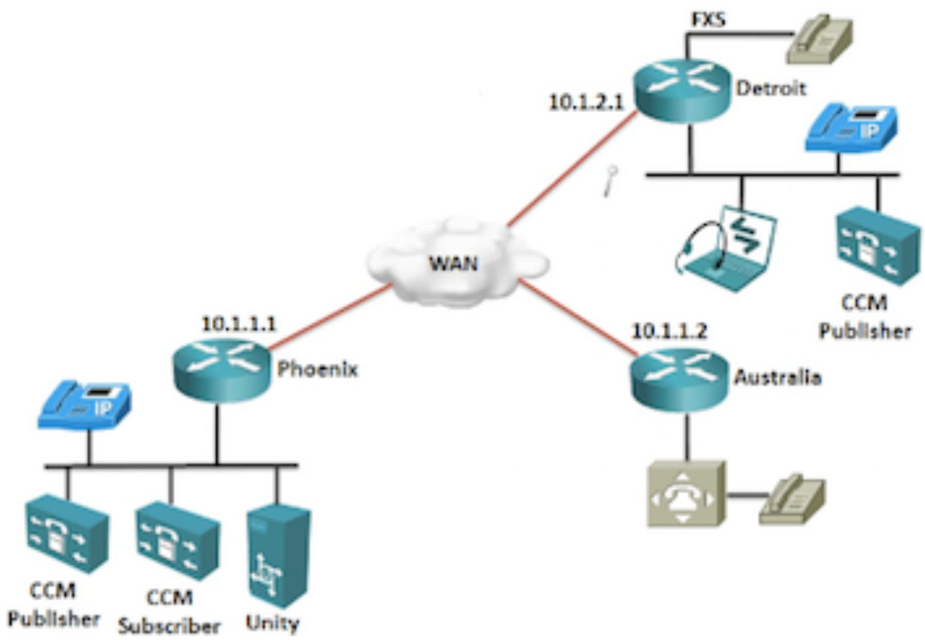
Dial Plan Architecture - Call Processing Models
<u>Single Site - single campus network</u>
- switch connects 3 call managers - primary, secondary, but also load balancing
- switch connects router to PSTN (T1, FXO, SIP Trunk) (e.g. Sprint SIP trunk w/ QoS

10mbps)
- switch connected to phones
- plain 711 uncompressed audio codec - convo 64k with other data for it about 80kbps
SIMPLE.

Central Call Processing
Think of Single Site arch adding a remote branch
Two single sites with central cluster CUCMs at one site, SRST on remote router (Call Mgr Express)
Routers at each site with both PSTN and WAN connections
(PSTN is good for specificity of emergency services, WAN primarily for CUCM comms)
G.729 compression between sites (8kbps)

Distributed Call Processing <---the way to go!
(after a Central Call Processing setup gets several hundred phones at the branch)
Each site it's own CUCM cluster - 2 different DBs, different extensions, etc, no replication
- set up intercluster trunk between CUCM clusters
- one goal- WAN primary comms- better comp, cheaper
- Each side needs independent WAN BW control (H.323 gatekeeper or SIP proxy)

Clustering Over a WAN <-- too needy!
You can have the cluster spread over both sites - with DB replication, single point of admin (failure!)
Higher BW requirements:
- Round trip delay 40ms to DB synchronize usually kills the idea
PLUS VoIP recommended less than 150ms one-way delay (2 way twice as much!)
For this, sites close together, Metro EN if possible ~ 900Kbps per 10,000 BHCA



Call Route Plans -what CUCM knows and what must be configured

Australia - a phase 1 VoIP setup
 - 2xx numbers - has PBX system, bought voice ints for gateway (T1,vWIC cards)
Detroit - 3xx - it's own cluster (Publisher is a giveaway)
Phoenix - 1xx - all it's stuff is in the local CUCM and Unity DB

These know about their local stuff but need route plans to talk to the others

Route Pattern --- 2xx
Route List --prioritized list of Route Grps for those patterns
Route Group1 Route Group2 (Devices 3 and 4) --lists of actual devices that help reach destinations
Device1 --- Device2

Think object-oriented telephony
 - Route Grp as an object
"PSTN devices"
"WAN Devices to AUS" 10.2.9.107
 - Route List as a prioritized list of groups

dial 201- matches 2xx - checks route list:
 - the #1 choice for routing 2xx is "WAN Devices to AUS"
(that one is down)
 - the #2 choice for routing 2xx is "PSTN devices"

[for PSTN, 201 has to be looked up in the transforms before given to PSTN]
201 transformations say DID to 55 2222 2234 (or something)

Creation goes from the bottom up- create devices first, then combine them into group, add the group to the route list and take my route pattern to tie them all together

Partitions and Calling Search Spaces (CSS)
Partitions - groups of dialable numbers (lines, route patterns, anything with a number)
CSS- a list of reachable partitions- assigned to a "dialing" entity, defines calling privileges
    CSS can not only be applied to phones, but also gateways, etc.
NORM_CSS - internal-pt, local-pt, [none]
MGMT_CSS - internal-pt, local-pt, [none]
    ("none" is at the bottom of any CSS)
Sample partitions:
- local 9.[2-9]xxxxxx
- ld 9.1[2-9]xxxxxx
- internal 1101, 1102 (extensions)

When you install CM,
- there is a partition called "none" containing all numbers
- there is a CSS called "none" which is assigned to all devices

In CUCM - in Directory Number Information, where it says Route Partition: <none> it doesn't mean not assigned- it really means this "none" partition! It is reachable by ever single CCS

If you start fresh, create the Local partition, it is assigning the permissions to call those numbers from the "none" partition to Local-pt. Phones won't be able to make local calls unless assigned that partition in their CSS (and the "none" CSS points to the "none" partition, so that means everyone)
THIS MEANS,
1) MAKE PARTITIONS FIRST (PLACEHOLDERS) - THEN ASSIGN #s TO THEM LATER
2) MAKE CALLING SEARCH SPACES, ASSIGN EMPTY PARTITIONS
3) ASSIGN CSS TO PHONES/ DEVICES
    - they will still have empty partitions, but will all contain "none" - nobody gets disconnected.
4) ASSIGN #'s TO PARTITIONS
    - (and that is also why "none" is at the bottom of all CSS lists)

Cisco Unified Mobility - not hard to set up, just lots of steps
Cisco Mobility Connect (aka Single Number Reach)
    - this is where users can have their phone also ring their home and mobile phones
    - Eats up trunk bandwidth- call comes in T1, more go out to ring the other phones
Cisco Unified Mobile Access (MVA)
    - Hide your mobile # rather than give it out
    - dial into enterprise CUCM system, check VM and stuff. Dial out from there, using your mobile

- CID is the corporate network's- not your mobile #

Mobility softkey allows
- hitting the key on desk phone to switch call to cellphone (and back)
- switching the office call you got on your mobile (through MVA) from mobile to desk phone

1. Add mobility softkey
   - CUCMAdmin>Device>Device Settings>Softkey Template
   - New/add, name it. Hit save and more options show up.
   - Under "related links" in the top right, you should see "configure softkey layout", click "go"
   - Template should be "mobility users".
   - Under call states "On Hook" and "Connected", add "Mobility (Mobility)"
   - (you have to hit "save" each one you add above, individually)

2. Configure end user
   - CUCMAdmin>User Mgmt>End User
   - Enter user, scroll to bottom of page Mobility Info" section
   - Check the two boxes: Enable Mobility (Connect) and Mobile Voice Access
   - Make sure Primary User Device is the main desk phone
   - Max wait time: in milliseconds, default 10000 - pickup timeout after softkey-switched to desk
   - Remote destination limit. Default is an absurd 4. 2 is better. Number of phones to forward rings to.
     - Higher the number, the more trunks it will use when phone rings.
     - Set remote destination profile later when configured (step 4)

3. Configure IP phone
   - CUCMAdmin>Device> Phone (find/ list)
   - Softkey Template - switch to "Mobility Users"
   - Owner user ID - set to your username from pulldown

4. Add remote destination profile RDP (w/ a shared line)
   - CUCMAdmin>Device>Device Settings>Remote Destination Profile
   - New, name it, enter username, device pool (same as route group?)
   - "think of as a phantom phone that rings the remote destinations"
   - Since this is seen as a device of sorts, it also gets a Calling Search Space (calls limits)
   - Also set rerouting CSS
   - Hit save. It sends you to the summary page with sidebar "Association information"
   - The link allows you to add a directory entry- do so and match the desk phone extension
   - You could add more extensions if you WANTED by repeating this step.
   - Add associated devices with the Remote Destination Profile name listed

5. Add remote destinations

- CUCMAdmin>Device>Remote Destination
  - This is where you ender the mobile phones, or other phones and devices.
  - End users can do this from their own configuration page but YOU get to train them on how to do it!
  - Remember you are entering the number as dialed from on premise so, 95125551212
  - Answer too soon timer - milliseconds again. Set so mobile (if off) can't just fire into it's VM
  - Answer too late timer - to keep mobile from ringing UNTIL it goes to it's VM
  - Delay before ringing - default 4000 (4 sec) before your cell rings in case you are at your desk
  - Mobility, and Mobile Connect checkboxes
  - Schedule times for this to be in effect
  - Ring this phone only if from, etc - access lists - set in next step, add them to the line here

6. Add access lists (optional)
  - CUCMAdmin>Call Control>Access List
  - Add new, name it, set owner, then allowed or blocked
  - Save

7. In Cisco Unified Serviceability
  - Service activation, bring up server, make sure Mobile Voice Access Service is on

8. Configure service parameters
  - CUCMAdmin>Service Parameters
  - Toward bottom of the page Mobile Voice Access number (DID)
  - Matching Caller ID with remote dest.: complete or partial match to number
  - Number of digits CID match- matches this many digits from the right (end) 7 would be 5551212
    - good in case system can't do with 9-area code

9. Mobile Voice Access
  - CUCMAdmin>Media Resources>Mobile Voice Access
  - Enter internal directory number
  - Region etc.

Done

ephone for sip

IP Phone Endpoints -

http://www.cisco.com/c/en/us/products/collaboration-endpoints/ip-phones/index.html
See also PDF of Endpoint Matrix- has phones and all Collab products
(is what you get on the above page when you click on "compare models")

Cisco IP Phone 8800 Series - the video phone solution
http://www.cisco.com/c/en/us/products/collaboration-endpoints/unified-ip-phone-8800-series/index.html

8900 and 9900 feature video

Cisco Unified IP Phone 7900 Series - Standard ACC campus phones
http://www.cisco.com/c/en/us/products/collaboration-endpoints/unified-ip-phone-7900-series/index.html

Cisco IP Phone 7800 Series - These are a little more primitive than 7900s, no color screen
http://www.cisco.com/c/en/us/products/collaboration-endpoints/unified-ip-phone-7800-series/index.html

Cisco Unified IP Phone 6900 Series
- http://www.cisco.com/c/en/us/products/collaboration-endpoints/unified-ip-phone-6900-series/index.html#

Cisco Unified SIP Phone 3900 Series - the 3905 is the only phone in this one
http://www.cisco.com/c/en/us/products/collaboration-endpoints/unified-sip-phone-3900-series/index.html

Cisco Small Business SPA500 Series IP Phones- the 500s look a lot like the 7800s but have a green center button, pads for 10+ extension buttons are modular add-ons.
Cisco Small Business SPA300 Series IP Phones- 2 models, made for people that want something that looks more like "just a phone" instead. The 3900 for the SB line.

The phones at ACC were generally 7900's

Troubleshooting Phone Connectivity
PoE - check cables/ pins; show power inline/ sh run; is the std right/ matching?
- 15.4W is the max power PoE delivers to a device (you might see it with power-hungry devices
- Normal is, for regular 79xx phones 6-7W, DXxxx video phones ~14W
- cdp can report usage need, but doesn't in cmd outputs
- power inline < auto I never I consumption I static > static makes high pri, consumption sets limits
- sh interface/ sh run should say if one is applied

Voice VLAN
- switchport voice vlan < dot1p I none I untagged I # >
- untagged is untagged on PVID, dot1p is pri tagged, # to specify vlan number
- SHOULD be tagged
- check ip/ DNS
- sh ip int x/x/x vlan
- switchport trunk allowed vlan 2
- is the phone itself seeing it?

Verify DHCP
  - check option 66/ 150 - some (mostly non-Cisco) phones only do 66 (DNS)
  - check DNS
  - check helper IP
  - check IP phone settings

Checking CME/ CUCM - check that phone shows up and autoreg is up

Troubleshooting Route Plan and Reporting
  - Symptoms: can't make call; fast busy signal; cutting off halfway through dialing

Verifying Router-Based Dial Plans
  - show dial-peer voice summary

```
Router#show dial-peer voice summary
dial-peer hunt 0
             AD                                   PRE PASS                     OUT
TAG    TYPE  MIN  OPER PREFIX    DEST-PATTERN     FER THRU SESS-TARGET         STAT PORT
1      pots  up   up             3456             0                            up   1/0/0
2      voip  up   up             ....             0   syst ipv4:172.29.100.201
9      voip  up   up             9[2-9].........  0   syst ipv4:24.51.113.21
10     voip  up   up             91[2-9]......... 0   syst ipv4:24.51.113.21
911    pots  up   up             911              0                            up   1/0/1
```

  - The "...." - this pattern is limited to 4 digits, and these are listed in order of matching priority
  - "debug voip dial-peer" shows how. It let the phone dial 4 digits and cut off there.

```
*Mar  1 00:18:35.763: //-1/AEA13A838005/DPM/dpMatchPeersCore:
   Match Rule=DP_MATCH_DEST; Called Number=3333
*Mar  1 00:18:35.763: //-1/AEA13A838005/DPM/dpMatchPeersCore:
   Result=Success(0) after DP_MATCH_DEST
*Mar  1 00:18:35.763: //-1/AEA13A838005/DPM/dpMatchPeersMoreArg:
   Result=SUCCESS(0)
   List of Matched Outgoing Dial-peer(s):
     1: Dial-peer Tag=2
```

  - If you dialed 3456, it would match both 1 and 2, matching 1 because it is more specific
  - When CUCM gets numbers, it gets the digits collected, then sends along at once

CUCM Route Plan Report (Unassigned Directory Numbers)
  - CUCM>Call Routing>Route Plan Report
  - you can filter the route plan by several criteria, one being "unassigned DNs"
  - These are numbers in the directory from old (now non-existing) phones - "orphaned"

--------------- This (below is ireelevant to unassigned directory numbers!!
  - CUCM>Device>Gateway <--was just used to add a gateway to send a Route Plan to
  - CUCM>Call Routing>Route/Hunt>Route Pattern
        Quick way to make route pattern without going through entire route pattern/ list/ group hierarchy
          - Name it, add pattern 9.1[2-9]xxxxxxxxxx (sic- 10 "x" digits), add router/gateway from list
          - Under called Party Transformations, put in Discard Digits "Predot" (to strip off

the 9)

- This will now show up at the bottom of the Route Plan Report

CDR Analysis and Reporting
Go to Cisco Unified Serviceablity
Tools> Service Activation
Select server - your CUCM server IP
- points out the Cisco CAR Web Service
- it's the service that generates call detail records
  - records to find out who dialed what- Like 911, or international calls
  - CDRonDemand generates to export for billing or similar services

You may have it turned on in Unified Serviceability, but in go to
CUCM>System>Service Parameter Configuration
Select your server, Service: Cisco CallManager
This lists all the parameter values
CDRenabled flag needs to be set as "True"
Cisco Unified Reporting doesn't give you all the info
- use Cisco Unified CM CDR Analysis and Reporting

To get to CAR, it doesn't give you a link (!)
You have to enter the url of the server and specify /car
On opening, it may complain the CDR isn't on in the node "sub"
--- This is one thing written to both publisher and subscriber servers! (CDR info)
May also complain about the flag "Call Diagnostics Enabled" being off.
This is how you check QoS logs if people complain about bad QoS, that way you can check load etc. causing such problems

RTMT- RealTime Monitoring Tool
CUCM Admin>Application>PlugIns>
- this section gives extra tools. This one downloads and installs on Windows or Linux
CPU, VM, Disk Usage, IM use, etc etc.. all the expected performance monitoring
SIP activity, registered phones, other sensors, SQL stats, etc.

For voice VLAN, port on switch is an access port, not a trunk.
Cisco phones are able to put VLAN tags on and use untagged frames for the attached workstation
CUCM obviously lives on the same VLAN as voice traffic (say VLAN 10)

IOS
show power inline -- for power info
show cdp neighbors --will also show phones, models
show cdp neighbors detail
show lldp neighbors -- says Cisco is switching to LLDP from CDP (?)
show cdp detail <device>
int fastEthernet 0/0
switchport access vlan 20

switchport voice vlan 10
show vlan
show interfaces fastEthernet 0/0 switchport

Powerup for first time:
1. Switch supplies power, phone boots SW image/ firmware
2 Switch provides VLAN info to phone via CDP. In SIP this is CTL (certificate-based to authenticate legitimate servers and devices)
3. Phone sends DHCP request, where it gets tftp info too (option 66 is DNS, option 150 is IP)
4. Phone gets config file from tftp server (cnf.xml with 3 CUCM servers, right FW vers. and locale)
5. Phone contacts first CUCM server listed in the config (CUCM Group), registers, gets softkey template - establishes SCCP/skinny connection
 With SIP, step 5 downloads dial rules and stuff from tftp

SCCP is like a dumb terminal, SIP is more independent

SCCP/skinny does everything through CUCM except the RTP/RTCP traffic which is endpoint to endpoint. SIP is replacing SCCP/skinny

Regular boot first time: switch detects with CDP, negotiates power needs. Switch gives voice VLAN, phone restarts into right VLAN, does the rest above. Dailtone can happen when CUCM is resolved.

Hard Reset
Unplug ethernet, plug back in holding # key down until speaker light comes on.
On keypad type 123456789*0# (all the keys on keypad in order)

Phone factory reset * * # * *

Note: one voice packet contains 20msec
If one drops it fills the space between by stretching the neighboring packets's contents

Audio vs Video requirements:
<30ms jitter
<150ms delay
<1% loss

c1700
c2600
c2691
c3620
c3640
c3660

c3725
c3745
c7200

Unity Connection Overview
- Voicemail, Auto-attendant (press 1 for...) aka TUI, Voice Recognition (VUI)
- 20,000 mailboxes per CUC server (that's if you templated it as a mail server and nothing else)
- Integrated IMAP server + Exchange integration
- Web-based voicemail access
- Integrates with traditional telephony PBX to handle it's VM
- Visual voicemail (Phone View) with Jabber or IP phones

Single-site or multiple-site considerations
Ask if the other sites need their own server or not

Remember that Unity and CUCM are separate servers, integrated with SIP or SCCP
CUCM- set up a VM profile - the voice mail "pilot" (extension) goes to the VM system

SCCP/skinny - Pilot number goes to a hunt or line group extension (another number) that has VM lines (VM ports, which you would have a set limit in licensing) if a line/hunt extension is "busy" it hops to the next

SIP creates a route pattern (a pilot again) that points to a SIP trunk. The SIP trunk needs an IP address and you give it Untiy's IP

Unity Connection Administration
Need user templates before making users to assign them to
Go to Templates>User Templates> click 'new'
- Edit menu has stuff to assign
  - Roles, password settings, transfer rules, message settings,phone menu, accounts, etc etc.

When you go to Users>Users, and Add new, you get a new user window, and get to choose from the builtin templates
- User type = with or without mailbox
- based on template:

Remember PBX CoS (class of service)? Outside of this it is handled by COR - It is also here (timers, features, and restrictions) for VM characteristics

Unity Connection - user templates
    user creation options - manual, BAT (CSV) bulk administration)
    Merge from CUCM with AXL
    Merge from LDAP (AD)

IMP
- extend basic CUCM (display status: on/off hook, unknown)

- internal and external IM
- user status and present contact methods
- protocols to connect with many clients (SIP, XMPP)

Jabber has built-in phone so you can plug in headset, AV conf, IM, VM control and comms hist
- user authentication: SOAP (Simple Object Access Protocol) over https
- hooks to Unity with XMPP (eXtensible Messaging and Presence Protocol) - reports status of users
- Svc delivers list of devices (CCMC-IP) - is it softphone or desk phone? see what it has control over
- CTIQBE Comp-Telephony Integ Quick Buffering and Encoding send cmds to IP phone like off-hook and synchronization between softphone and desk phone
- CAST Cisco Audio Session Tunnel (keep laptop video and phone audio in sync)

Is CAST "hello WebEx troubles?"

Unity - Exchange (OWA/WEBDAV) and AD/LDAP, but also MS Live Communications Server integration (SIP does it)

Jabber links to CM-IMP which links to CUCM but if Jabber client needs to search the user DB it can go straight to the LDAP server or over to the Unity sever with IMAP
iPhone calendar hooked up to Exchange can tell Jabber "in a meeting" so the whole world knows
The logs of all the communications are in PostgreSQL (CDR/CAR too?)

Setting up IM and Presence:
Lots of this is integrated now, so it is actually simpler
<u>Step 1: Configure CUCM end user, phone, associate user to directory number</u>
- Go to CUCM Admin>User Management>End User
- Add username, PIN, password and last name, hit save to make all options appear.
- Scroll down to "Service Settings and click checkbox labeled:
    - Enable user for Unified CM IM and P (Configure IM/P in the associated UC Service Profile)
    - Also present is a checkbox for Exchange integration
- Below is Device Info. Add desk phone to "Controlled Devices" device association (here on ext 2101)
- Scroll to Extension Mobility, and make sure "Allow Control of Device from CTI" is checked
- Scroll to Permissions Information" at the bottom, add Access Control Group: Standard CTI Enabled
- Go to Device>Phones and find the phone we added (ext 2101, 7900 series phone)
    - Make sure Allow Control of Device from CTI is checked, hit save
    - In the sidebar on the left under associations, pick the line
    - At the bottom of the Directory # Confirmation, click Associate End Users, pick user
    - Hit apply configuration, and then save

REMEMBER- YOU MAY HAVE TO HIT SAVE TWICE IN DIRECTORY NUMBER

SCREENS!

Step 2: Create a Cisco Unified Client Services Framework CSF device
- Go to Device>Phones and click add.
    - Under "phone type" choose "Cisco Unified Client Services Framework"
    - This will be used to represent Jabber client, and communicate through it
    - Owner User ID needs to be set to the original user's username.
    - Give it a similar username to our user, like username_jabber
    - Device Pool: default; Phone button framework: Standard Cisco Services Framework;
      SIP profile: Standard SIP
    - Save and apply. In sidebar on left, "associations", pick the line, give same ext as user (2101)

REMEMBER- YOU MAY HAVE TO HIT SAVE TWICE IN DIRECTORY NUMBER SCREENS!

Step 3: Associate the CSF to the user
- Go to CUCM Admin>User Management>End User
    - Device Info. Add Jabber client proxy "username_jabber" to "Controlled Devices" device assoc
    - Save it
- FIRST We are going to make a template for all these sevices
- Go CUCM Admin>User Management>User Settings>UC Service, then click find to find UC services
    - JabberClient and JabberClientDirectory are in by default. Click "Add New"
    - This lets you add a UC service of type: "Voicemail, MailStore, Conferencing, Directory, IM/P, CTI, and Video Conference Scheduling Portal"
    - Add CTI, name it JabberCTI, give it the IP of the server running the CTI service, apply, save
    - *This page also tells you the TCP port used 2748 for this service (and other services)*
    - Add IM and Presence, do the same, VoiceMail, product:Unity, name UnityServer.
- Go CUCM Admin>User Management>User Settings>Service Profile
    - This will be a profile-template using the stuff we just set up that we can add to every user
    - Click find with field blank, and it finds JabberClient, open it
    - Mail Client, select UnityServer; Directory, JabberDirectory; CTI, IMP
    - at the end click checkbox "Make this default service profile for the system"

- Go to CUCM Admin>User Management>End User
- Go back to user's main account username page
- Scroll down to "Service Settings and click checkbox labeled:
    - Enable user for Unified CM IM and P (Configure IM/P in the associated UC Service Profile)
    - Make sure that is checked!
    - Choose the new system default JabberClient, that we fixed up.

- Here is also a checkbox where you can enable/disable it to snoop in Exchange calendar for "meeting info"

- click save apply/save etc and the user is set up

Step 4: Set up IMP configuration

The connection between CUCM and Jabber needs to be set up

- Go CUCM Admin>System>Security>SIP Trunk Security Profile, then click find to for existing profiles
    - Add new, call it IMP_New
    - Click checkboxes to accept: presence subscription, out-of-dialog, unsolicited notification, and replaces header
    - Save
- Go CUCM Admin>Device>Trunk. Add new.
    - Type SIP Trunk, Service type: none, hit NEXT
    - Name SIPTrunkToIMP
    - Device pool: default. Scroll down to SIP Trunk Security Profile: IMP_New (we just made)
    - If it isn't integrated, you have to set SIP trunk destination.
    - SAVE and RESET

- Log into the Cisco Unified CM IM and Presence Administration system

  (it looks just like the CUCM and Unity Admin screens)
    - Go to Presence>Settings>Standard configuration
    - Go down to CUCM IM and Presence Publish Trucnk. Select the SIPTrunkToIMP. Save it.
    - (notice how it got that from the CUCM database automatically for us?)
    - Go to Presence>Gateway Say New

Type is CUCM, Publisher, Presence Gateway: <ip-address> [[you could also make another and instead of "Publisher" select "Exchange" to hook up an Exchange server]]