These notes cover the current 640-461 (ICOMM v8) examination to complete the CCNA Voice certification. {extracted from kccvoip training}

The following notes may help narrow the study topics to the relevant areas. The 'Study Summary' for each section highlights the main items covered by the examination. *** This information is not supported or endorsed by Cisco Systems, Inc. ***

**NOTE** the actual Vue/Pearson examinations **DO NOT ALLOW** you to go back and change or mark any questions, as many other demo and training examinations do. **AND** not all questions are multiple choice and may require you to fill in the blank, drag & drop responses, telnet simulation or input your response to a diagrammatic exhibit (very few simulation questions as compared to the CCENT/CCNA exams). Unfortunately, there are many questions on the use of the web/GUI to configure the CUCM/CME and are now mainly non-technical questions regarding administration 😐 We do try to fit in (when time permits) content from the original 640-460 course and items from CCNP-Voice in order to keep the engineers interest and provide a better understanding of VoIP in the real-world.

see also  **CCNA Voice 640-641 example questions**,  **IOS Configuration Examples**,  **CCNA Voice 640-640 example questions**

The Vue / Pearson tests can be booked online via [www.vue.com/cisco](http://www.vue.com/cisco) (since Cisco changed from Thomson/Prometric to Vue/Pearson there seems to be less testing centers available outside of the US, so check on their web site for centers and schedules in your area)
PSTN and Legacy Telephony Study Summary

**REMEMBER**

Loop Start vs **Ground Start** and that LOOP START is susceptible to **GLARE**

**Supervisory Signaling**

- On-hook signal
- Off-hook signal
- Ringing

**Address Signaling**

- DTMF
- Pulse

**Informational Signaling** (all the rest)

- Dial Tone
- Ringback
- Busy
- Congestion
- Reorder
- Receiver off-hook
- No such number
- Confirmation

**Analogue to Digital Conversion**

(human hearing range 20-20kHz at very best, human speech 200-9kHz, PSTN uses 300-3400Hz)

- Nyquist theorem (300Hz – 4,000Hz, sample @ 2 x highest frequency)
- Sample (PAM)
- Quantize & Encode (PCM)
- Compress (CODEC)

**Time Division Multiplexing (TDM)** time slots for each channel

- T1 has 24 x 64kbps channels mainly in the Americas and Japan
- E1 has 30 x 64kbps channels mainly in EU

**CAS = Channel Associated Signaling** = uses the same bandwidth/channel as the voice or data. **T1 using CAS will steal bits (every eighth bit of every sixth DS0 time slot) often called robbed bit signaling (RBS).** An ESF
T1 frame is sent as 24 DS0 frames (8 bits x 24) PLUS a single framing bit = **193 bits**. The older SF (super frame) standard T1 frame was sent as 12 DS0 frames with framing bits on each of the DS0.

**E1 CAS does not use RBS but has DS0:1 for framing and DS0:17 for voice signaling not really CAS, but considered so because it uses the same format of ABCD bits as T1 and can therefore be compatible and interconnected.**

T1 and E1 will use loop start, ground start or E&M (1-5) for the interface connections and signaling  
(NOTE: Cisco do not support E&M type 4).

**REMEMBER CAS T1** *(rbs=8th bit every 6th frame), 24th timeslot (chan 23) for signaling,*  **CAS E1** *(framing on 1st timeslot =chan 0), 17th timeslot (chan 16) for signaling,* configure for PSTN trunks using ‘ ds0-group # timeslots x-x ’

**CCS** = **Common Channel Signaling** = ‘complete channel for signaling’ uses a separate, dedicated channel (OUT OF BAND SIGNALING). **T1 dedicates 24th time slot to signaling.**  **E1 dedicates 17th time slot to signaling.**  Most popular signaling is Q.931 and SS7.

**REMEMBER CCS T1**  24th timeslot (chan 23) for signaling, configure for PRI trunks using ‘ pri-group # timeslots x-x ’ and don’t forget the global config to set the PRI ISDN switch type.  **CCS E1**  17th timeslot (chan 16) for signaling, configure for PRI trunks using ‘ pri-group # timeslots x-x ’ and don’t forget the global config to set the PRI ISDN switch type.

**REMEMBER THE PSTN TERMS**  Analogue telephone/fax/modem, local loop, central office (CO), trunk, private switch (PBX/PABX), digital telephone, ip telephone, numbering plans such as ITU e.164 (15 digit max)

**PBX FLAVORS**  Key Systems *(PSTN to extensions - one to one)*  or  **PBX** *(internal dialing and PSTN sharing etc)* or hybrid

**CODEC COMPLEXITY** *(for DSP sizing etc)*

G.711 (a-law and u-law), G.726, G.729a and G.729ab are medium

G.728, G.723, iLBC, G.729 and G.729b are high complexity

**REMEMBER RTP**  Real-Time Protocol and RTCP (Real-Time Control Protocol) and how they use their port numbers, payload types etc.
Cisco IOS Study Summary

The examinations use a syllabus based upon extracts from IOS commands and basic knowledge of the current ‘small enterprise’ network devices including the 29xx Catalyst series switches, 28xx G1, 29xx G2, 39xx G2 ‘standard IOS’ ISR routers plus the CME, CUE and basics of the GUI for CME, CUE, CUCM, CUC and CUP.

The latest 640-461 CCNA Voice now requires knowledge of 15.x IOS with Cisco Call Manager Express >7. It is possible to get-by using the trusty 26xx/28xx G1 routers and IOS >12.4 and CME >4, if you ensure you study the few missing commands that would be seen on the IOS 15xx and CME >7. In addition to the equipment needed for the CCENT and CCNA studies, and the capability of running Call Manager Express you will need two or more IP phones and soft phones (Cisco Communicator + CUPC) and access to the full Cisco Unified CallManager, Unity Connection and Unified Presence running on suitable servers or VmWare ESX server with sufficient capacity.

This document is used as checklist within the KCC CCNA Voice FastTrack and Flex training courses;

- Switch and Router basic ops reminder and configuration as seen on the CCENT and CCNA (memory use and functions… RAM, FLASH, ROM, NVRAM)

- Router and Switch CLI (Command Line Interface) and exec mode basics for ;
  - interface and inline power configuration and monitoring
  - VLAN configuration for voice and data
  - CDP functions, DHCP, DNS, TFTP & ip phone boot sequence

- CME CLI for ;
  - setting up the CME as TFTP server
  - initial configuration requirements & file structure etc.
  - CCNA routing and addressing configuration reminder
  - dial-peer configuration
  - dial planning and digit manipulation configuration
  - CME GUI and it’s limited functionality
- service module configuration (CUE) [removed from exam 2011]

- Cisco Unity Express features and specifications
  - single number reach / unified mobility
  - directory
  - call forwarding
  - call pickup
  - intercom
  - paging
  - after-hours blocking

CUCM;

- Dialplans and call flow
- partitions and call search spaces
- GUI access to serviceability, administration, OS and DRS
- GUI service activation, features;
  - extension mobility
  - unified mobility
  - native presence
  - hunt groups
  - intercom
  - music on hold
  - users, groups and rights

- CLI
- TFTP administration
- basic knowledge of trunks and gateways

CUC;

- GUI access to serviceability, administration, OS and DRS
- GUI service activation, features
- CLI
- monitoring and reporting

CUP;

- GUI access to serviceability, administration, OS and DRS
- GUI service activation, features
- CLI

SUPPORT SERVERS/SERVICES;

- LDAP and AD
- TFTP
- DNS and DHCP
- NTP
- Provisioning & end-point boot process
BASIC IOS FUNCTIONS REQUIRED BY CCNAV; (PRACTICE !)

<table>
<thead>
<tr>
<th>FUNCTION</th>
<th>COMMAND (may be abbreviated to first few non-ambiguous characters of each command)</th>
</tr>
</thead>
<tbody>
<tr>
<td>DHCP exclude ranges</td>
<td>ip dhcp excluded-address {range}</td>
</tr>
<tr>
<td>DHCP define pool</td>
<td>ip dhcp pool {name}</td>
</tr>
<tr>
<td>DHCP network</td>
<td>network {network, mask}</td>
</tr>
<tr>
<td>DHCP gateway</td>
<td>default-router {address}</td>
</tr>
<tr>
<td>DHCP DNS server</td>
<td>dns-server {address}</td>
</tr>
<tr>
<td>DHCP TFTP server</td>
<td>option 150 ip {address}</td>
</tr>
<tr>
<td>DHCP directed broadcast</td>
<td>ip helper-address {server address}</td>
</tr>
</tbody>
</table>

Practice configuration of voice VLAN and data VLAN connecting to a router on a different network for DHCP services. Remember your ‘router on a stick’ from the CCNA training and the ip helper commands ☺

CME GUI ADMIN FUNCTIONS REQUIRED BY CCNAV;

REMEMBER

to enable the GUI on CME you must configure ;

- the CME with ip to be reachable from PC and/or CCP
- level-15 username and password
- http services must be enabled
- check any local authentication for telnet/ssh on CME router

ie

```
kcc-CME-11(config-if)#ip address 10.10.30.122 255.255.255.128
kcc-CME-11(config-if)#no shutdown
kcc-CME-11(config)#username fred privilege 15 password cisco123
kcc-CME-11(config)#ip http server
kcc-CME-11(config)#ip http secure-server
kcc-CME-11(config-line)#login local

kcc-CME-11(config-line)#transport input telnet ssh
```

by default CCP (v2.7) runs from the router flash (8xx and 19xx routers with >15.2 IOS) and will try to connect to the CME router using telnet and http, but can be set to use ssh and https instead (both shown above). [NOTE – CCP has replaced SDM as the goofy baby GUI within CCNA although SDM is still used in the CCNA-Security exams]
**BASIC IOS ADMIN FUNCTIONS REQUIRED BY CCNAV:**

(PRACTICE !)

<table>
<thead>
<tr>
<th>FUNCTION</th>
<th>COMMAND (may be abbreviated to first few non-ambiguous characters of each command)</th>
</tr>
</thead>
<tbody>
<tr>
<td>copy config from tftp server to RAM for CME</td>
<td><code>Router # copy tftp system</code></td>
</tr>
<tr>
<td>save/copy running-config (RAM) to NVRAM</td>
<td><code>Router # write memory or copy running-config startup-config</code></td>
</tr>
<tr>
<td>copy file from tftp server to flash memory</td>
<td><code>Router # copy tftp flash</code></td>
</tr>
<tr>
<td>copy file from flash to tftp server</td>
<td><code>Router # copy flash tftp</code></td>
</tr>
<tr>
<td>delete start-up (NVRAM) configuration</td>
<td><code>Router # write erase or erase startup-config</code></td>
</tr>
<tr>
<td>view IOS version information</td>
<td><code>Router &gt; show version</code></td>
</tr>
<tr>
<td>view current configuration (RAM)</td>
<td><code>Router # show running-config or write terminal</code></td>
</tr>
<tr>
<td>view saved (startup) configuration</td>
<td><code>Router # show config or show startup-config</code></td>
</tr>
<tr>
<td>view basic files system (flash)</td>
<td><code>Router # show flash (or dir)</code></td>
</tr>
<tr>
<td>view router utilization</td>
<td><code>Router # show processes</code></td>
</tr>
<tr>
<td>disable CDP for entire router</td>
<td><code>Router (config) # no cdp run</code></td>
</tr>
<tr>
<td>disable CDP on an interface</td>
<td><code>Router (config-int) # no cdp enable</code></td>
</tr>
<tr>
<td>show interfaces and ip addressing</td>
<td><code>Router &gt; show ip interface brief</code></td>
</tr>
</tbody>
</table>
show routing table

Router > show ip route

show ntp status

Router # show ntp status

show ip arp table

Router # show ip arp

Be prepared to use the exam simulator to diagnose and administer ip phones, dailpans and css within the VoIP lab, from GUI of CUCM, CME, CUC and CUP. Configure and diagnose from CLI the DHCP pools, voice VLANs and dial-peer configuration and administration.

NETWORK LAYER UTILITIES;

REMEMBER

ARP Address Resolution Protocol will resolve a mac address from a given ip address. A device may send an ARP broadcast to ask every station on it’s network for the mac address of a given IP address. REMEMBER HOW the ip address and mask dictate if the device should send traffic to it’s local network or to it’s gateway.

DNS Domain Name System will resolve domain names to IP addresses. So a device looking for cisco.com will request a domain lookup from it’s DNS server to be able to send traffic to the IP address of cisco.com (and then using ARP to resolve the IP address of cisco.com to a mac address in order to send it’s traffic). The DNS server on CUCM can support up to 1000 phones.

DHCP Dynamic Host Configuration Protocol can be used to supply IP addresses to any device either via static configuration (mapped to mac address) or via a pool of addresses. DHCP can also provide much more information to the end device such as multiple DNS server addresses and TFTP server addresses etc. REMEMBER to exclude addresses from your DHCP pool for router and other networking devices. USE IP HELPER to direct the DHCP broadcast to the DHCP server from other networks that require the DHCP services. DHCP OPTIONS and how the ip phones use the option 150 during their boot up procedure etc.

NTP Network Time Protocol said (by Cisco) to be crucial to the Cisco VoIP operation. Make sure you know the stratum values expected from a good NTP source and how to configure the routers to make use of NTP.

TFTP Trivial File Transfer Protocol said (by Cisco) to be mandatory to the Cisco VoIP operation (even though the ip phone can boot without TFTP, for the exam Cisco say IT CAN NOT).
REMEMBER according to Cisco:

CUCME/CCME up to 450 users (or 240/250 users in some older books and refer to the pre-G2 ISR router series of 28xx), no redundancy, router based;

2901 - 35 max phones
2911 - 50 max phones
2921 - 100 max phones
2951 - 150 max phones
3925 - 250 max phones
3945 - 350 max phones
3925E – 400 max phones
3945E – 450 max phones

Comm Manager Business ED, to 500 users, no redundancy, server based

CUCM up to 30,000 per cluster, full redundancy, server based

Review the maximum number of supported users for each of the Cisco products and know the differences between the three Unity products.

CUE up to 250 users and 300 mailboxes, no redundancy, email relay, router based

CUC up to 7500 users and up to 20,000 mailboxes, active/active redundancy, email relay, server based appliance

CU up to 7500 per server to 25,000 max, active/passive redundancy support, full email integration, windows server based toy

Be aware of the Cisco application products such as IVR/AA, Contact Center, Mobility and Emergency Responder.
LAN SWITCHING Study Summary

- Protocol Type Fields and header formats – CCNA knowledge reminder
- **VLANs overview** inter-vlan routing, collision domain / broadcast domain and segments, voice VLAN configuration and native VLAN. CCNA knowledge PLUS voice vlan configuration and basic QoS knowledge.
- **Trunking/Tagging Protocols & VTP basics** (VTP modes, tagging specifications ISL/802.1q)

Power Over Ethernet (PoE) Study Summary

Cisco Inline Power = the Cisco pre-standard power over Ethernet system that uses a FAST LINK PULSE (FLP) tone to check for Cisco device requiring inline power. A Cisco inline power device will loop the tone back to the switch if it requires power. The switch will make a minimum of 6.3W power available until the actual value is received using CDP.

The new **802.3at PoE – (called PoE plus)** = IEEE standard increases the 802.3af to **25.5W** {note – some proprietary PoE-Plus now handle up to 51W by using all pairs}

**802.3af PoE = IEEE** standard delivers a small current to detect a PoE device that requires power. Each 802.3af device has a resistor on the power lines to let the switch know it’s power requirements based on the following four classes;

- Class 0 = (no value requested, just ‘give me power’) 15.4W,
- Class 1 4W,
- Class 2 7W,
- Class 3 15.4W

Review the Cisco IP Phones and their power standards

Network Protocols Study Summary  (CCNA + VoIP)

- **TCP/IP** (RFC 793, UDP, port numbers and type numbers (RFC 1700), DNS, ARP, ICMP, SCCP, SIP, RTP, RTCP, MGCP, H323)
- **IP Addressing and classes (subnet masking before VLSM), default routes … CCNA knowledge.**
- **Encapsulation in IP** header sizes and compression savings (IP,TCP,UDP,RTP - overheads **40 bytes becomes 2-4 bytes**) Ethernet overhead = 20 bytes, PPP overhead = 6 bytes, Frame Relay = 4-6 bytes, GRE/L2TP = 24 bytes, MPLS = 4 bytes, IPSec = 50-57 bytes
- IP and MAC addressing flow
- **DNS, DHCP (and helper) and general WEB traffic flow**
- NTP configuration and stratum
• FTP TFTP (and using the CCME as a TFTP server)
• **IOS commands** (CCNAV sub-set of commands for CME)
• **KNOW** CDP and how the ip phone uses it, what it can show, how it can help fault finding
• **KNOW** the skinny IP Phone boot sequence
• **KNOW** the CCME telephony-service main configurations
• Cisco IP phones can be loaded with SIP, SCCP or MGCP firmware

<table>
<thead>
<tr>
<th>VoIP Protocol</th>
<th>Standard</th>
<th>Support</th>
<th>Architecture</th>
</tr>
</thead>
<tbody>
<tr>
<td>H.323</td>
<td>ITU</td>
<td>Excellent</td>
<td>Peer to peer</td>
</tr>
<tr>
<td>MGCP</td>
<td>IETF</td>
<td>Fair</td>
<td>Client – Server</td>
</tr>
<tr>
<td>SIP</td>
<td>IETF</td>
<td>Very Good</td>
<td>Peer to peer</td>
</tr>
<tr>
<td>SCCP</td>
<td>Cisco</td>
<td>Proprietary</td>
<td>Client – Server</td>
</tr>
<tr>
<td>IAX</td>
<td>Digium</td>
<td>Proprietary</td>
<td>Peer to peer</td>
</tr>
</tbody>
</table>

**common configuration & show commands:**  (practice these commands !)

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show ip protocol</td>
<td>View routing protocols in use for ip</td>
</tr>
<tr>
<td>show controller {serial</td>
<td>ethernet</td>
</tr>
<tr>
<td>show debug</td>
<td>View current debug setting</td>
</tr>
<tr>
<td>show voice port summary</td>
<td>view pots voice ports</td>
</tr>
<tr>
<td>show telephone-service tftp-bind</td>
<td>Show tftp configuration for CME</td>
</tr>
<tr>
<td>show ip interface {brief}</td>
<td>View IP interface details</td>
</tr>
<tr>
<td>show telephone-service all</td>
<td>View CME configuration</td>
</tr>
<tr>
<td>show dialplan number xxxxx</td>
<td>Test and display dialplan for xxxxx</td>
</tr>
<tr>
<td>show ip interface {brief}</td>
<td>View IP interface details</td>
</tr>
<tr>
<td>show ntp status</td>
<td>View ntp status</td>
</tr>
<tr>
<td>ip host name {tcp-port-number} address1 address2...</td>
<td>configuration of host table</td>
</tr>
<tr>
<td>ip route prefix mask {next hop</td>
<td>output interface}</td>
</tr>
<tr>
<td>ip name-server server address1 {server address2...}</td>
<td>configure name server(s) for DNS</td>
</tr>
<tr>
<td>show ephone unreg</td>
<td>See registration attempts</td>
</tr>
<tr>
<td>show run</td>
<td>begin dial-peer</td>
</tr>
<tr>
<td>auto qos voip cisco-phone (on a switch)</td>
<td>Trust if from Cisco phone + enable qos</td>
</tr>
<tr>
<td>auto qos voip trust</td>
<td>Trust existing marking + enable qos</td>
</tr>
<tr>
<td>auto qos voip (on a router)</td>
<td>ACL or NBAR to identify QoS</td>
</tr>
</tbody>
</table>
CCME – CCNAV NEED TO KNOW:

LICENSING REQUIRED ON CME

- IOS License
- Feature License
- Phone User License

<table>
<thead>
<tr>
<th>CCME initial config</th>
<th>Config</th>
<th>command</th>
</tr>
</thead>
<tbody>
<tr>
<td>Go into telephony service configuration mode</td>
<td>config term followed by telephony-services</td>
<td></td>
</tr>
<tr>
<td>Set memory limits for ephones to be used and directory numbers</td>
<td>(config-telephony)#</td>
<td>max-ephones {number}</td>
</tr>
<tr>
<td></td>
<td>(config-telephony)#</td>
<td>max-dn {number}</td>
</tr>
<tr>
<td>Set CME source address</td>
<td>(config-telephony)#</td>
<td>ip source-address (address)</td>
</tr>
<tr>
<td>Create phone configs</td>
<td>(config-telephony)#</td>
<td>create cnf-files</td>
</tr>
</tbody>
</table>

REMEMBER - main steps to start the CME (as above)

- Set maximum phones and directory numbers (**max-dn**, **max-ephone**)
- Set source ip address
- Generate configuration files for the ip phones
- Firmware load files configured for TFTP use (tftp location and/or CME tftp & alias global commands etc.)

<table>
<thead>
<tr>
<th>ephone-dn configuration</th>
<th>Description</th>
<th>note</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone-dn {tag} {dual}</td>
<td>Tag identifies the logical ephone directory number</td>
<td>Default = single line</td>
</tr>
<tr>
<td>number {number} secondary {number}</td>
<td>Assign directory number to the ephone-dn</td>
<td></td>
</tr>
<tr>
<td>name {directory name}</td>
<td>Set internal callerid name</td>
<td></td>
</tr>
<tr>
<td>call-forward {op} {number}</td>
<td>Assign directory number to the ephone-dn</td>
<td></td>
</tr>
<tr>
<td>park-slot</td>
<td>Set as parking slot</td>
<td></td>
</tr>
<tr>
<td>pickup-group</td>
<td>Set pick up group number</td>
<td></td>
</tr>
<tr>
<td>intercom {number} label {txt}</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Paging</td>
<td>Define dn as paging number</td>
<td></td>
</tr>
<tr>
<td><strong>ephone configuration</strong></td>
<td><strong>Description</strong></td>
<td><strong>note</strong></td>
</tr>
<tr>
<td>--------------------------</td>
<td>-----------------</td>
<td>---------</td>
</tr>
<tr>
<td>ephone {tag}</td>
<td>Tag identifies the logical ephone device</td>
<td></td>
</tr>
<tr>
<td>mac {mac address}</td>
<td>Assign ip phone mac address to ephone logical tag</td>
<td></td>
</tr>
<tr>
<td>button {number} {op} {number}</td>
<td>Assign dn lines to ip phone button lines with the various button functions</td>
<td>SEE BUTTON OP TABLE BELOW</td>
</tr>
<tr>
<td>preference {value}</td>
<td>Set preference for ephone</td>
<td>Default = 0 (best)</td>
</tr>
<tr>
<td>Restart</td>
<td>Warm reboot of ephone</td>
<td>For restart after minor changes such as button line or speed dial changes</td>
</tr>
<tr>
<td>Reset</td>
<td>Cold reboot of ephone</td>
<td>For reset after main changes such as DHCP, date, firmware, locale, services, voicemail etc…</td>
</tr>
<tr>
<td>Huntstop</td>
<td>Stop hunting</td>
<td>Default</td>
</tr>
<tr>
<td>Huntstop channel</td>
<td>Stop hunting on channel</td>
<td>Prevents call going to call-waiting on dual line</td>
</tr>
<tr>
<td>Paging</td>
<td>Set ephone paging group membership</td>
<td></td>
</tr>
<tr>
<td>pin {number}</td>
<td>Set pin number for login</td>
<td></td>
</tr>
<tr>
<td>after-hour exempt</td>
<td>Allow after-hours exemption</td>
<td></td>
</tr>
<tr>
<td>BUTTON COMMAND</td>
<td>Description</td>
<td>note</td>
</tr>
<tr>
<td>---------------------</td>
<td>--------------------------------------------------</td>
<td>-------------------------------------------</td>
</tr>
<tr>
<td>button {number} :</td>
<td>Assign dn line to ephone</td>
<td></td>
</tr>
<tr>
<td>{number}</td>
<td>button line</td>
<td></td>
</tr>
<tr>
<td>button {number} b</td>
<td>Assign with call waiting</td>
<td></td>
</tr>
<tr>
<td>{number}</td>
<td>beep, no ring</td>
<td></td>
</tr>
<tr>
<td>button {number} f</td>
<td>Assign with feature ring</td>
<td></td>
</tr>
<tr>
<td>{number}</td>
<td></td>
<td></td>
</tr>
<tr>
<td>button {number} m</td>
<td>Assign as monitor line</td>
<td>Monitor single number</td>
</tr>
<tr>
<td>{number}</td>
<td></td>
<td></td>
</tr>
<tr>
<td>button {number} w</td>
<td>Assign as watched device</td>
<td>Monitor entire ip phone</td>
</tr>
<tr>
<td>{number}</td>
<td></td>
<td></td>
</tr>
<tr>
<td>button {number} o</td>
<td>Assign as overlay, no call waiting</td>
<td></td>
</tr>
<tr>
<td>{number}</td>
<td></td>
<td></td>
</tr>
<tr>
<td>button {number} c</td>
<td>Assign as overlay with call waiting</td>
<td></td>
</tr>
<tr>
<td>{number}</td>
<td></td>
<td></td>
</tr>
<tr>
<td>button {number} x</td>
<td>Assign as expansion overlay</td>
<td></td>
</tr>
<tr>
<td>{number}</td>
<td></td>
<td></td>
</tr>
<tr>
<td>button {number} s</td>
<td>Assign as silent ring, no call waiting</td>
<td></td>
</tr>
<tr>
<td>{number}</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**WAN Protocols & CODEC Considerations,**

- CODEC for WAN <T1 G.729
- CODEC DEFAULT for dial-peers = G.729
- CODEC DEFAULT for CUE = G.711ulaw
- G.729 defaults to 20ms = 20bytes per packet
- G.711 defaults to 20ms = 160bytes per packet
- VAD can save about 35% bandwidth
- RTP header compression results in the 40 byte (IP/RTP/UDP) headers being stripped to leave an overhead of 2 bytes plus 2bytes of checksum
- RTP port numbers range from 16384 to 32767
- RTP uses the even numbered ports, RTCP uses the odd number
- Max end to end delay advised as 150ms
- Jitter max = 30ms
- Packet Loss should be 1% or less

**BYTES PER PACKET = (SAMPLE SIZE x CODEC BANDWIDTH) / 8**

<table>
<thead>
<tr>
<th>CODEC</th>
<th>Complexity</th>
<th>Bandwidth Consumed</th>
<th>MOS</th>
</tr>
</thead>
<tbody>
<tr>
<td>G.711</td>
<td>Medium</td>
<td>64kbps</td>
<td>4.1</td>
</tr>
<tr>
<td>G.729</td>
<td>High</td>
<td>8kbps</td>
<td>3.92</td>
</tr>
<tr>
<td>G.729a</td>
<td>Medium</td>
<td>8kbps</td>
<td>3.7</td>
</tr>
<tr>
<td>G.729b</td>
<td>High</td>
<td>8kbps</td>
<td>4.0</td>
</tr>
<tr>
<td>iLBC</td>
<td>High</td>
<td>15.2kbps</td>
<td>4.1</td>
</tr>
</tbody>
</table>
**CUCM ADMIN**

**remember** – you need the CLI for some commands:

```
admin:utils ntp status
ntp (pid 16947) is running...

<table>
<thead>
<tr>
<th>remote</th>
<th>refid</th>
<th>st</th>
<th>when</th>
<th>poll</th>
<th>reach</th>
<th>delay</th>
<th>offset</th>
<th>jitter</th>
</tr>
</thead>
<tbody>
<tr>
<td>129.138.141.172</td>
<td>aCTS.</td>
<td>1 u</td>
<td>32</td>
<td>64</td>
<td>377</td>
<td>170.996</td>
<td>-3.842</td>
<td>9.010</td>
</tr>
</tbody>
</table>
```

synchronised to NTP server (129.138.141.172) at stratum 2
  * time correct to within 10 ms
  * polling server every 64 s

```
admin:utils system
  utils system boot*
  utils system restart
  utils system shutdown
  utils system switch-version
  utils system upgrade*
```

```
admin:show network
  show network all
  show network cluster
  show network dhcp*
  show network eth0
  show network failover
  show network ip_contrack
  show network ipprefs*
  show network ipv6*
  show network max_ip_contrack
  show network route
  show network status
```

CLI is required for shutdown and restart, change of version during upgrade, modify network address and DNS etc. {although this would change the license in the real world}, ping, trace, packet capture, modify admin accounts, display server load and status etc....

```
admin:utils system shutdown

Do you really want to shutdown ?

Enter (yes/no)? yes

Appliance is being Powered - Off ...
Warning: Shutdown could take up to 5 minutes.
Shutting down Service Manager will take some time...
Service Manager shutting down services... Please Wait
```

more details:

remember the CUCM GUI options and features;

and be very familiar with the GUI tabs 

the SYSTEM tab;
above - date/time group configured for each timezone

above - region configuration  (remember the region and locations for CAC BW etc)

NOTES -:
above - the heart of the dial plan is the route pattern – route list – route group
{note knowledge of the special ‘local route groups’ are not required for CCNA}

NOTES -:
above - partitions and calling search space  VERY IMPORTANT to know their function within CUCM
above - configuration of CUCM to connect to CUC know the difference between voicemail ports, pilot, profile and voicemail boxes….
above - be familiar with adding phones and configuration of devices

above - remember the pull down sub-menu within phone configuration to change the phone subscription services (for extension mobility etc)
above - the User Management tab to administer users, roles and groups etc.

above - Bulk Administration tab showing the sub menu for Mobility
above - access the other GUIs from the pulldown Navigation Menu

NOTES -:
above – from the Call Routing tab select Route Plan Report;

above - CUCM Dial-Plan Report shows the entire dial-plan
CUPS ADMIN;

Cisco Unified Presence Administration

Find and Settings Gateways

Presence Gateway (1 - 1 of 1)
Find Presence Gateway where Presence Gateway begins with

<table>
<thead>
<tr>
<th>Presence Gateway</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>192.168.0.119</td>
<td>CUCM-PUB</td>
</tr>
</tbody>
</table>

Add New Select All Clear All Delete Selected
above - CUPC application

be familiar with RTMT
Cisco Unified Reporting
For Cisco Unified Communications solutions

System Reports
Unified CM Cluster Overview
Unified CM Data Summary
Unified CM Database Status
Unified CM Device Summary
Unified CM Device Distribution Summary
Unified CM Extension Mobility
Unified CM Geolocation Policy
Unified CM Geolocation Policy with Filter
Unified CM Lines Without Phones
Unified CM Multi-Line Extensions
Unified CM Phone Feature List
Unified CM Phones
Unified CM Version

Unified CM Cluster Overview

Provides an overview of the Unified CM cluster.
Created on Sun Apr 24 12:54:58 GMT+00:00 2011

Unified CM Cluster Name

Lists the cluster name from the Enterprise Parameter and the publisher server name/IP.

Unified CM Provisioned Servers

Lists all servers in the cluster by either name or IP as provisioned in the database and whether the server address is obtained from the host file on the local server.

Unified CM Version

Checks the Unified CM version running on each installed server and returns a summary checking to

The CUCM Version on all servers is cm-ver-7.1.3.30080-1.

above - diagnostics from the CUCM Unified Reporting
ALSO REMEMBER the irrelevant:

- CUC supports up to 20,000 mailboxes
- the ‘Cisco Troubleshooting Methodology’ steps:
  - define problem
  - gather facts
  - consider possibilities
  - create action plan
  - implement action plan
  - observe results
- PVDM-2-8 provides .5 DSP, PVDM-2-16 provides 1 DSP etc..
- CUE max mailboxes = 300