Cisco CCNA-Voice Check List - Training Notes

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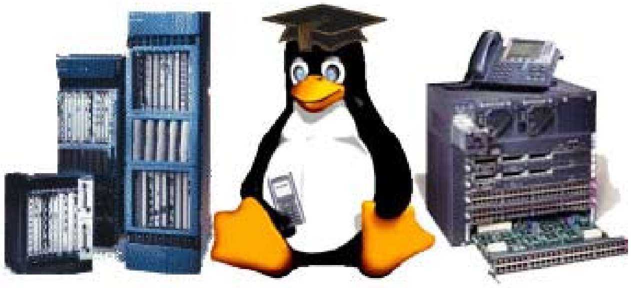

These notes cover the current 640-460 examination as the single exam option for CCNA Voice to complete the CCNA Voice certification.

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. *** please report any errors/comments.

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 CCA GUI to configure the SBCS range of products and several non-technical questions regarding product range and specifications ☹

see also CCNA Voice example questions, IOS Configuration Examples,

The Vue / Pearson tests can be booked online via 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 (extended super frame) 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

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, 26xx /18xx/28xx 'standard IOS' routers as seen in the CCENT and CCNA studies **plus the CME**, CUE and basics of the GUI for CME, CUE and the UC500 SBCS range using the Cisco Configuration Assistant application.

The 2500 and 2600 are fine for CCENT and CCNA as excellent home training routers and are available at very low cost, but the CCNA Voice requires >12.4 IOS with Cisco Call Manager Express, so you may need memory and flash expansion to run the required IOS on a 2600 (CME will not run on 2500). 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/or soft phones. The SBCS UC500 is only GUI driven and not essential in the training lab. The few exam questions on the SBCS are mainly regarding product specification, default addressing and basic GUI functions.

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

- Switch and Router basic ops 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 ;

    - file system messages commands for logging etc. {lab #1}

    - interface configuration and monitoring {all labs}

    - VLAN configuration for voice and data

    - VTP configuration

    - CDP functions, DHCP, DNS, TFTP & ip phone boot sequence {all labs}

- CME CLI for ;

    - loading the software {lab #1}

    - setting up the CME as TFTP server {lab #2}

    - initial configuration requirements & file structure etc. {lab #2}

    - CCNA routing and addressing configuration {all labs}

    - CDP functions, DHCP, DNS, TFTP & ip phone boot sequence {all labs}

    - ePhone-dn and ephone configuration {all labs}

    - dial-peer configuration {all labs}

    - dial planning and digit manipulation configuration {all labs}

- CME features (phone button configuration, after hours, overlay, shared line, paging, hunting, intercom, pickup groups, call parking, auto attendant, music on hold, phone registration/provisioning, emergency and pstn backup)
- CME GUI and it's limited functionality {all labs}
- service module configuration basics (CUE)
- Cisco Unity Express features and specifications
- Cisco Small Business Communications Systems features and specifications
- SBCS UC500/520 basic administration

BASIC IOS FUNCTIONS REQUIRED BY CCNAV; (PRACTICE !)

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

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 ☺

BASIC IOS ADMIN FUNCTIONS REQUIRED BY CCNAV;

(PRACTICE !)

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

NOTE:CCNAV makes use of 'standard catalyst' 29xx switches, 'standard IOS' routers (26xx,18xx,28xx etc) and CCME version >3, CCUE version >2.3.4

Be prepared to use the exam simulator to **configure an entire VoIP lab**, from CME initial configuration, DHCP pools, VLANs, ephone configuration and administration.

Reference Models Study Summary

- OSI Reference model (as seen on the CCENT/CCNA etc.)
- TCP/IP Reference model (as seen on the CCENT/CCNA etc.)
- Cisco VOIP Reference model (end points, applications, call routing, infrastructure)
- Frames, Packets and Segments (RTP, RTCP, UDP, TCP)
- Flow control methods (buffering, congestion avoidance and windowing, basic QoS)

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)

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.

OSI MODEL AND EXAMPLES;

OSI Layer Name	Description	Example
Application layer 7	Application / user interface (including user authentication etc)	Telnet, HTTP, FTP, SMTP, SNMP, **SIP**, **MGCP,** POP3, FTP
Presentation layer 6	Data translation /presentation / encryption	JPEG, EBCDIC, ASCII, GIF, MPEG, MIDI, Encryption....
Session layer 5	Session control, allocation/tracking	Op systems, SQL, NetBIOS, DECnet
Transport layer 4	Multiplexing /control, Data delivery using flow control and error recovery & segmentation etc	TCP, UDP, SPX
Network layer 3	logical addressing and path determination (routing)	IP, IPX, AppleTalk, X.25
Data Link layer 2	frame construction, mac addressing, error detection using frame check sequence (switching)	802.2/802.3, VTP, HDLC, ATM, PPP, Frame Relay, Ethernet, CDP......
Physical layer 1	Electrical connections & signals… physical media	Cables specifications; RJ45, V.35, EIA232, Ethernet
TCP/IP Layer Name	**Description**	**Example**
Application	Application / user interface / Presentation and session control) maps to Layers 5, 6 & 7 of OSI	Telnet, HTTP, FTP, SMTP, SNMP, VoIP, POP3, FTP, SIP, MGCP
Transport	Multiplexing /control, Data delivery using flow control and error recovery etc .. maps to Layer 4 of OSI	TCP, UDP
Internet	logical addressing and path determination maps to layer 3 of OSI	IP
Network Access	Maps to Layer 1 & 2 of OSI model = mac protocols & physical media etc	802.2/802.3, PPP, Frame Relay, Ethernet, CDP......

CISCO VOIP REFERENCE MODEL

Layer Name	Description	Example
End Points	Terminating point for voice	Ip phones, gsm, video phone, soft phone clients etc…
Applications	Application services	Voice mail, call center applications, ACD, CRM, 911 server….
Call Processing	CONTROL = Supplying dial tone and doing the call switching	Call Manager = unified communications manager, CCME, UC500/SBCS, Asterisk, Avaya IP PBX etc…
Infrastructure	Network equipment	Firewalls, voice gateways, voice router, voice enabled switches etc…

REMEMBER according to Cisco ;

Cisco UC500/SBCS 8 to 48 users, no redundancy, router based, **fixed sizes**

CUCME/CCME up to 250 users (or 240 users in some older books), no redundancy, router based

Comm Manager Bussiness 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, no redundancy, email relay, router based

CUC up to 7500 users, no redundancy, email relay, server based

CU up to 7500 per server to 25,000, redundancy support, full email integration, server based

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

- Ethernet Standards (mac specifications, **cable lengths & types**)

- Spanning Tree basic functions (now including RSTP 802.1w and PVSTP) – no need to know the timing and protocol details, but **essential to know the port naming, election procedure, bridge ID and basic spanning tree functionality. = CCNA knowledge**

- 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.

- **Trunking/Tagging Protocols & VTP basics** (VTP modes, tagging specifications ISL/802.1q)

REMEMBER: VTP MODES on all Cisco Catalyst switches (flooded every 5mins & when ever there has been a change);

Function	**Server**	**Client**	**Transparent**
source VTP messages	yes	yes	no
listen to VTP messages	yes	yes	no
create/edit/delete VTP messages	yes	no	local
save VTP messages	yes	no	local

TRUNK/TAGGING DETAILS;

- Cisco's ISL encapsulation (adds 26 bytes overhead) tagging for VLAN identification for Fast Ethernet or Gigabit Ethernet links only

- 802.1Q is the IEEE standard (subset of Cisco's ISL) for VLAN tagging adds a 4 byte shim

- VTP is Cisco's method for distribution of VLAN configuration information

- VTP pruning increases available bandwidth by restricting flooded traffic to contain only the required/configured VLANs for that trunk and not sending all available VLAN information

see also VLAN Overview

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.

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 & CUE)
- **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 configurations**
- REMEMBER - CUE needs a static route **IF configured to use the loopback interface** of the CCME router
- Cisco IP phones can be loaded with SIP, SCCP or MGCP firmware
- SBCS will only support SCCP ip phones, but will support SIP trunks

VoIP Protocol	Standard	Support	Architecture
H.323	ITU	Excellent	Peer to peer
MGCP	IETF	Fair	Client – Server
SIP	IETF	Very Good	Peer to peer
SCCP	Cisco	Proprietary	Client – Server
IAX	*Digium*	*Proprietary*	*Peer to peer*

common configuration & show commands; (practice these commands !)

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

CCME – CCNAV NEED TO KNOW;

LICENSING REQUIRED ON CME

- IOS License
- Feature License
- Phone User License

CCME install	Description	Command
Install CME software	Install from TFTP to flash	copy tftp://{file} flash:
Extract software to flash directories	Extract files from tar file	archive tar /xtract tftp://{file} flash:
CCME initial config	Config	command
Go into telephony service configuration mode	config term followed by telephony-services	
Set memory limits for ephones to be used	(config-telephony)#	max-ephones {number}
and directory numbers	(config-telephony)#	max-dn {number}
Set CME source addres	(config-telephony)#	ip source-address (address}
Create phone configs	(config-telephony)#	create cnf-files

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.)

ephone-dn configuration	Description	note
ephone-dn {tag} {dual}	Tag identifies the logical ephone directory number	Default = single line
number {number} secondary {number}	Assign directory number to the ephone-dn	
name {directory name}	Set internal callerid name	
call-forward {op} {number}	Assign directory number to the ephone-dn	
park-slot	Set as parking slot	
pickup-group	Set pick up group number	
intercom {number} label {txt}		
Paging	Define dn as paging number	
ephone configuration	Description	note
ephone {tag}	Tag identifies the logical ephone device	
mac {mac address}	Assign ip phone mac address to ephone logical tag	
button {number} {op} {number}	Assign dn lines to ip phone button lines with the various button functions	SEE BUTTON OP TABLE BELOW
preference {value}	Set preference for ephone	Default = 0 (best)
Restart	Warm reboot of ephone	For restart after minor changes such as button line or speed dial changes
Reset	Cold reboot of ephone	For reset after main changes such as DHCP, date, firmware, locale, services, voicemail etc…
Huntstop	Stop hunting	Default
Huntstop channel	Stop hunting on channel	Prevents call going to call-waiting on dual line
Paging	Set ephone paging group	

	membership	
pin {number}	Set pin number for login	
after-hour exempt	Allow after-hours exemption	

BUTTON COMMAND	Description	note
button {number} **:** {number}	Assign dn line to ephone button line	
button {number} **b** {number}	Assign with call waiting beep, no ring	
button {number} **f** {number}	Assign with feature ring	
button {number} **m** {number}	Assign as monitor line	Monitor single number
button {number} **w** {number}	Assign as watched device	Monitor entire ip phone
button {number} **o** {number}	Assign as overlay, no call waiting	
button {number} **c** {number}	Assign as overlay with call waiting	
button {number} **x** {number}	Assign as expansion overlay	
button {number} **s** {number}	Assign as silent ring, no call waiting	

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

CODEC	Complexity	Bandwidth Consumed	MOS
G.711	Medium	64kbps	4.1
G.729	High	8kbps	3.92
G.729a	Medium	8kbps	3.7
G.729b	High	8kbps	4.0
iLBC	High	15.2kbps	4.1

So for example, it can be seen that a G.729 call using the default 20ms would have a sample size of (20ms x 8kbps) / 8 = 20 Bytes

ADD the overheads (assume no compression) ;

IP = 20 bytes, UDP = 8 bytes, RTP = 12 bytes, Ethernet = 20 bytes

That is 60 bytes of overhead plus the voice payload of 20 bytes

= 80 bytes for every 20ms

Bandwidth required for 1 second (1000ms) would be

80 bytes x (1000/20 ms)

= 4000 bytes/second

= 4000 x 8 bits per second

= 32kbps

{this calculation by formula only required @ CCVP level}

ALSO **REMEMBER the irrelevant :**

- CUE supports up to 250 mailboxes (up to 50 on AIM)
- CUE default AA recordings are: welcome, office open, office closed, holiday
- CUE GDM is accessed via an associated voicemail subscriber
- CUE uses TRACE and not debug to troubleshoot problems
- Access the CUE AIM via the 'service-engine interface' using the 'service-module service-engine 1/0 session' command from the CME router CLI
- CUE software download upgrade command is only required for version 2.3.4
- CUE license requires CME/CM, GDM count, mailbox count, IVR count
- SBCS is 'simple and affordable' - provides simplicity, predinfined hardware and licensing, plug and play feature rich suite, predefined use case assumptions, CCA configuration (to allow auto phone assignment, dial plan setup, voice and data vlan setting, firewall setup and cisco best practice QoS)
- UC520/SBCS manager defaults to 192.168.10.1 (seed for CCA)
- UC520/SBSC data defaults to 192.168.1.0/24 VLAN 1
- UC520/SBCS auxiliary/voice vlan defaults to 10.1.1.0/24 (VLAN 100)
- CCA is a powerful ☺ GUI that connects via http or https for 800 series routers, Cat Express switches, UC500/520, WLAN controllers and HWIC APs
- SBCS 'key-system' configuration has no COR, 'PBX' has COR enforced