CCNA Voice IIUC 640-460 Notes

Chapter 01

Perspectives on Voice Before Convergence

Key Terms

Analog Signal – Method of transmitting a signal by use of the properties of the transmission medium (Electrical signals, etc).

Loop start signaling – When a phone is taken off the hook, a connection between the Ground (tip) and Battery (ring) wires is made, completing the signal.

Ground start signaling – Does not provide a dual tone when the phone is lifted.

Glare – When a phone is picked up to make an outgoing call at the same time an incoming call is arriving. Loop start is susceptible to this.

Dual-tone multifrequency (DTMF) – Each telephone button uses a pair of high and low electrical frequencies to generate a signal when pressed.

Pulse dialing – Rotary connects and disconnects from local loop circuit to specify digits.

Pulse-amplitude modulation (PAM) – The process of sampling an analog signal many times as it changes to convert it to a numeric value.

Pulse-code modulation (PCM) – The samples (PAM) value is converted into an 8-bit binary number.

Time-division multiplexing (TDM) – Allows multiple conversations to be carried over a 4-wire path at the same time using time slots to differentiate between the conversations.

Channel associated signaling (CAS) – Signaling information is transmitted using the same bandwidth as voice.

Common channel signaling (CCS) – Signaling information is transmitted using a dedicated signaling channel.

Robbed bit signaling (RBS) – The process of stealing bits from a voice channel to send signaling information (Like CAS).

Super frame (SF) – Sends groups of 12 T1 frames (192-bit) at a time. All bits are used for T1 synchronization.

Extended super frame (ESF) – Sends groups of 24 T1 frames at a time.2000-bits for synchronization, 2000-bits for error checking, & 4000-bits as a supervisory channel

Q.931 – A popular signaling protocol used with CCS (Used for ISDN circuits)

Local loop - PSTN link between customer and the telco.

Private branch exchange (PBX) – Allows a company to run a larger private voice network. Usually used to create unique extensions for all devices.

Key system – Allows a company to run a private voice network and share extensions on all phones.

Signaling system 7 (SS7) – Out of band signaling method used to communicate call setup, routing, billing, and information messages. Standard used across the world.

E.164 – International numbering plan containing country code, national destination code, and subscriber number.

Quantization – The process of matching analog samples to a voltage scale.

Notes

- Ground wire = tip, Battery wire = ring
- Ground start signaling allows a PBX to reduce the problem of glare
- Supervisory Signaling includes:
  - On-hook signal – connection between tip and ring is broken
  - Off-hook signal – tip and ring connected (complete signal
  - Ringing – alternating current (AC) is sent across one wire to make the phone ring
- Informational signaling is used after an off-hook signal is generated by the phone
  - The phone company uses this to provide dial tone and information to the user including:
  - Dial tone – PC is ready to receive digits
  - Busy – remote phone is in use
  - Ringback – remote phone is currently ringing
  - Congestion – LD telco network cannot complete call
  - Reorder – local PC cannot complete call
  - Receiver off-hook – Local phone has been off hook for a period of time
  - No such number – dialed number is not valid
  - Confirmation – PC is attempting to complete call
- Nyquist sampled the human voice at 4000 Hz (8000 digital samples/sec)
- Quantization divides the voltage ranges into 16 total segments (0 to 7 positive/negative)
  - Values in higher segments are not as accurately sampled as lower values
- Digital technology caused a problem with sending signaling information. CAS and CCS solved this
- The PSTN is made up of the following pieces:
  - Analog telephone – Converts audio to electrical signals
  - Local Loop – The link between with customer and the telco SP
  - Central Office switch – Provides signaling, digit collection, call routing, setup, and teardown to devices on the local loop.
  - Trunk – Provides connection between switches (CO or private)
  - Private switch – Allows a company to operate a mini PSTN

- Digital Phone – Converts audio to binary digits, and usually connects to a PBX.

- CAS uses the eighth bit on every sixth sample to transmit signaling information.
- T1/E1 time slots 24/16 are used for CCS signaling
- Key systems typically have a shared line approach with all users having all lines
- E.164 numbers can be a maximum of 15 digits
    - North American Numbering Plan (NANP) uses E.164 in the following form
    - Country Code
    - Area Code
    - CO or exchange code
    - Station code
    - Are and CO/Exchange codes are combined in the E.164 NDC

Chapter 02

Perspectives on Voice After Convergence

Key Terms

Cisco Unified Communication 500 (UC500) – An appliance that supports from 8 to 48 users and incorporates many voice features.

Cisco Unified Communications Manager Express (CME) – Runs on an integrated services router and supports up to 250 phones.

Cisco Media Convergence Server (MCS) – The server hardware platforms that support Cisco Unified Communications Manager software.

Cisco Unified Communications Manager Business Edition – Appliance based with support for up to 500 phones. Provides Unity and mobility functions as well.

Cisco Unified Communications Manager – Full CM that can support up to 30,000 phones per cluster.

Cisco Unity Express – Basic VM software that runs on an ISR.

Cisco Unity Connection – Appliance based, provides advanced call routing features.

Cisco Unity – Full VM appliance that provides messaging integration and other advanced features.

Interactive Voice Response (IVR) – Plays a message to a user and requests that the user press a button is response

Auto Attendant – Allows users to direct themselves to the correct person in a company by following prompts.

Cisco Unified Contact Center Express – Routes calls to the correct groups and provides automatic call distribution to agents/support staff.

Cisco Unified MeetingPlace – Provides voice, video and data conferencing on one call.

Cisco Unified Presence – Provides status and reachability information for voice users.

Cisco Unified Mobility – Single number reach features.

Cisco Emergency Responder – Automatically updates a user’s location information based on where they are on the network.

Notes

- VOIP Advantages
  - Reduced communication costs
  - Reduced cabling costs
  - Seamless voice networks (Integrate well with current data networks)
  - Easy adds, moves, and changes
  - IP Softphones (Reduced hardware costs)
  - Unified e-mail, voicemail and fax
  - Increased productivity – single number reach, etc
  - Feature-rich communications
  - Open compatible standards
- Cisco Unified Communications Model
  - Endpoints
  - Applications
  - Call Processing
  - Infrastructure
- Unified CME can run on certain routers or on UC500 appliances
- Unified Communications Manager Business Edition
  - Linux appliance based
  - Support for up to 500 IP phones
  - UCM – Call processing
  - Unity – Integrated VM
  - Unified mobility – single number reach features
- Unity express runs on a router as an AIM or NM(E)
  - Support for up to 250 users
- Unity Connection
  - For more than 100 but less than 7500 users
  - Advanced call routing rules and speech recognition
  - Can be run on UCM BE if less than 500 users
- Unity
  - Supports full unified messaging (Fax, VM, and e-mail in one system)
  - Up to 7500 users per server (network to 250,000)
- Unified Contact Center
  - Runs on a dedicated server and provides automatic call distributor service (ACD)
  - ACD distributes calls in an organization as well as IVR capabilities
- Unified Mobility allows users to be reached with one number
- Phones with XML app support
  - Entry Level Phones: 7906G, 7911G, 7931G (Basic XML service)
  - Business-Class: 7940G(GE) series
- VT advantage brings video capabilities to a computer (Camera and software)
- ATA 186/188 allow you to tie legacy devices (Faxes, etc) to the VoIP network

Chapter 03

Connecting IP Phones to the LAN Infrastructure

Key Terms

802.3af Power over Ethernet (PoE)

Cisco Inline Power – Prestandard way of providing power to devices through a network cable.

Cisco Discovery Protocol (CDP) – Protocol used to allow Cisco devices to discover each other.

Virtual LAN (VLAN) - Allows for breaking up a network into multiple broadcast domains.

Trunking – The practice of carrying traffic for multiple VLANS from one switch to another.

Inter-Switch Link (ISL) – Trunking protocol that was only support by Cisco devices (Replaced by 802.1Q).

802.1Q – Standards based protocol that allows for traffic from multiple VLANS to be sent.

VLAN Trunking Protocol (VTP) – A Cisco protocol that replicates the VLAN database to other switches in the VTP domain.

Dynamic Trunking Protocol (DTP) – Allows switches to dynamically negotiate trunk links.

Router-on-a-Stick – A router that has been configured in such a way to allow it to move traffic for multiple VLANS through one physical interface (Uses subinterfaces).

Switched Virtual Interface (SVI) – A routed interface on a switch.

Network Time Protocol (NTP) – Synchronizes the time on a network device to a more accurate NTP device.

Notes

- Phones can be powered from a switch using the following methods
  - Cisco Inline power (Prestandard to PoE)
  - 802.3af PoE
- Cisco Inline Power process
  - Cisco prestandard PoE device physically connects to the switch
  - Switch sends a Fast Link Pulse (FLP) signal tone to the device. If the device supports the Cisco Prestandard it will loop the tone back.
  - If the switch sees the signal loop back it applies 6.3W of power to the line.
  - The device boots and communicates its actual power requirements to the switch using CDP.

    - If CDP is disabled, the switch automatically sends the maximum amount of power (15.4W).
- PoE (inline) are able to send from 0 to 15.4W of power
- PoE 802.3af
    - Uses a constant DC current to detect an unpowered PoE device
    - PoE devices use a resistor to send a small amount of current back to the switch
    - PoE power levels come in 4 classes (0-3)
    - 802.3af can send power over all 4 copper pairs, so it is compatible with Gig-Ethernet standards
- VLAN Benefits
    - Reduced broadcast domain size/increased performance
    - Improved manageability
    - Independent from physical topology
    - Increased security – Traffic between VLANS must pass through a layer-3 device
- On a trunk port the VLAN tag is appended to the end of a packet
- An IP phone forms a "mini trunk" with the switch
    - Packets on the voice VLAN are tagged
    - Data VLAN packets are part of the native (untagged) VLAN, which is set on a port by port basis
- Voice VLAN information is provided to an IP phone via CDP
    - Phone that do not support CDP will need to be configured with the VLAN information
- VTP Configuration
    - vtp mode [client|server|transparent]
    - vtp domain DOMAIN_NAME
    - vtp password <password>
    - show vtp status
- 802.1Q Trunk Configuration
    - Interface sub-configuration mode
    - switchport trunk encapsulation [dot1q|isl|negotiate]
    - switchport mode [access|dynamic|trunk]
- Inter-VLAN routing configuration on a router
    - No IP address on the physical interface (Create sub-interfaces)
    - Interface fa0/1.200
        - encapsulation dot1q
        - Set IP address
        - Set IP helper
- IP Phone Boot Process
    - Physical connection with a switch is made and power is provided (if applicable)
    - Switch delivers VLAN information to the phone using CDP
    - Phone sends a DHCP request on its voice VLAN
    - DHCP offer is provided, along with option 150 which has the IP address for the TFTP server

- TFTP server is contacted and configuration file downloaded (Including a list of call processing agents)
- The phone attempts to contact the first server in the list. It moves down the list if this process fails.

- Router DHCP Server Configuration
  - ip dhcp excluded-address <start address> <end address>
  - ip dhcp pool <pool name>
    - network <ip address> <mask>
    - default-router <ip address>
    - dns-server <ip address>
    - option 150 ip <ip address>
  - You can create a separate scope and use the “hardware address MAC_ADDRESS” command to reserve a specific IP address for a device
- NTP Configuration
  - ntp server <ip address>
  - clock timezone <name> <hours>
  - show ntp associations

Chapter 04

Key Terms

Survivable remote Site Telephony (SRST) – Configuration that allows a router to act as a failover if the IP phone cannot reach the CME.

IOS License – Feature set of the IOS version running on the router.

Feature License – Allows CME software to support a given number of IP phones.

Phone User License – License that allows each phone to communicate with CME.

Notes

- CME Maximum supported phones
  - 1861 – 8
  - IAD2430 – 24
  - 2801 – 24
  - 3250 – 10
  - 3270 – 50
  - 2811 – 35
  - 2821 – 50
  - 2851 – 100
  - 3725 – 144
  - 3745 – 192
  - 3825 – 175
  - 3845 - 250
- In order to run CME on a router a IOS version that supports CME is required
  - Feature license (Seat License) allows a certain amount of phones
  - A phone user license must be purchased for each IP phone supported by CME
- Line design models
  - PBX model gives most of the IP phones a unique extension
  - Keyswitch model provides shared numbers across all phones on the system
    - Each extension usually maps to a PSTN line
  - Hybrid model allows for shared lines, but does not require the use of PSTN lines for calls between extensions
- CME installation requires a series of files
  - Basic files – Core CME files including phone firmware
  - GUI files – CME web management interface files

- XML template file – Dictates the structure of the CME GUI. Allows different admin levels
- MOH files – Audio for music on hold
- Script files – TCL files that provide AA and ACD functions
- Miscellaneous files – Custom ringtones and backgrounds on phones

- Archive command is used to extract a tar archive of CME files
  - archive tar /xtract <source> <destination>
- IP Phone boot process
  - IP phone connects to the switch & receives power (if applicable)
  - Voice VLAN information is delivered to the phone using CDP
  - IP phone sends a DHCP request on its voice VLAN
  - IP phone accepts the DHCP offer and receives TFTP server information (option 150)
  - Phone contacts the TFTP server for firmware and configuration files (Including CME servers)
  - IP phone starts by trying to contact the first CME server. Moves down the list if this fails
- TFTP Server configuration
  - Global configuration mode
  - tftp-server <file location> alias <file name>
  - tftp-server flash:/phone/7940-7960/P0030800500.bin alias P0030800500.bin
  - show tftp-service tftp-bindings
- Configuring maximum phone and DNs
  - Telephony configuration mode (telephony-service)
  - max-ephones <number> (Physical phones)
  - max-dn <number> (Directory numbers)
  - These commands reserve memory on the router
- Configure CME what firmware to use for each phone model
  - Telephony configuration mode (telephony-service)
  - load <model> <firmware file>
  - load 7960-7940 P0030800500
  - This also creates the configuration file for the phones
    - Manually generate CNF files from Telephony configuration mode
    - create cnf-files
    - Should be done when a change is made that effects the phone boot process
- CME Source IP address
  - Used as the source address when CME communicates with phones
  - Telephony configuration mode (telephony-service)
  - ip source-address <ip address>
- Default phone configuration file
  - XMLDefault.cnf.xml is used for phone types that do not have a configuration
  - View configuration using “more system:/its/XMLDefault.xml”

Chapter 05

Basic CME IP Phone Configuration

Key Terms

Spanning Tree Protocol (STP) – Method a preventing layer-2 loops in a network.

Ephon-dn – CME configuration that represents a single directory number.

Ephone – Represents a single IP telephony device.

Feature ring – Causes the IP phone to ring with three consecutive pulse. Configured with the “f” separator.

Overlay Line – Allows shared lines by assigning multiple line instances to a single physical button. Configured using “o, c, or x” separators.

Monitor mode – Line instance that cannot be used for making or receiving calls, rather only monitoring the status of a line.

Watch mode - Line instance that cannot be used for making or receiving calls, rather only monitoring the status of a line.

Auto-registration – Allows an IP phone to register with CME, even if it does not have a configuration.

Auto-assignment – Allows distribution of ephone-dns to auto registered phones.

Notes

- Switch voice VLAN configuration
    - Interface sub-configuration
    - switchport voice vlan <vlan number>
    - spanning-tree portfast (disables STP on this port)
- Ephone-dn modes
    - Single-line ephone-dn – Only able to make or receive one call at a time
    - Dual-line ephone-dn – Able to handle two simultaneous calls. Supports call waiting, conference calling, and consultative transfers.
    - Single-line is the default mode, unless specified
- Ephone-dn configuration
    - ephone-dn <number> [dual-line]
        - number <dn> [secondary <number>]
- Ephones
    - Ephones are the CME equivalent of a physical (or soft) phone

    - ephone <ephone number>
        - mac-address <phome MAC>
- Ephone button assignment
    - Used to assign an ephone-dn to a physical button on an ephone
    - Enter ephone configuration
    - button <physical button> <separator> <ephone-dn>
    - button 1:2
    - The ephone should be restarted after changes using the restart command
- Ephone button separators
    - ":" – Normal ring
    - "b" – Call waiting beep, no ring
    - "f" – Feature ring (triple ring on incoming calls)
    - "m" – Monitor mode
    - "o" – Overlay line/no call waiting. Shared line experience
    - "c" – Overlay line/with call waiting
    - "x" – Overlay expansion/rollover.
    - "s" – Silent ring. Also disables call waiting beep.
    - "w" – Watch mode. Watches all lines on the phone for which the phone is the primary.
- Show ephone configuration
    - show ephone
    - shows phone information including MAC and button assignments
- Shared lines are handled by assigning the same ephon-dn to multiple ephones
    - You can create multiple dual-line ephone-dns with the same directory number for better flexibility
- Preference command can be used to create a preference for an ephon-dns with the same dn
    - preference <0-10>
    - Lower preference is better
- The huntstop command can be used when multiple ephone-dns with one dn are used on multiple phones
- Overlays allow you to assign multiple ephone-dns to one extension
    - Create multiple ephone-dns with the same dn
    - Use preference and huntstop commands
    - Create a button on each ephone: button 1o10,11 (10 and 11 are ephon-dns)
- Use "**#" on a physical phone to unlock the "erase configuration" option
- Debug commands
    - debug tftp events
    - debug ephone register
- By default CME supports auto-registration
    - show ephone attempted-registrations – Failed ephone registrations
- Auto assignment can distribute ephone-dns automatically
    - Telephony configuration mode (telephony-service)
    - auto assign <start dn> to <end dn> type <model>

- auto assign 20 to 24 type 7975
- Type argument is optional
- Use restart all command after configuring auto-assign

- System message is changed using the “system message” command

Chapter 06

Configuring Cisco Unified CME Voice Productivity Features

Key Terms

Local directory – Directory that is built as caller ID information is entered for ephone-dns

H.450.3 – Standard for forwarding calls without hairpinning.

H.450.2 – Standard for transferring calls without hairpinning.

Hairpinning – When a call is transferred or forwarded from a phone, but the audio path still travels through the original phone.

Call park – Puts a call on hold is a virtual parking slot for retrieval.

Call pickup – Allows for another ringing phone to be answered remotely.

Directed pickup – A form of call pick up where the extension of the running phone must be dialed in order to answer it.

Local group pickup – Call pickup form used to answer a ringing phone inside of a local group.

Other group pickup – Pick up a phone in another group. The group number must be entered after pressing the GPickUp softkey,.

Notes

- Local directory
  - Use the name command in an ephone-dn to add caller-ID information
  - directory command can be used to configure the display of the directory
  - directory entry command can be used to enter manual entries
- Call forwarding configuration
  - Use call-forward command under ephone-dn
  - Call-forward pattern command is used to configure number that support H.450.3 forwarding (avoid harpinning)
- Configuring call transfer
  - Consult – Speak with the party before transferring the call
  - Blind – Transfer without consult
  - transfer-system [full-blind|full-consult|local|consult] – System wide configuration
  - transfer-mode [blind|consult] – Ephone-dn configuration
  - Use transfer-pattern to allow transfers to non-locally managed numberss
- Call parking

- You can configure an ephone-dn as a park-slot using the park-slot command

- Call pickup
  - Assign an ephone-dn to a group using pickup-group <group number>
  - Directed pickup – Pickup another ringing phone by pressing PickUp key and dialing the DN of the ringing phone.
  - Local group pickup – Pick up a ringing phone in the same group by pressing GPickUp key and pressing "*".
  - Other pickup group – Answer a rining phone in another group by pressing GPickUp and entering the group number.
- Intercom
  - Intercom DNs should use a number that cannot be dialed from an IP phone (EX: A100)
  - Configure intercom as a ephone-dn
    - Number A100
    - Intercom <number> label "<label>"
- Paging
  - A phone can only be a member of one paging group
  - CME allows you to create paging numbers that page multiple groups
  - Configure a paging DN using the "paging" command
  - Configure phones to be part of a paging group: paging-dn <paging dn>
- Call block and exemption
  - Define after hours days and time using after-hours [block|date|day} command
  - Create an after-hours block pattern: after-hours block pattern <number> <patteren>
  - Configure after-hours exemption on an ephone-dn using the after-hours exempt command
  - You can also set a pin (pin <number>) for after-hours authentication
- Call Detail Records (CDR)
- Music on Hold (MOH)
  - Telephony configuration mode (telephony-service)
  - Configure MOH using moh <filename>
  - Multicast moh <multicast IP address> port <port number>
- CME GUI
  - Configure web server: ip http [server|secure-server]
  - Configure web server root: ip http path <path>
  - Configure web server location: ip http authentication <method>
  - Enter Telephony configuration mode (telephony-service)
    - Configure web admin login
    - web admin system name <username> secret <0|5> <password>
    - The dn-webedit and time-webedit commands allow adding of ephone-dns and editing of time via the GUI

Chapter 07

Gateway and Trunk Concepts

Key Terms

Nyquist theorem – The process of converting analog audio signals to digital format by sampling at twice the highest frequency of the audio.

Mean Option Score (MOS) – A subjective method of determining voice quality where listeners hear a phrase read over a voice network and rate it from 1 to 5.

G.711 – Uncompressed audio codec consuming 64 kbps.

G.726 – Compressed audio codec consuming 32 kbps.

G.728 – Compressed audio codec consuming 16 kbps.

G.729 – Compressed audio codec consuming 8 kbps.

Internet Low Bitrate Codec (iLBC) - Compressed audio codec consuming 15.2 kbps.

Real-Time Transport Protocol (RTP) – UDP based protocol responsible for the transport of audio packets. Uses random, even numbered UDP ports from 16,384 to 32,767.

Real-time Transport Control Protocol (RTCP) – UDP based protocol that is responsible for transporting audio statistics. Uses random, odd numbered UDP ports from 16,384 to 32,767.

Foreign Exchange Station (FXS) – Analog interface that connects to a legacy analog device and provides dial tone.

Foreign Exchange Office (FXO) – Analog interface that connects to a carrier CO or PBX and receives dial tone.

Ear & Mouth (E&M) – Analog interface type that acts as a trunk to a PBX.

H.323 – Protocol suite used to allow multimedia communication over network based environments.

Media Gateway Control Protocol (MGCP) – Voice signaling protocol that allows voice gateways to be controlled by a centralized call agent (client/server fashion).

Session Initiation Protocol (SIP) – Voice signaling protocol that was designed as a light-weight alternative to H.323.

Skinny Client Control Protocol (SCCP) – Voice signaling protocol used to control IP phones (Cisco proprietary).

Internet Telephony Service Provider (ITSP) – Provides VoIP trunk capability to the PSTN to provide a cost savings over traditional telephony service providers (TSPs).

Quantization – The process of assigning analog signals a numeric value that can be transported over a digital network.

Notes

- Nyquist frequency range
    - Desired to sample the 300-4000Hz frequencies
- G.729
    - G.729a – less quality, but more processor efficient
    - G.729b – VAD detection
- Calculating Bandwidth requirements
    - Determine the audio bandwidth required for the audio codec
        - Bytes Per Packet = (Sample Size * Codec bandwidth) / 8
        - Sample Size is in seconds (20-ms = .02)
        - Codec Bandwidth is in bps (64 kbps = 64000 bps)
    - Determine data link, network, and transport layer overhead
        - Data link
            - Ethernet: 20 bytes
            - Frame relay: 4-6 bytes
            - Point-to-point Protocol (PPP): 6 bytes
        - Transport
            - IP: 20 bytes
            - UDP: 8 bytes
            - RTP: 12 bytes
        - Every voice packet uses RTP, UDP, and IP or 40 bytes
    - Add any additional overhead amounts
        - GRE/L2TP: 24 bytes
        - MPLS: 4 bytes
        - IPsec: 50-57 bytes
    - Add everything together
        - Total Bandwidth = Packet Size * Packets per Second
        - Total Bandwidth (Bytes per second) * 8 = Total Bandwidth (bits per second
    - Subtract bandwidth savings measures
        - VAD
        - RTP Header compression
- Digital Signal Processors
    - Performs sampling, encoding, and compression on audio coming into a router
    - Voice Interface Cards (VIC) require DSPs to be functional
    - Minimum number of DSP chips required depends on the codec being used

- Voice activity detection (VAD) allows for detection of silence and saves on bandwidth
- High complexity codecs
  - iLBC
  - G.723
  - G.728
  - G.729(b)
- Trunking to other VoIP Systems
  - H.323 – One of the first voice signaling protocols. Allows voice video and data to transmit across ISDN connections as well as LAN environments.
  - SIP – Lightweight and scalable compared to H.323
  - MGCP – Configuration is done at a central location called the call agent
  - SCCP – Cisco proprietary and generally not used by gateways

Chapter 08

Configuring and verifying Gateways and Trunks

Key Terms

Dialed Number Identification Service (DNIS) – Term used to describe the dialed number information delivered to a voice processing device.

Automatic Number Identification (ANI) – Caller ID information that is delivered to a voice processing device.

Dial peer – Configuration used to define dial plan information on a Cisco router.

Foreign Exchange Station (FXS) ports – Analog interfaces that allow for connection of legacy devices to the VOIP network.

Foreign Exchange Office (FXO) ports – Analog interfaces that allow the VoIP system to connect to legacy telephony networks.

Private Line Automatic Ringdown (PLAR) – Configuration to enable immediate dial applications. Such as an emergency phone that dials 911 as soon as it is picked up.

Direct Inward Dial (DID) – Allows users of the PSTN to dial directly into an individual phone without going through an receptionist first.

Notes

- Use show voice port summary to view voice ports infomation
- T1 CAS Interface
  - controller t1 <module/port number>
  - framing [esf|sf]
  - linecode [ami|b8zs]
  - clock source [free-running|internal|line]
  - ds0-group <group number> timeslots <T1 timeslots> <type>
    - ds0-group 1 timeslots 1-24 fxo-loop-start
- T1/PRI CCS Interface
  - Isdn switch-type <switch type>
  - controller t1 <module/port number>
  - pri-group <group type>
    - pri-group timeslots 1-24
- Dial peers
  - POTS – Define reachability for traditional voice connections
  - VoIP – Define reachability for any VoIP connection through a device with an IP address

- Call legs represent connections to or from a gateway from a POTS of VoIP source
  - On one WAN connection there is an incoming and outgoing call leg (based on router perspective)
- Configuring POTS dial peers
  - dial-peer voice <DP tag> pots
  - destination-pattern <pattern>
    - Note: the router will strip any explicitly defined digits from a POTS dial peer
  - port <port number>
  - Debug using: debug voip dialpeer
- Configuring VoIP dial peers
  - dial-peer voice <DP tag> voip
  - destination-pattern <pattern>
  - session target ipv4:<next hop IP>
  - codec <codec>
- forward-digits <number> - The number of right-justified digits to forward
- prefix <number> - Adds the numbers to the front of the dialed numbers
- The preference command can be used in dial peer configuration to create failover links
  - Multiple identical dial peers with the exact preference will be chosen randomly

- Wildcards
  - "." – Matches dialed digits from 0-9 and "*"
  - "+" – Matches one or more instances of the preceding digit, up to 32 total digits
  - "[]" – Matches a range of digits. "^" before the range means does not match
  - "T" – Matches any number of dialed digits from 0-32 digits
  - "," – Inserts a one second pause between dialed digits
- PLAR
  - Configure under voice port
  - connection plar <digits to dial>
- How are inbound dial peers matched?
  - Dialed number DNIS using the incoming called-number command
  - Match called ID information (ANI) using the answer-address command
  - Match called ID information (ANI) using the destination-pattern command
  - Match an incoming POTS dial peer by using the port command
  - If no match is found using these methods, use dial peer 0
- Digit manipulation
  - The most specific destination pattern always wins
  - Once a match is found, the call is immediately processed
  - Test dial peer matching using show dialplan number <number>
  - [no] digit-strip can be used to enable or disable digit stripping
  - Num-exp <match digits> <set digits> takes the matched number and transforms it to the set digits
    - Num-exp 0 5000 transforms 0 to 5000

- Translation
    - Create a translation rule using voice translation-rule <rule number>
    - rule <number> <match> <set>
    - Create translation profiles with voice translation-profile <name>
        - Translate [called|calling|redirect-called|redirect-target] <translation rule>
    - Assign profile to a dial peer using translation-profile <direction> <profile name>
- Network requirements for voice
    - End to end delay – 150 ms or less
    - Jitter – 30 ms or less
    - Packet loss – 1% or less

Chapter 09

Cisco Unity Express Concepts

Key Terms

Cisco Unity Express – The unified messaging platform that is integrated into a CME router and capable of supporting up to 250 users.

Automated Attendant (AA) – Provides the ability to answer and direct incoming phone calls to the appropriate person without the need for human intervention.

General Delivery Mailbox (GDM) – A mailbox that is shared between a group of subscribers.

Cisco Unity Express greeting/prompt – A recorded message played to a caller.

Cisco Unity Express password – Used to authenticate the subscribers via GUI.

Cisco Unity Express PIN - Used to authenticate the subscribers via TUI.

Cisco Unity Express administrator – A subscriber that is a member of the administrators group.

Cisco Unity Express subscriber – A user account configured in Cisco Unity Express.

Cisco Unity Express Advanced Integration Module (AIM-CUE) – Entry level hardware supporting up to 50 users.

Cisco Unity Express Network Module (NM-CUE) – Lower-midlevel hardware supporting up to 100 users and 100 hours of recording.

Cisco Unity Express Network Module with Enhanced Capability (NM-CUE-EC) – Upper mid-level hardware platform providing support for up to 250 users and 300 hours of recording.

Message waiting indicator – Alerts the subscriber that a new message has arrived to the inbox by turning on a light.

Message notifications – Generate a call, send an email or page to the subscriber when a new message is received.

Live reply – Allows the subscriber to use the received caller identification number (ANI) and place a call to that caller during voicemail playback.

Live record – Enables recording of a live call. Call is delivered to the subscribers voicemail box.

Public distribution list – A collection of subscribers that is available to all Cisco Unity Express subscribers to use a distribution list.

Private distribution list – A collection of subscribers created by a single subscriber for use only by that subscriber.

Integrated messaging – Provides access to voicemail via an e-mail client.

Voice Profile for Internet Mail (VPIM) – Feature that allows one voicemail system to exchange messages with another.

Cisco Unity Express Auto Attendant Script – A collection of software scripts that defines the actions to be performed on a received call.

Cisco Unity Express Editor – A software application that is used to create custom Cisco Unity express custom scripts.

Administration via telephone (AVT) system – Allows the administrator to quickly record prompts and enable alternate AA greetings via a telephone.

Cisco Unity Express Graphical user Interface (GUI) – Provides subscribers with a web interface to manage Cisco Unity Express features and functions.

Cisco Unity Express Telephony User Interface (TUI) - Provides subscribers with a telephone interface to manage Cisco Unity Express features and functions.

Voice View Express – An XML application that allows subscribers to access their voicemail via the services button on an IP phone.

Mailbox subscriber features – Mailbox features that Cisco Unity Express offers to a configured subscriber.

Mailbox caller features – Mailbox features that Cisco Unity Express provides to a caller where the caller may not be a subscriber.

Notes

- Unity Versions
  - Unity Express - 250 mailboxes; Router based
  - Unity Connection – 7500 mailboxes; Win/Linux based
  - Unity – 7500 per server; Includes unified messaging
- Cisco Unity Express hardware platforms
  - AIM-CUE, NM-CUE, NM-CUE-EC, NME-CUE
- Simultaneous connections supported by CUE is determined by the voice ports and varies by hardware platform (6 to 24).
- CUE requires PC-based apps for custom scripting and historical reporting.
- CUE licensing is per mailbox and is detailed in the package name.
- General delivery mailboxes are used for a group of people.
  - Retrieval for GDM messages is through a prompt in the user VM access prompt

- Passwords are used for GUI management, PINs for AVT
    - Passwords and PINs can have the same level of security (Expiry, history, lockout, etc).
- Messages are played back to a user based on their type
    - Broadcast, expired, urgent, new, saved
- VoiceView Express allows for GUI management of VMs from the phone
- Integrated messaging provides access to VMs through an e-mail client (IMAP)
- Voce Profile for Internet Mail (VPIM) allows different VM systems to exchange messages
- Auto attendant greetings with default scripts
    - Welcome prompt
    - Business Open prompt
    - Business Closed prompt
    - Holiday prompt
- CUE management
    - Accessing the CUE command line from the router is used by accessing the service engine: service-module service-engine 1/0 session

Chapter 10

Cisco Unity Express Configuration

Key Terms

Service-engine – An interface that is created after the CUE hardware is installed into a router.

Service-module – The internal interface for CUE through which CUE will route calls to the SE for CUE to process.

Cisco Unity Express CLI – The command line interface used to configure and administer CUE.

Cisco Unity Express GUI – The interface that provides subscribers and administrators with a web page to manage CUE features and functions.

MWI – Method of alerting a subscriber that a new message has arrived to their mailbox.

Notes

- Use the “ip unnumbered <interface>” command to allow access to the service engine through another interface
- The service module should be configured with an IP on the same subnet as the SE
  - Configured under the SE interface
  - Service-module ip address <IP Address> <Mask>
  - Service-module ip default-gateway <IP Address>
  - Ensure there is a route so the router knows to route requests for the SM through the SE
  - Ip route <IP Address> <Mask> service-engine <number>
- Connecting to the CUE CLI is done using the following command: service-module service-engine 1/0 session
- Downloading CUE software
  - Requires and FTP server
  - The software download [server|clean|upgrade|status|abort} command is used for software installation
- Installing CUE software is done with the software install [clean|upgrade] url ftp://IP command
- Upgrade can be performed from version 2.3.4 to 3.1
- Features enabled by CUE licenses
  - Support for CUCM or CME
  - Number of supported General delivery and subscriber mailboxes
  - Number of IVR ports
- The show software command can be used to view versions and licenses
  - Show software [directory|download server|licenses|packages|versions]

- Post installation configuration tool configures hostname, domain, primary & secondary DNS/NTP servers, timezone, and admin credentials
- CUE dial peer s are used to route calls to CUE (such as when a user wants to check VM or use AVT)
    - The dtmf-relay sip-notify option allows for the relay of tones to CUE
- Telelphony service configuration
    - Voicemail <extension>
    - Dn-webedit (allows addition of DNs from GUI)
    - Time-webedit (allows time to be edits from GUI/do not use with NTP server)
- MWI dial peers
    - CUE sends a string of digits (followed by the extension) to turn the MWI on or off

Ephone-dn 19

Number A40….

Mwi on

-
    - CUE sends A409000 to turn the MWI for extension 9000 on.
    - Create another dial peer with the “mwi off” option
- Initialization wizard is the GUI version of post-installation
    - Allows import of CME users and management of CME through single GUI
    - VM number, operator ext, AA, and AVT options
- CUE module state
- Debugging & troubleshooting
    - Checking service engine: show interface service-engine 1/0
    - Service module stats: service-module service-engine 1/0 status
        - Shows state and the platform of CUE
    - Use ping and show route commands to ensure connectivity to service engine
    - Check dial peers: show dial-peer <extension>
    - Debug ccsip <option> to debug call processes
    - MWI debugging with debug ephone mwi
    - Trace files (debugs) are used by TAC

Chapter 11

Introducing the Smart Business Communications System

Key Terms

Unified Communications -

CCA – GUI used to configure, manage, troubleshoot, and maintain the SBCS.

Cisco Smart Assist – The wizards in the Cisco Configuration Assistant the helps with configuration and maintenance of the SBCS suite.

Voice expansion port – Voice/WAN interface card (VWIC) that allows for PSTN expansion in all UC520 models (Does not support WAN connectivity).

LAN expansion port – Port that is automatically configured as the external port (Cable or DSL) on the UC520.

Notes

- Suite is designed to make UC more affordable and easier to roll out
  - UC500
  - Catalyst Express switches
  - 521 WAP
  - 526 WAP Controller
  - CCA
- Typical configuration by automatically deploying extensions, configuring dial plans, configuring VLANS, firewall policies, and best-practice QoS
- Base configuration in all models
  - Console port
  - 3.5mm MoH port
  - 8-port PoE switch
  - 4 FXS ports
  - WAN port
  - 1Gb LAN Expansion port
- UC520 available in 16, 24, 32, and 48 user configurations
- WAP support and expansion
  - Up to three 521 WAPs in standalone mode
  - Requires a 526 WAP controller which can support up to 6 WAPs each
    - CCA can support up to 12 WAPs
    -
- No routing protocol support

- Beware of PoE limits to integrated switch

Chapter 12

Configuring and Maintaining the UC500 Series for Voice

Key Terms

Community – A group of devices managed by CCA through IP.

Cisco Discovery Protocol (CDP) – Protocol that allows devices to discover one another as well as allowing switches to send voice VLAN information.

Smartports – CCA macro that aids in the configuration of individual ports.

Key System – A private voice system where all phones usually share extensions.

Direct Inward Dial (DID) – Allows PSTN users to reach a phone directly without using an AA or receptionist.

Power failover (PFO) – Allows the UC520 to place calls using an analog phone to the PSTN without power. Bridges the FXO and FXS ports.

Notes

- Power failover feature manually bridges the FXO and FXS port to allow an analog phone to function.
- IP defaults
    - Voice VLAN: 10.1.1.0/24
    - Data VLAN: 192.168.10/24
- Topology view is built using CDP
- Default extension layout is three digits in the 200 range