Post 1

Thursday, January 24, 2019 11:49 PM

From <https://www.piratemoo.net/moosings/voip/voip-ch1-notes/>

VOIP: CH1: NOTES

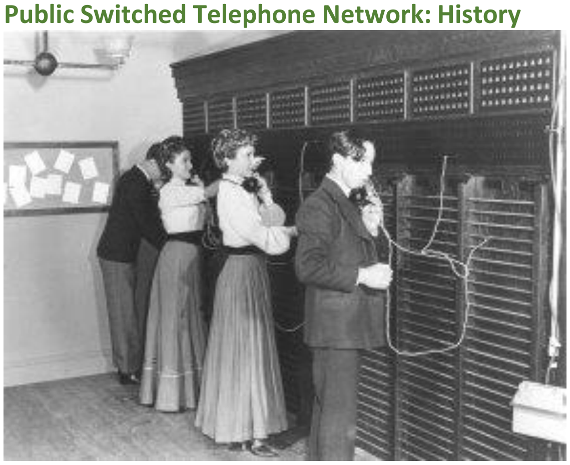
January 26, 2017 Moo Comments 1 Comment

Our books didn't go into much detail regarding any of the history related stuff, because they're primarily lab focused and Cisco based. However, I decided to research the slides a little bit more in order to provide more insight for myself (and hopefully others checking this out) on these basics. I have to say, I'm super glad I did too! Enjoy!

PSTN	**Public Switched Telephone Network:** Variety of phone networks/services worldwide • AKA **POTS: Plain Old Telephone Service** • Voice-oriented telephone networks that shares circuits through packet-switching **Consists of:** • Telephone lines (also undersea phone cables) • Fiber optic cables • Microwave transmission links • Cellular networks • Communications satellites **Interconnected by switching centers:** Allows phones to comm w/each other

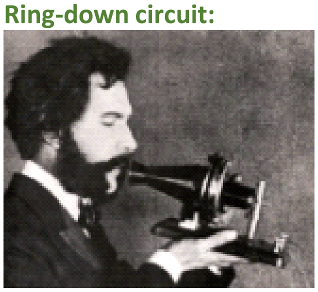
	• Originally fixed-line network with analog telephone systems: Now mostly digital **Adheres to standards created by ITU-T (International Telecommunication Union)** • Coordinates w/all entities involved w/standards in telecomm industry • 1865: Became specialized agency under UN: 1947 • 1993: Renamed ITU-T • E.163/164: Standards provide single global address space for phone numbers
POTS	**Plain Old Telephone Service** • Basic phone service supplying standard • Single-line telephones, telephone lines and access to PSTN
CO	**Central Office** • Local phone company office where all local loops in a given area connected to • Circuit switching of subscriber lines **CO switch:** Terminals local loop/makes initial call-routing decision
Local Loop	Interface to phone company network • Typically single pair of wires that carry a single conversation • Home/small business may have multiple local loops
Trunk	Providing path between 2 switches

Public Switched Telephone Network: History



1875: Alexander Bell: Formed American Bell Telephone Company: Dubbed as liquid transmitter originally
1876: Alexander Bell patented 1st improvement in telegraphy: First voice transmission over wire

Ring-down circuit:



First voice transmission: No number dialing/ringing

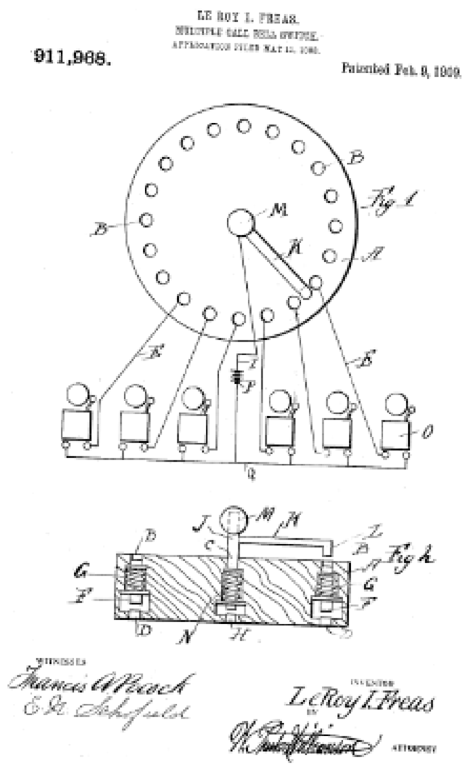
- Physical wire connected 2 devices: Like tin can phone as kid
- Usrs whistled into phone to speak
- Bell added year later to make signaling easier

Moving voices across the wire:

Carbon microphone	**Battery**	**Electromagnet**	**Iron diaphragm**	**Physical cable bet each loc**

- A needle was wired to a battery: Movement varied str of current passing bet contacts
- Converted sound waves into an electric signal: Travelled along wire to receiver

Wasn't scalable: Each phone wired to local phone exchange wired w/trunks: Hierarchical until expansion
The 1st switch: A few years later, Bell developed a way to map any phone to another w/out direct connections: The switch



- With switched telephones users only needed a connection to a **centralized office (CO)**
- CO would bring call to its destination: Much more scalable

Operators: Tasked with building networks/selling services to customers: Bell Telephone Company

- Many companies were monopolized w/governments, although these monopolies eventually changed (AT&T)

Exchanges: Operators (originally teenage boys, then later mostly women) handled calls

    - Operators had large switchboards w/2 pin connection sockets (jack socket) for every pair of wires

- Users would ring operator: Give name/number of party
- Operator would connect patch cord between 2 phones for comm: Bell would ring
- “Human switches” or operators later evolved into analog switched: Then electronic ones

Automation: Introduced pulse dialing bet phone/exchange: Address signaling/multi-frequency: SS7

- AK Erlang established mathematical foundation method required to determine capacity/config reqs of equip for QoS

1970s: Telecomm industry implemented packet-switched data services via X.25 protocol

1980s: End-to-end circuit switched services: **B-ISDN: Broadband Integrated Services Digital Network**

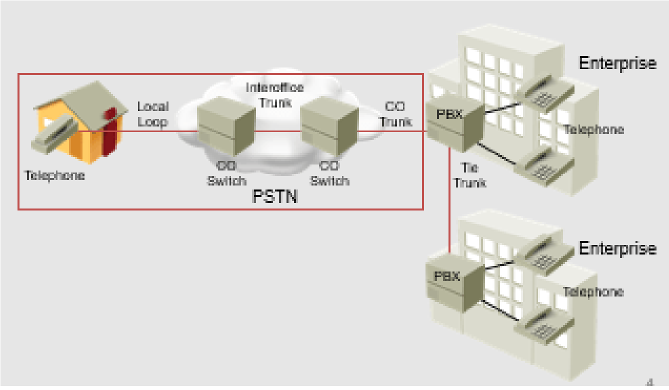
- Oldest phone networks still use analog for last mile loop to end usr

- DSL/ISDN/FTTx/cable modems common

Analog: Voice signals are carried across the wire w/amplifiers and eventually evolved into digital signals carried w/repeaters

Repeater: Repeats whatever binary data it receives

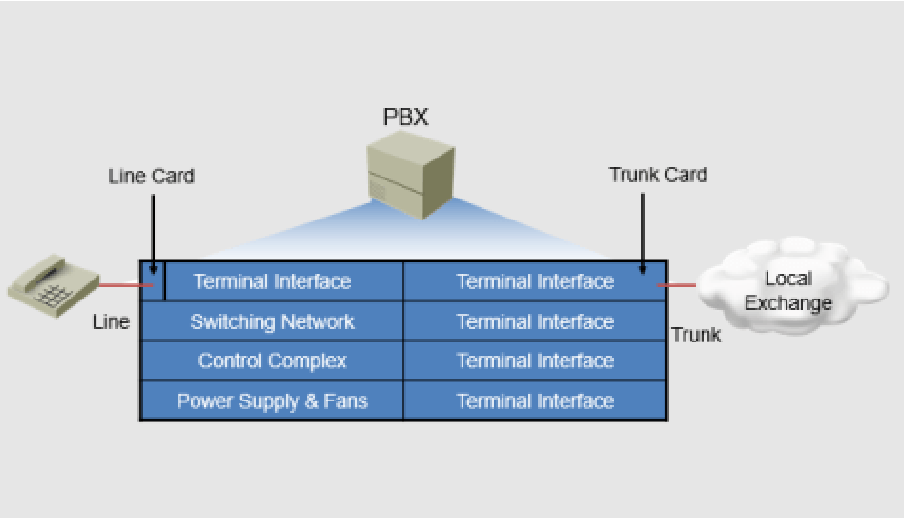
VoIP: Not transmitted over a circuit-switched network, but over a packet-switched one

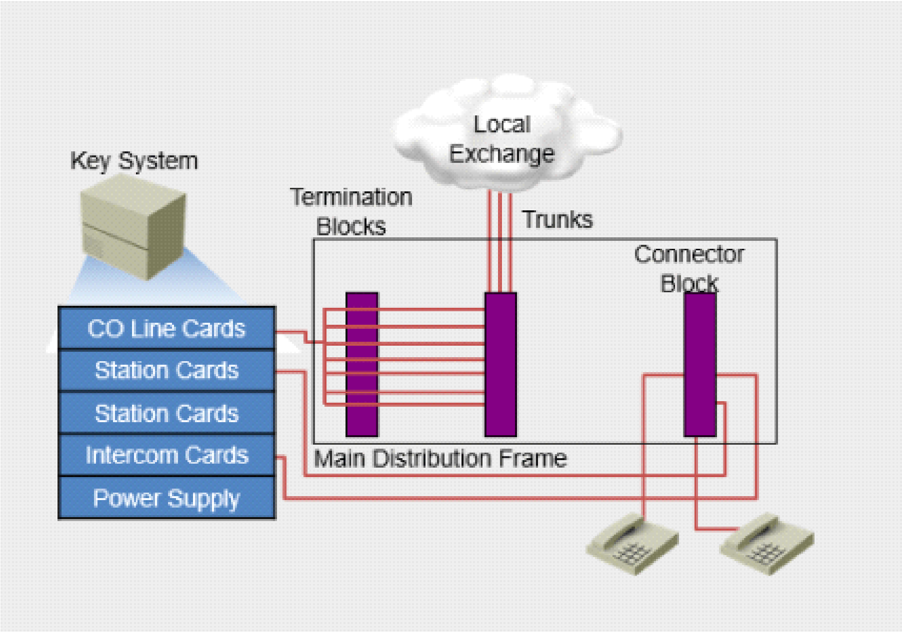


What is a PBX?

PBX	**Private Branch Exchange** • Digital/analog phone switches • Located w/customer: Used to connect public/private phone networks • Smaller/Privately owned version of a CO switch used by phone companies • Originally analog: Today: Digital • Operates in a manner similar to the public phone system **Why?** • Businesses install PBX's to reduce # of phone lines needed to be leased from the phone company • Otherwise you would have to rent 1 line for everyone with a phone • Only need to rent as many lines from provider as the max # of staff making external calls at one time **Every phone in a business is wired to the PBX** • Usr picks up phone/dials outside access code (usually 9) • PBX connects person to outside line and onto PSTN
Key System	Defined by their individual line selection buttons for each connected phone • Distinguished from PBX • Allows station usr to see/control calls directly • Buttons "light" up to indicate when certain lines are in use • Traditionally used by companies w/less than 50 employee • To place call: Press button to select phone company's CO lines • Usually 1 unit that as controller for a limited # of lines/extensions

PBX vs. Key System





	PBX	Key System
Technology	Primarily Digital	Analog or digital
Switch Functionality	Similar to CO switch	Not a switch
Typical Installation	Large company (50+ usrs)	Small company (less than 50 usrs)
Method for Accessing Outside Trunks	Dial 9/other access #'s	Press a button

Signaling Types

3 types of signaling used in telephony network:

1. **Supervisory:** Comm's state of telephony device
2. **Address:** Sends info about digits dialed
3. **Informational:** Comm's current state of call

Signaling: Can be sent in-band/out-of-band:

- **In-band:** Sends signaling in same communications channel as voice
- **Out-of-band:** Sends signaling in separate communications channel from voice

A call placed from residential phone uses all 3 types

- <u>When you lift handset:</u> Switch in phone closes to start current flow
- Notifies phone company you want to make a call (**supervisory**)
- <u>Phone company:</u> Sends dial tone to indicate it's ready to receive dialed digits (**informational**)
- User dials digits by pressing #'s on keypad (**address**)

Address Signaling:

2 types:

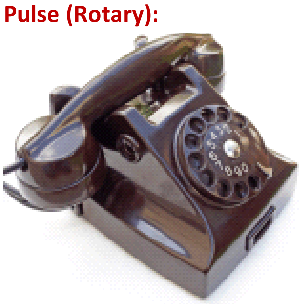
1. **Dual tone multifrequency**
2. **Pulse**

Dual tone multifrequency:

DTMF: Each button on keypad of touch-tone pad/push-button phone associated w/set of high/low frequencies.

- **Keypad:** Each row of keys ID'd by a low-frequency tone
- Each column is associated w/a high-frequency tone
- The combo of both tones notifies phone companies the # being called: Thus the term

Pulse (Rotary):



Pulse: Large numeric dial-wheel on a rotary-dial phone spins to send digits to place call

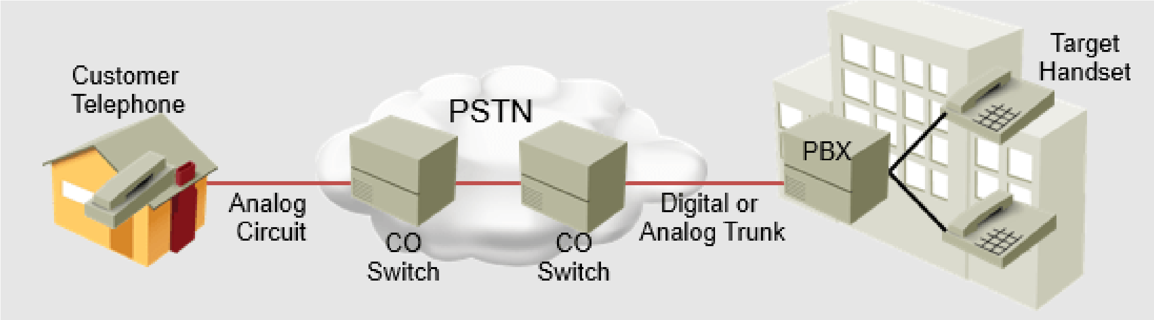
- Must be produced at specific rate w/in a certain lvl of tolerance
- Each pulse consists of a "*break*" and "*make*" (achieved by opening/closing local loop circuit)
  - **Break segment:** Time the circuit is open
  - **Make segment:** Time the circuit is closed

Signaling System 7: SS7

SS7	**Signaling System 7** • Standard CCS: Common Channel Signaling system used w/ISDN • 1975: Developed by Bellcore • Used between phone companies • Only international protocol defined by ITU-T's Q.700-series recommendations in 1988 **SIGTRAN:** IETF defined lvl 2/3/4 protocols compatible w/SS7: Suite of protocols called SIGTRAN • Use **SCTP: Stream Control Transmission Protocol** **SS5/Earlier systems:** • Used in-band signaling • **Bearer channels:** Call-setup info sent by playing special multi-frequency tones into phone lines ○ Directly accessible by users **Exploited w/devices like the blue box** • Played tones required for call control/routing • SS6/SS7 implemented out-of-band signaling: Carried in a separate channel: Keeping speech path separate **Functions:** • Informational signaling • Call setup • Call routing • Call billing • Toll-free number resolution • Uses out-of-band signaling • SMS: Short Message Service
CCS	**Common Channel Signaling** • Signaling system used in phone networks • Utilizes statistical multiplexing protocol

	• Specified channel is exclusively designated to carry signaling info for all chans on sys Example: SS7

PSTN Call Setup



1. Customer phone goes off hook creating closed circuit
2. Customer's CO switch detects current is flowing: Generates dial tone to phone
3. DTMF/Pulse digits dialed by customer
4. CO switch collects digits/performs an SS7 lookup: Lookup determines destination CO switch
5. Supervisory signaling indicates to the far-end analog/digital trunk that inbound call arrived
6. PBX determines which internal extension call should go to: Causes target handset w/ext to ring
7. Ringback generated to customer phone by their local CO switch
8. Target handset goes off hook: Circuit is built end-to-end

Understanding Traditional Telephony: Recap

Traditional telephony network composed of:

- PSTN, PBXs, key switches, signaling, call setup and numbering plans

Placing call through PSTN can involve:

- Analog/digital circuits, CO switches and interoffice trunks

PBX: Used in larger installations: Similar to CO switch

Key systems: Used at smaller sites: Have fewer features than PBX

- Usrs have shared line appearances on all phones

Supervisory signaling: Comm's state changes in an analog phone/digital handset

Address signaling: Comm's dialed digits using DTMF/pulse

Informational signaling: Communicates with the caller or called party

Post 2

Thursday, January 24, 2019 11:50 PM

ANALOG/DIGITAL CIRCUITS WITH VOIP: CODECS/PROTOCOLS

February 22, 2017 Moo Comments 0 Comment

Analog Circuits: Include FXS, FXO and E&M circuits: Resistors/capacitors/inductors/diodes/transistors/op amplifiers

- Susceptible to noise: variations in voltage

Analog telephones: Most common in home/SOHO/Small business envs: Direct connection to PSTN usually made

Receiver/Transmitter	2-wire/4-wire hybrid [black box converter]	Dialer: DTMF/Pulse

- Copper/POTS phones: Reliable/good voice quality
- Typical func: Hold/mute/redial/speed dial/extensions: Limited scalability

FXO/FXS: Always paired: W/out a PBX: Phone is connected directly to FXS port provided by phone company

FXS: Foreign Exchange Station: FXS Interface: Connects directly to standard phone/supplies ring/voltage/dial tone

- Unit used at subscriber station end of foreign exchange circuit: RJ-11
- Allows connections to: Phones/faxes/key sets/PBX's/Provisions local service
- Port that actually delivers analog line to subscriber: *"Plug on the wall"*

FXO Interface: Foreign Exchange Office: Port that receives analog line: Connects to CO/Station/PBX: RJ-11

- Plug on phone/fax/equip on analog phone system
- Delivers on-hook/off-hook indication (loop closure)
- Connects directly to office equipment
- Used to make/receive calls from PSTN
- Can be used to connect through PSTN to another site
- Answers inbound calls

Summary
FXS: Ports simulate CO to an analog phone/fax that is attached to port **FXO:** Ports connect Cisco voice gateway to CO switch or analog port on PBX **Analog circuits:** FXS/FXO/E&M circuits

Digital Circuits: Digitizing Analog Signals

1. Sample analog signal regularly
2. Quantize sample
3. Encode value into bin expression
4. Compress the samples to reduce BW

3 components to analog-to-digital conversion process:

Sampling	Sample analog signal at periodic intervals: Output of sampling is a **PAM: Pulse Amplitude Modulation** signal

	• Sampling rate must be 2x highest freq to produce playback that appears neither choppy/too smooth
Quantization	Match PAM sig to segmented scale: Measures height of PAM sig/assigns integer # to define that amplitude • Scale made up of 7 divisions/chords: Each cord subdivided into equally spaced steps
Encoding	Convert integer base-10 # to bin #: Output of is a bin expression in which each bit is either a 1: Pulse/0: No pulse

To digitize analog signal: Samples must be taken regularly/quantized to bin value/may optionally be compressed

- T1/E1 circuits are most common digital circuits

Understanding Packetization:

DSP	**Digital Signal Processor:** Chip performs sampling/quantization/encoding/compression of digitization • Electronic circuit compresses voice signals/generates tones/decodes received compressions • Can emulate modems for fax relay • Used in both directions to convert from analog/digital voice sig to VoIP: Or vice versa • # of simultaneous calls a chip can handle depends on type of DSP/codec being used **Additional DSP Functions**: Conferencing/Transcoding bet 2 diff codes/Echo cancellation **Cisco rtrs:** DSP's implemented through PMDM modules • Critical to Cisco Unified Comm System: Translate traditional voice data to IP packets/back **PVDM module: Packet Voice DSP Module:** Provides digital sig processing resources to a sys • Also performs compression/voice-activity detection/jitter mgmt/echo cancellation to improve voice

RTP: Real-Time Transport Protocol: Commonly used w/IP networks:

- End-to-end transport funcs for apps transmitting real-time: Audio/video/simulation over multicast/unicast network services
- Services such as payload type ID/seq #'ing/time stamping/delivery monitoring to real-time apps
- RTP is used to carry voice and video data across the IP network, and RTCP is used to provide feedback on the RTP stream.

Voice packaged into RTP segments:

- Segments encapsulated into UDP
- UDP segments encapsulated into IP packets
- IP packets encapsulated into specific L2 they traverse

Payload Type	Sequence Number	Time Stamp	Payload

Randomly picks even ports from UDP: Port range 16384–32767

RTCP: RTP Control Protocol: Monitors QoS of IP RTP connection: Conveys info about ongoing session

- Can monitor quality of data/provide control info: Feedback of current network conditions

Allows hosts involved in RTP session: Exchange info about monitoring/controlling it:

- Packet count/delay
- Octet count

- Packet loss
- Jitter (variation in delay)
- Separate flow from RTP for UDP transport use

Paired w/RTP stream: Uses same port as RTP stream +1 (odd-numbered port)
Packetization:

L2 Header	IP Header	UDP Header	RTP Header	Voice Payload

How it works:
Packetization voice: Performed by DSP

- DSP packages voice samples/compressed voice into IP packets
- Voice data collected until packet payload full
  - Carries payload of RTP segments
- Encapsulated in UDP segment: Into IP packet: L2 fmt

Common Codecs: Most common codecs? G.711/G.729/iLBC
G.711: PCM: Pulse Code Modulation: Commonly used waveform codec: Narrowband audio: Toll-quality audio 64Kbit/s

- Uncompressed high quality voice

G.729: Standard for IP PBX (Private Branch Exchange) vendors; as well as PSTN

- Digitizes analog voice sigs producing output at 8Kbps
- Uses 8:1 compression
- Alg: **CS-ACELP: Conjugate-Structure Algebraic Code-Excited Linear Prediction**
  - Compresses digital voice in packets of 10 millisecond duration
- Uses less BW at sacrifice of quality b/c of compression
- Now that high BW connections inexpensive/avail: SIP buyers no longer need to worry about this
- Mostly used in VoIP apps where BW has to be conserved
- Extended w/various features: Commonly designated as G.729a/G.729b

iLBC: Internet Low Bitrate Codec: Free speech codec for robust voice comm over IP:

- Less BW then G.711 for similar voice quality
- Designed for narrow band speech
- Payload bit rate of 13.33 Kbit/s w/encoding length of 30 milliseconds or 15.20Kbp/s w/20 millisecond

Comparison Chart:

Codec	G.711	iLBC	G.729
BW not including overhead	64Kb/s	13.3Kb/s	8Kb/s

VoIP Signaling Protocols:
Signaling generates/monitors call control info bet 2 endpoints to:

Establish connection	Monitor connection	Release connection

Must pass supervisory/info/address signaling

1. **Supervisory:** Comm’s state of telephony device
2. **Address:** Sends info about digits dialed

3. **Informational:** Comm's current state of call

Can be peer-to-peer or client/server based:

Peer-to-peer	Allows endpoints to contain intelligence to place calls w/out assistance
Client/server	Puts endpoint under control of centralized intelligence point

VoIP: Enables rtr to carry voice traffic [phone/fax] over IP network

- DSP segments voice signal into frames: Then frames stored in voice packets
- Voice packets transmitted using variety of signaling protocols

H.323: Stable/mature/vendor-neutral protocol widely deployed: Recommendation from ITU-T: Standard for interoperability

- Developed to promote compatibility in videoconference trans over IP
- Provide consistency in audio/video/data trans if LAN didn't have guaranteed QoS
- Addresses call control/mgmt for both point-to-point/multipoint conferences/gateway admin of media traffic/BW

H.3x: H.323 part of this larger group for multi-media interoperability

SIP: Session Initiation Protocol: Standard for initiating interactive usr sessions that involve multimedia

- L7: Establishes multimedia sessions/Internet telephony calls/modify/terminate
- Can invite participants to unicast/multicast sessions that don't involve initiator
- Name mapping/redirection: Possible for usrs to initiate/receive comm/services from any loc
    - Networks can ID usrs wherever they are
- **Request-response protocol:** Deals w/requests from clients/responses from servers
- Participants ID'd by SIP URLs
- Reqs sent through any transport protocol [UDP/SCTP/TCP]
- Determines end sys used for session: Comm media/params

SCCP: Skinny Client Control Protocol: Proprietary: Network terminal control protocol: UDP

- Originally dev by Selsius Sys: Acquired by Cisco in 1998
- Lightweight IP-based for session signaling w/Cisco Unified Comm Manager and Cisco Unified IP Phones
- End stations in network, VoIP/machines w/VoIP capability: Run program called Skinny Client
    - Helps min cost/complexity

Comparison Chart:

Protocol	Used on Gateways	Used on Cisco Unified IP Phones	Architecture
H.323	Yes	No	Peer-to-Peer
SIP	Yes	Yes: Cisco Unified IP Phones/3rd Party Phones	Peer-to-Peer
SCCP (Skinny)	Yes: Limited	Yes: Cisco Unified IP Phones Only	Client/Server

From <https://www.piratemoo.net/moosings/voip/analogdigital-circuits-with-voip-codecsprotocols/>

Post 3

Thursday, January 24, 2019 11:50 PM

CH.3 VOIP PHONES: CONFIGS

February 22, 2017 Moo Comments 0 Comment

Preparing Network to Support Voice: Phones segmented in separate logical networks: Network segmentation/control

- Allows admins to create/enforce QoS: Add/enforce sec policies

Voice VLANs: Separates voice/data traffic: Unnecessary IP renumbering: Simplifies QoS configs

Reqs 2 VLANs: 1 etho cable drop for Cisco IP Phone/PC plugged into phone: 2 subnets: 1 for data/1 for voice

1. Data traffic
2. Voice traffic

Access port can handle two VLANs: Access/Voice

- Access used for PC plugged into phone
- Voice used for voice/signaling: Originates/terminates on phone
- Spanning-tree PortFast mode: Causes STP to enable port quickly

Config Voice VLANs
S1(config)# int fa0/1
S1(config-if)# switchport access vlan 12
S1(config-if)# switchport mode access
S1(config-if)# switchport voice vlan 112
S1(config-if)# spanning-tree portfast

Verify:
S1# sh int fa0/17 switchport

DHCP: Assigns IP addresses/subnet masks for 1/more subnets: Assigns default gateway/DNS servers (optional)

- Needs to be customized to assign TFTP server to voice VLAN that IP phone on
- Config separate DHCP scope for IP phones as best practice

Config DHCP
R1(config)# ip dhcp excluded-address 10.112.0.1 10.112.0.10
R1(config)# ip dhcp pool mypool
R1(dhcp-config)# network 10.112.0.0 255.255.255.0
R1(dhcp-config)# option 150 ip 10.112.0.1
R1(dhcp-config)# default-router 10.112.0.1
R1(dhcp-config)# dns-server 10.100.0.1 10.100.0.2

option 150: Informs IP phone of TFTP server address
TFTP server: Contains config files/firmware on phone

default-router IP Sets default gateway assigned to DHCP clients
dns-server primary-IP [2ndary IP] Sets DNS server[s] are assigned to DHCP clients (optional)

Phone Bootup:
IP phone powers on/performs POST/boots: Phone uses CDP: Cisco Discovery Protocol to learn voice VLAN/initializes IP stack
- IP phone sends broadcast req IP address
- DHCP server selects free IP from pool/sends: W/other options
- IP phone initializes: Applies IP config to IP stack
- IP phone requests config file from TFTP server defined in option 150
- Config file contains: IP of call agent to register to

NTP: Network Time Protocol:
- Correct clock sync impt for performance/troubleshooting/CDRs
- Cisco devices: Internal sys clock: Set from # of sources [internal calendar sys/NTP]
- Allows devices to sync to clock master
- Local NTP server can have attached clock/sync w/more authoritative source
- Free NTP servers avail

IP phone gets displayed time from call control platform
- Cisco Unified Comms Manager
- Cisco Unified Comms Manager Express
- Time of internal clock of Cisco Unified Comms call control platform should be sync w/NTP server
- Time of Cisco Unified Comms call control platform is used to stamp all syslog/trace msgs

Config
R1(config)# clock timezone zone hours-offset Sets local time zone
R1(config)# clock summer-time zone recurring [start-date end-date] Specifies daylight savings time
R1(config)# ntp server IP Allows clock on router to be sync w/specified NTP server

Preparing Network to Support Voice Summary
- Voice VLANs used to separate voice/data traffic: Config on ints of switch IP phone connects to
- NTP: Allows sync to CUCME rtr to single clock on network
- IP phone reqs firmware/config/lang when boots
- Uses TFTP DHCP option 150 to DL config file: Needed to register w/call control device
  - Uses its MAC address as part of created filename which ID’s phone
  - Config file contains vers of firmware to use/IP/port phone will register w/

Introducing CME: Cisco Unified Comms Manager Express (CME)

Features/Benefits	Supports deployments of up to 240 phones on single router • Extended capabilities to small office: Before only avail to larger enterprises • Based on IOS: GUI/CLI
Supported Platforms	Cisco 2800 & 2900 Series: **ISR: Integrated Services Rtrs** Cisco 3800/3900 Series Cisco Unified Comms 500 Series for Small Business

Global Telephony Cmds: At min rtr needs to know:
- Max # of phones allowed
- Max # of phone numbers to be assigned
- IP rtr uses to respond
- Phones need a default template file created

telephony-service Enters CME global config
no auto-reg-ephone Optional: Prevents problems w/phones auto registering
max-dn 12 Max # of extensions
max-ephone 8 Max # of phones
ip source-address 192.168.0.1 port 2000 Assigns address for rtr to respond to phone reqs
create cnf-files Builds XML template files for phones

Ephone: Physical IP phone: Must have:
- MAC assigned before anything else
- ephone-dns(s) can be tied to phone using button cmd: More than 1 ephone-dn can be used on a phone

ephone-dn
- Dir # [phone #/extension]
- # of extensions: Limited by rtr model and max-dn cmd
- Must have dir # assigned before anything else [PT only supports dir #'s]
- ephone-dns Single-line/dual-line/octal line

ephone-dn 1 Creates dir # 1
number 1000 Assigns port # 1000
ephone-dn 2 Creates dir number #2
number 100 Assigns # 1001 to dir #2

Ephone/Ephone-dn Concepts: Modular IOS SW constructs
Ephone: Represents config/setting of phys phone
- Max # of supported ephones determined by license/HW platform: CUCME supports max of 240
- SW config of phys phone
- Assigned unique phone-tag
- Phys device can be IP/analog phone attached to an ATA
- MAC address of IP phone/ATA used to tie SW config to HW
- Can associate 1/more ephone-dns w/an ephone
- # of line buttons varies based on model of phone

ephone 1 Creates phone 1: Creates ephone instance/enters subconfig mode
mac-address 0000.0000.0000 Assigns MAC address of phone: Associates

phys device w/ephone
button 1:1 Assigns line button 1 to dir 1

- **button 1:2** Assigns phone line button 1 to dir #2 (extension 1001)

R1(config-ephone)# button *button-number* {separator} **dn-tag [[button-number** {separator}**dn-tag]]**

- Associates ephone-dn(s) w/specific buttons(s) on IP phone

Button Separators

-:	**Normal ring**
-b	**Beep but no ring**
-f	**Feature ring**
-s	**Silent ring**

Ephone-dn: Numeric destination that can be associated w/1/more ephones

- Can have more than 1 ephone-dn associated w/it
- Max # of extensions same as max # of ephone-dns

Ephone-dn Features	Has a primary dir # assigned to it/can have an optional 2ndary # • dn-tag Unique value assigned when ephone-dn created • Can be single/dual line ○ **Single line:** Can terminate 1 call at a time ○ **Dual line:** Can terminate 2 simultaneous calls • Packet tracer only supports single line dn's

Config
R1(config)# ephone-dn dn-tag Creates an extension (ephone-dn) for a Cisco IP phone line
R1(config-ephone-dn)# number dn-number Associates dir # w/ephone-dn instance

Example:
R1(config)# ephone-dn 7
R1(config-ephone-dn)# number 1001 Assigns primary extension # to an ephone-dn

R1(config-telephony)# max-dn 10 [max-dn #]

- Sets max definable # of ephone-dns that can be config in sys
- Max # of supported ephone-dns is a function of license/HW platform
- Default: 0
- Make most efficient use of mem: Don't set param higher than needed

R1(config-telephony)# max-ephones 4 [max-ephone #]

- Sets max definable # of ephones that can be config in sys
- Max # of supported ephones function of license/HW platform
- Default: 0
- Make most efficient use of mem: Don't set param higher than needed
- Attempts to create 5th ephone-dn will fail

Example: Basic config

R1(config)# ephone-dn 7
R1(config-ephone-dn)# number 1001
R1(config)# ephone 1
R1(config-ephone)# mac-address 000F.2470.F8F8
R1(config-ephone)# button 1:7

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Post 4

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VOIP CH 4 (CONDENSED CHEAT SHEET STUFF)

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Configuring CME to Support Endpoints

R1(config)# telephony-service Enters telephony-service mode
R1(config-telephony)# max-ephone *max-ephones* Sets max # of ephones that may be defined in sys: Default: 0
R1(config-telephony)# max-dn *maximum-directory-numbers* Sets max # of ephone-dns that may be defined sys: Default: 0
R1(config-telephony)# ip source-address *IP* [port *port*] ID's addr/port which IP phones comm w/CUCME
Auto Registration
R1(config-telephony)# auto-reg-ephone Enables auto registration of new ephones not in config/enabled by default

IP Phone Firmware/XML Config Files: Certain files are necessary for proper op of a Cisco Unified IP phone:

- Firmware
- XMLDefault.cnf.xml
- SEPAAAABBBBCCCC.cnf.xml (A/B/C is MAC addr of device)

R1(config-telephony)# create cnf-files Builds specific XML files necessary for IP phones

Automated Deployment of Endpoints:

- Automated setup: Don't have to config ephones
- Automated deployment of IP phones\
- **auto assign** [telephony-service config] Performs auto assignment
- All ephone-dns you want to be deployed: MUST be same type (single-line/dual-line)

R1(config-telephony)# auto assign *start-dn* to *stop-dn* [type *phone-type*]

- Ephone-dns auto assigned to new ephones that are config
- Phones can take up to 5 min to register
- Wait for all phones to register before saving config

Example:
R1(config)# telephony-service
R1(config-telephony)# auto assign 1 to 10 type 7920
When new IP phone registers w/CUCME sys: New ephone created w/MAC of IP phone

- Existing ephone-dn assigned to new ephone from range defined for type of phone
- Lowest unassigned ephone-dn in matching statement range used
- If all ephone-dns in a range assigned: Some phones may not receive an ephone-dn
    - May receive an ephone-dn from auto assign cmd w/out type
- If new IP phone doesn't match any auto assign cmd w/type: auto assign w/out type used

Verify: **sh run**
telephony-service
max-ephones 10
max-dn 10
ip source-address 10.90.0.1 port 2000
auto assign 1 to 10
create cnf files
!
ephone-dn 1
number 9000
!
ephone 1
mac-address 0000.0000.0000
button 1:1

Dial Peers and Destination Patterns: Gateways

- Translate bet diff networks: Req DSP resources to perform translation

Can be analog gateways: Analog station | Analog trunk | **Can be digital gateways**

Gateway: H.323 term: Describes component of H.323 telephony network that translates bet 1 tech/another

- Typically bet traditional telephony/TCP/IP

Voice Ports

Analog:

FXS	FXO

Digital Ports

CAS T1/E1	PRI T1/E1	BRI

Call Legs: Voice call over a packet network is segmented into discrete call legs: Associated w/dial-peers

Call leg: A logical connection bet 2 rtr/gateways or bet a rtr/gateway and an IP telephony device

Dial Peers: An addressable call endpoint: Establish logical connections (call legs) to complete end-to-end calls

- Can use dial peers inbound/outbound/both

Dial peers define the properties of the call leg:

Codec	QoS markings	VAD	Fax rate

Cisco voice-enabled rtrs typically use 2 types of dial peers:

POTS dial peers	Connect to traditional telephony network such as FXO/FXS/E&M/BRI/PRI/T1/E1/CAS T1/E1 • Defines the chars of traditional telephony connection • POTS dial peer maps dial str to specific voice port on local rtr/gateway ○ Normally: Voice port connects rtr/gateway to local PSTN/PBX/Phone/fax
VoIP dial peers	Connect over IP network using IP address • Defines attributes of packet voice network connection • Voice-Network dial peers map dial str to remote (IP) device

POTS Dial Peers

<u>**Example:**</u>

R1(config)# dial-peer voice 20 pots

R1(config-dialpeer)# destination-pattern 1234

R1(config-dialpeer)# port 1/0/1

- Dial peer X will be used to match outbound when rtr receives call setup msg for 1234

Destination Pattern Options: **Common destination pattern wildcards:**

Plus (+)	Preceding digit occurs 1/more times
Asterisk (*)/pound sign (#)	*Not* valid wildcards; are DTMF tones
Comma (,)	Inserts a one-second pause
Period (.)	Specifies any one wildcard digit
Square brackets	Indicates a range of digits within the brackets
T	Indicates a variable-length pattern

VoIP: Dial peer mapped to IP/DNS name of destination VoIP device that terminates call

R1(config)# dial-peer voice 50 pots

R1(config-dialpeer)# destination pattern 2010

R1(config-dialpeer)# port 1/1/1

R1(config)# dial-peer voice 40 voip

R1(config-dialpeer)# destination-pattern 1…

R1(config-dialpeer)# session target ipv4: 10.10.10.1

ITSP: Internet Telephony Service Providers

Cost savings: Cost per line less than traditional offerings | Long distance charges lower

- Can purchase lines in increments of 1 instead of larger blocks found in E1s/T1s/PRI
- When not in use for voice: Can use unused BW from connection for other apps
- SIP: Most common protocol used by ITSPs
    - Implement: Create VoIP dial peer w/correct settings for ITSP to which you're connecting

ITSP: Internet Telephony Service Provider: Offers digital telecomm services based on VoIP: Provisioned via

Internet
VSP: Voice Service Provider: ITSPs AKA: VSP: Simply VoIP Providers

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