Intro to Integrated/Unified Communications

September 18, 2023 1:07 PM

What is integrated communications?
- aka Unified Communication Services

Integration of communication and collaboration tools:
- IP Telephony - VOIP
- Video Conferencing
- Voice Mail
- Instant Messaging
- Presence - status, such as online
- Content Sharing
- User Mobility - being able to do it from anywhere (phone, laptop, desktop, etc.)

integrated communications add value to a business by:
- increasing productivity - quicker to use and have everything in one place
- enabling better collaboration
- supporting remote workers
- save money - don't have to drive in or pay for parking

Microsoft:
- live communications server (2003-2005)
- office communicator (2007-2015)
- skype for business (2015-2017)
- Microsoft teams (2017-now)

Cisco:
- cisco IP phone
- cisco IP communicator
- cisco Jabber
- cisco Webex Teams
- cisco Webex Calling

Challenges in implementing or modernizing unified communications services:
- infrastructure
- choosing a provider - where are the data centres? might be laws that need them in Canada for ex
- software and hardware integration
- security policies
- project funding - probably the most important, will it save that much money?
- user training - having to get people used to a new thing

Voice over IP intro:
key component of unified communications services
- sending voice (and video) over an IP-based network using VOIP capable phones (hardware or software)

topologies:
- on prem - at minimum requires:

call server - core component
- call processing - originating, routing and terminating calls

- signaling and device control - call setup/call teardown
- dial plan administration - set of configurable lists to do call routing
- directory services - storing user info; users can authenticate here
- phone feature administration - hold, transfer, forward, conference, speed dial, redial
- sends configs to phone from TFTP server (using MAC and IP)

gateway - device used to connect an internal network to a different system (PSTN) or the rest of the world. (essentially does a translation between two systems.

- VOIP protocols
- codecs
- end points
- PoE
- DHCP and TFTP servers

- hosted provider/cloud - most common now

VOIP Protocols (SCCP / SIP)

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VoIP basic operation

DHCP server

- assigns network config to a phone

TFTP server

- phones contact TFTP server (usually running on call server) for configuration download

Call server

- phones then register with the call server, which makes the server aware of the phone and the phone's MAC and IP address tied to that phone number

Call setup and connection

- once phones are registered, call sessions can be initiated
- the calling end point signals the call server indicating the number dialed - phone 1 dials phone 2 and the call server sees this
- call server contacts destination and routes call to it if available
- RTP protocol is used to transmit voice data

Call termination

- signalling protocols come back into the connection and provide messages to the calling parties regarding tearing down the session

VOIP Protocols

- Signaling Protocols: **SIP, SCCP, H.323**
- Transport Protocols: **RTP**

SCCP - Skinny Client Control Protocol (CISCO PROPRIETARY)

- lightweight cisco proprietary signaling protocol for VOIP devices
- used for phones' registration, call messages, and controlling phone display
- TCP port 2000
- messages are easy to read on Wireshark

basic header format:

- data length - tells you amount of info carried in the packet
- header version - tells you the version of the skinny header
- message id - provides hex value for type of SCCP packet

registration process:

- post boot and once it has network config and TFTP, it communicates with call manager via UNICAST
- **Message ID: RegisterRq**
- **DeviceName**: includes mac address of phone
- call server responds with
- **RegisterACKMessage**
- and a **KeepAliveInterval value**
- they then exchange info regarding interface setup and features
- do this through **Capability** and **Template** messages

calling process:

- picking up the handset is called "going off-hook"
- phone informs server it is off-hook
- server sends a call-state message acknowledging the off-hook state
- server sends **StartTone** message

- dialing a number
- generates dial tone, destination number can be dialed
- server routes the call to destination and sends a message to the source that is time to ring

tearing down the call:

- begins with **OnHook** message
- followed by **CloseRecieveChannel**
- then **StopMediaTransmission** messages

RTP - Real-Time Transport Protocol

- used for voice packets containing voice data
- also has Real-Time Control Protocol (RTCP)

SIP - Session Initiation Protocol

- generic standard from IETF
- operates at **application layer**

SIP Registration:
- boot up and contact **DHCP** for networking
- contact **TFTP** for profile configs
- register with **call server/registrar**

SIP Components:
- **user agent client (UAC) -** the SIP phone, initiates requests or responds to SIP transactions
- **User Agent Server (UAS)/Registrar -** accepts requests and sends back responses - accepts REGISTER messages and updates location
- **URI (uniform resource identifier) -** part of SIP addressing - IP or username

sip:<user>@domain:port
sip:<user>@host:port
sip:<phone number>@domain
sip:<phone number> @ host

SIP is designed to setup and teardown calls
- also includes user availability, session handling info, but it relies on another protocol called **SDP (session descriptor protocol)** for negotiating parameters of the multimedia connection

SIP messages and message structure
start lines:
- SIP URI (uniform resource identifier)
- VERSION
- METHOD (REGISTER, ACK)

message headers:
- contain mandatory fields and rules

Requests:
- **VIA :** tells nodes involved where to send SIP packet, rules
- **FROM :** identity of the initiator in URI format
- **TO :** specifies the recipient of the request in URI format
- **CALL-ID :** value that groups all of the messages from a dialog together
- **CSEQ :** helps you find out what number of packet that is in a transaction

Responses: ??? are these request or responses
- **1xx Provisional :** request received, processing
- **2xx Success :** request successfully received, understood and accepted
- **3xx Redirection :** further action needs to be taken for completing the request - forwarding a call
- **4xx Client Error :** request contains bad syntax, or cannot be fulfilled by server - probably a mistyped number
- **5xx Server Error :** the server failed to fulfill a valid request - server might be down
- **6xx Global Failure :** the request cannot be fulfilled at any server - system is down

SDP - Session Description Protocol - provides purpose of the session and describes the media content to be transferred
includes:
- Version
- Originator (owner)
- Session ID
- Session Name
- Connection Information
- Media Type
- Media Port

SIP Trunks
- replace the need for physical lines and allow for easier management of phone services
- backbone of users phone lines connected to a telephony network
- allows connecting a VoIP PBX system to a PSTN network or connecting two different VoIP systems

signaling protocols: SIP, SCCP
transport protocols: RTP

VOIP QOS

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What impacts VoIP quality

Latency: audio latency is impacted by the time it takes to encode the audio and the time for the packet to travel to the destination

- according to cisco, 150ms is normal for VoIP, 300ms and over is unacceptable
- can be impacted by:

hardware capabilities (routers transmitting specific data rates)
software and configuration (firewalls, NAT, QoS misconfiguration)
network congestion (oversubscription)
location (distance)

- measured with ping tests, audio test tools
- could lead to long pauses and overlapping noises during calls

Jitter: variation in latency between packets

- increased by network hardware, congestion, wireless networks
- high jitter can result in choppy voice, glitches
- max recommended jitter one way = 30ms; more is distorted voice quality
- ideally it should be below 20ms

Packet Loss: packets not arriving to destination

- can occur anywhere in the network
- Voice data is based on UDP - no mechanism to deal with error checking, acknowledgements of segments, recovery
- **packet loss should be below 1%**
- impact: not hearing chunks of conversation
- can be caused by:

resource limitations (network congestion leading to full buffers, low memory)
misconfigurations (duplex mismatch, routing loops, QoS misconfig)
security configs (ACLs)
software bugs

Codecs, Voice Compression: determines how many voice samples are sent per frame

- bigger the value, the higher the bandwidth utilization (more voice samples are packed int the payload field of UDP/RTP packet)
- **network header overhead would be lower** (impact of a packet loss on voice quality would be bigger)

cRTP - compressed RTP - can be used to reduce the bandwidth consumed by VoIP traffic

Quality of Service:

- used to achieve the bandwidth and latency needed for a specific application
- look at network as a whole when trying to do QoS - allows you to identify how to best achieve the best QoS
- **upgrading the link -** best solution but most expensive, still could be congestion
- QoS cannot solve codec delay / sampling delay.

Quality of Service tools and functions:

- packet classification and marking
- congestion avoidance (queuing techniques)
- traffic conditioning: policing and shaping

Packet Classification and Marking - classifying data into class-maps (traffic groups) and mark it (stamp it in the packet header) based on priority

- can be done at Layer 2 or at Layer 3 of the stack

layer 2 marking - COS (class of service) - priority field in layer 2

- frame contains **3 priority bits** used to mark traffic as highest or lower priority (lowest number lowest priority) - for most it is 0-5 with 3 saved for networking
- can be set at device itself, however, mac address changes with each router, the COS (class of service) value resets to 0 - lowest priority

layer 3 marking - DS (differentiated services) - DSCP

- IPv4 and IPv6 packet header has an **8-bit** field size for ToS(IPV4) or Traffic Class(IPv6) used in marking traffic for QoS
- two methods of layer 3 marking

IP Precedence - obsolete, not very scalable
DSCP - Differentiated Services Field Codepoints

IP Precedence - 3 bits > 8 possible classes of traffic (0-7)

- dont use last 2 bits tho because they are used for routing updates and hello packets
- 5 is highest
- allows routers to group and queue traffic flow based on the precedence bits settings
- bits can be set in a few ways - destination phone numbers, ACLs with subnets, port numbers

DSCP - differentiated services field codepoints

- uses the 6 left most bits in the TOS field of the IP header - this allows for 64 (2^6) classes for marking (0-63)
- IETF selected **21** of these values and named them to avoid issues when traffic travels between separate networks
- these names are called **per-hop behaviours (PHBs)**

Default - value 0
Expedited Forwarding (EF) - value 46 - cisco recommends this for VoIP

★ slide 15 - 19 is missing here idk what was going on

Campus QOS service considerations:
- high bandwidth campuses - worry is focused on managing packet loss
- HD video for example - very sensitive to packet drops
- perform QOS in hardware rather than software where possible
- best practice is to classify and mark applications as close to their sources
    - with IP phones, and shared ports (data and VoIP) you can have them on the phones themselves (data vs voice), then the switch trusts the classification of the device
    - or you can have it on the L2 switch to do the classification
- VLAN based policy

how to do this on cisco stuff
- match-all has to match **everything**
- match-any can match anything

```
class-map match-all voice
match protocol rtp
class-map match-all eigrp
match protocol eigrp
class-map match-all dns
match protocol dns
class-map match-any sccp-ssh
match protocol ssh
match protocol skinny
!
policy-map marking
class voice
set ip dscp ef
class eigrp
set ip dscp cs7
class dns
set ip dscp cs1
class sccp-ssh
set ip dscp af31

int f0/1
ip address 10.10.10.2 255.255.255.252
service-policy output marking
```

Traffic policing and Shaping

Policing:
- limits bandwidth by **discarding** traffic
- can re-mark excess traffic
- should be used on higher-speed interfaces
- can be applied inbound or outbound
- traffic that conforms is transmitted
- **traffic that exceeds is either forwarded with a decreased priority or is dropped**

Shaping:
- limits excess traffic by buffering
- buffering can lead to a delay
- recommended for slower-speed interfaces
- cannot re-mark traffic
- can only be applied in the outbound direction