# NETW2100 - Integrated Communications Study Guide

# Module 1 - Intro

## Understand the components of Unified communications:

### major components:

- IP telephony - VoIP
  - VoIP - sending voice / video over IP based network using VoIP capable phones (hardware or software)
- Conferencing / Video Conferencing
- Voice Mail
- Instant Messaging
- Presence - online, away, offline
- Content Sharing
- Email
- User Mobility - not being locked to one device

### Why?

- increases productivity - quicker to use and have everything in one place
- enables better collaboration
- supports remote workers
- save money - don't have to pay for parking, or drive in

### Topologies:

- on prem
  - minimum requirements:
    - call server - important core component
    - gateway - translates between internal network to external (PSTN)
    - VOIP Protocols
    - Codecs
    - End points
    - PoE

    - DHCP and TFTP servers
- hosted provider / cloud

# Understand the role of the Call server in VoIP systems

## Call Server

The call server is responsible for the following:

| Service | Use |
| --- | --- |
| 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 (dial 0 before # to get to outside) |
| Directory Services | storing user info; users authenticate here |
| Phone feature administration | hold, transfer forward, conference, speed dial, redial |
| phone setup | sends configs from TFTP server - which is usually on call server - (using MAC and IP) |

# Understand the role of signaling protocols and transport protocols in VoIP systems

## SIGNALING PROTOCOLS:

- SIP - Session Initiation Protocol
- SCCP - Skinny Client Control Protocol (CISCO Proprietary)
- H.323 - same idea as SIP and SCCP - superseded by SIP

## TRANSPORT PROTOCOLS:

- RTP - Real-Time Transport Protocol

## SIP

Application layer - L7

### COMPONENTS:

- UAC (User Agent Client) - SIP phone, initiates requests or responds to SIP transactions

- UAS (User Agent Server) / Registrar - accepts requests and sends back responses - accepts `REGISTER` messages and updates location
- URI (Uniform Resource Identifier) - part of SIP address - IP or username
  - sip:`user`@`domain`:`port`
  - sip:`user`@`host`:`port`
  - sip:`phone #`@`domain`
  - sip:`phone #`@`host`
  Relies on a protocol called
- Session Descriptor Protocol (SDP)

### SIP Messages and message structure:

**Start Lines:** (requests: `Request-Line` | responses: `Status-Line` & `Status-Code`)

- SIP URI
- Version
- Method (`REGISTER`, `ACK`, `INVITE`)

**Message Headers:** (requests)

- VIA - where to send SIP packet
- FROM - identity of initiator - URI form
- TO - specifies the recipient of request - URI form
- Call-ID - value that groups messages from a dialog together
- CSeq - help identify transactions and order them

**Message Headers:** (responses)

- 1xx : Provisional - request received, processing
- 2xx : Success - request successful received, understood, and accepted
- 3xx : Redirection - further action needs to be taken
- 4xx : Client Error - bad syntax or can't be fulfilled by server - probably wrong number
- 5xx : Server Error - server might be down, server failed to fulfil request
- 6xx : Global Failure - system is down, no server working

# Module 2 - VoIP protocols

## Understand the process of phones registration, calling and call termination

### Registration

| Protocol | Process |
| --- | --- |
| SIP | 1. Phones boot up and contact DHCP for network config<br>2. Phones contact TFTP for profile config<br>3. Phones register with call server / registrar |
| SCCP | 1. Phone boots up, DHCP for networking and option 150 - CME for TFTP config<br>2. Communicates with Call Manger through unicast messages<br>(`MESSAGE ID: RegisterRQ`) & (`DeviceName: <mac address>`)<br>3. The server responds with `RegisterACKMessage` & `KeepAliveInterval`<br>4. After registering - phone and server exchange info regarding interface setup and supported features<br>(`Capability` & `Template` messages) |

### Calling

| Protocol | Process |
| --- | --- |
| SIP | 1. Phone calls out using a SIP URI (sip:`phone-num`@`host`), request `ACK` by server<br>2. Server routes the call to the recipient, which now rings<br>3. Packets go to the call server that acts as the bridge from caller to callee through UDP/RTP |
| SCCP | 1. Phone is picked up - `OffHook` message is sent to server, server acknowledges this<br>2. Server sends `StartTone` message to phone - indicating time to dial<br>3. `DialedNumber` is the number the phone dials, individual buttons are `KeypadButton`<br>4. Server then routes the call and sends a message to phone that it is time to ring<br>5. Channel between end points is set up for voice packets over UDP/RTP |

### Call Termination

| Protocol | Process |
| --- | --- |
| SIP | 1. Someone hangs up and initiates a `BYE` request<br>2. Other person sends `STATUS 200 OK (BYE)` packet which ends the call |
| SCCP | 1. Someone hangs up the phone, `OnHook` message is sent<br>2. `CloseRecieveChannel` message is sent<br>3. `StopMediaTransmission` message is sent |

# Module 3 - QoS and Traffic Shaping (includes the Research Project)

### understand what factors and how they impact VoIP quality

| Name | Effect |
| --- | --- |
| Latency | The amount of time it takes to encode the audio + the time it takes to reach its destination.<br>Can lead to long pauses and overlapping noises during calls, speakers interrupting each other.<br>Impacted by: low router transmitting speeds, firewall, NAT, QoS misconfigs, network oversubscription, distance.<br>150ms one way is normal for VoIP, **over 300 is unacceptable**. |
| Jitter | The variation in delay of received packets / variation of latency between each data packet.<br>Can cause choppy voices, glitches.<br>Impacted by: old network hardware, congestion, wireless networks.<br>**30ms is MAX** one way recommendation for jitter, ideally 20ms or below. |
| Packet Loss | Not all packets arriving at destination, or discarded.<br>Can cause chunks of conversation to be skipped or not heard.<br>Impacted by: low memory, duplex mismatch, routing loops, ACLs, software bugs<br>Packet loss should be **below 1%**. |
| Codecs, Voice Compression, cRTP | The Codec determines how many voice samples are sent per frame, the bigger the value the more bandwidth utilization.<br>Impact of packet loss on voice quality would be bigger.<br>cRTP - compressed RTP can be used to reduce bandwidth consumed by VoIP traffic |

## understand what is the purpose of QoS

The point of QoS is to prioritize certain applications over others so they can always run and not have to fight network congestion like all other applications.
QoS uses queuing techniques to avoid congestion - packet classification and marking.
**Classification** - grouping packets into class-maps (traffic groups).
**Marking** - stamping the packet in the packet header based on priority.

## what are the differences between COS and DSCP

**COS** - Class of Service - priority field in **layer 2** (Data Link layer / MAC layer)

- 3 bits as priority bits (0-7)
- can be set on the device itself, however MAC address changes with each router, so COS value resets to 0
- can be used for internal, not so great if it has to pass router

**DSCP** - Differentiated Services Field Codepoints - **Layer 3 Marking**

- uses 6 left most bits in TOS field of IP header - this allows for 64 classes for marking (0-63)
- IETF selected **21** of these values and named them - called PHBs (Per-Hop Behaviors)
  - Default - value of 0 - 000000
  - Expedited Forwarding (EF) - value of 46 - 101110 - cisco recommends this for VoIP
  - Assure Forwarding (AFxy) - 12 classes

## understand how Assured Forwarding is used by the routers when making decisions on what type of traffic is prioritized and dropped if congestion occurs

**AFxy** - is the general format for the Assured Forwarding packets. A higher X is prioritized, then if they are the same following that, the lower Y is prioritized. So, AF41>AF21, AF31>AF33
**EF** - is the highest value and beats out all AF
untagged traffic is **DF** and is beaten by any tagged traffic.

| Application Class | Per-Hop Behavior | Admission Control | Queuing & Dropping | Application Examples |
|---|---|---|---|---|
| VoIP Telephony | EF | Required | Priority Queue (PQ) | Cisco IP Phones (G.711, G.729) |
| Broadcast Video | CS5 | Required | (Optional) PQ | Cisco IP Video Surveillance / Cisco Enterprise TV |
| Realtime Interactive | CS4 | Required | (Optional) PQ | Cisco TelePresence |
| Multimedia Conferencing | AF4 | Required | BW Queue + DSCP WRED | Cisco Unified Personal Communicator, WebEx |
| Multimedia Streaming | AF3 | Recommended | BW Queue + DSCP WRED | Cisco Digital Media System (VoDs) |
| Network Control | CS6 | | BW Queue | EIGRP, OSPF, BGP, HSRP, IKE |
| Call-Signaling | CS3 | | BW Queue | SCCP, SIP, H.323 |
| Ops / Admin / Mgmt (OAM) | CS2 | | BW Queue | SNMP, SSH, Syslog |
| Transactional Data | AF2 | | BW Queue + DSCP WRED | ERP Apps, CRM Apps, Database Apps |
| Bulk Data | AF1 | | BW Queue + DSCP WRED | E-mail, FTP, Backup Apps, Content Distribution |
| Best Effort | DF | | Default Queue + RED | Default Class |
| Scavenger | CS1 | | Min BW Queue (Deferential) | YouTube, iTunes, BitTorent, Xbox Live |

This diagram is top down in order of importance.

## (look at AF classes and understand what do the numbers mean - AF11, AF31, AF12....)

- Assured Forwarding (AFxy) - 12 classes
  - x - decimal value of first 3 bits (higher number = better queuing)
  - y - decimal value of next 2 bits (higher number = higher drop probability)

## understand what class should VoIP packets should be placed in

Cisco claims VoIP packets should be put in the Expedited Forwarding (EF) category.