

Chapter 1

- Signaling protocols generate control traffic/information between 2 endpoints
 - Peer to Peer
 - Client-Server
- Peer to Peer Signaling Protocols
 - H.323 – Umbrella of standards for voice, video, and data transmission
 - Includes the following protocols
 - H.225 Call signaling – establish a connection between 2 H.323 endpoints or an endpoint and a gateway
 - H.225 Registration, Admission, and Status – provides registration, admission, status, bandwidth changes, and disengage procedural messages between endpoints and gatekeepers
 - H.245 control signaling – provides end to end control messages about capabilities, opening/closing logical channels, flow control, general commands
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 - SIP – Similar to http with URLs and status codes
 - Works with SAP (session announcement protocol) and SDP (session description protocol)
 - H.323 alternative
 - UAs (user agents) initiate sessions
 - SIP devices do not register with a call manager
- Client-Server Signaling Protocols
 - MGCP – protocol for PSTN gateway control
 - Controls VOIP gateways that are connected to call agents
 - Provides signaling for edge devices that do not have a full voice signaling protocol
 - All dial plan info resides on a separate call agent (not the CM)
 - Plain text protocol
 - Call manager knows of and controls individual voice ports on an MGCP gateway
 - SCCP – TCP skinny client control protocol between call manager and client
 - Cisco proprietary
 - Used between call managers and endpoints
 - All events cause traffic to the CM (the CM will tell the endpoint what to do in response to that event)
- Media Transmission Protocols
 - RTP – real time protocol
 - Communicates directly between endpoints
 - Uses multiplexing and checksum offered by UDP

- Communicates on an even port, vs RTCP communicating on an odd port
- RTCP – real time control protocol
 - Provides control info for RTP
 - Primary function is to provide feedback on the quality of service provided by RTP
 - Monitors traffic sent, jitter, feedback, delay and can signal the device to use higher compression codec if needed
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- cRTP – compressed RTP
 - RTP header is generally larger than the data it carries (40 byte header! – 20 IP bytes, 8 UDP, 12 RTP)
 - Header compression results in 2-4 byte headers (the endpoints actually cache the header info and don't necessarily compress it)
 - Useful on WAN links
- sRTP – Secure RTP
 - Provides encryption, message authentication, integrity, and replay protection
 - sRTCP also
 - Uses AES encryption only
 - Uses HMAC-SHA1 for integrity and authentication – calculated for entire IP packet
- VOIP Gateways – connects VOIP systems to PSTN, PBX, or other TDM systems
 - Analog gateways provide connectivity from IPT to POTS or connect fax machines, etc. to VOIP systems
 - Analog trunks connect IPT to PSTN or PBX using FXO ports and E&M ports to connect to legacy PBX's
 - Digital gateways connect IPT to PBX or PSTN using digital (PRI/BRI) trunks