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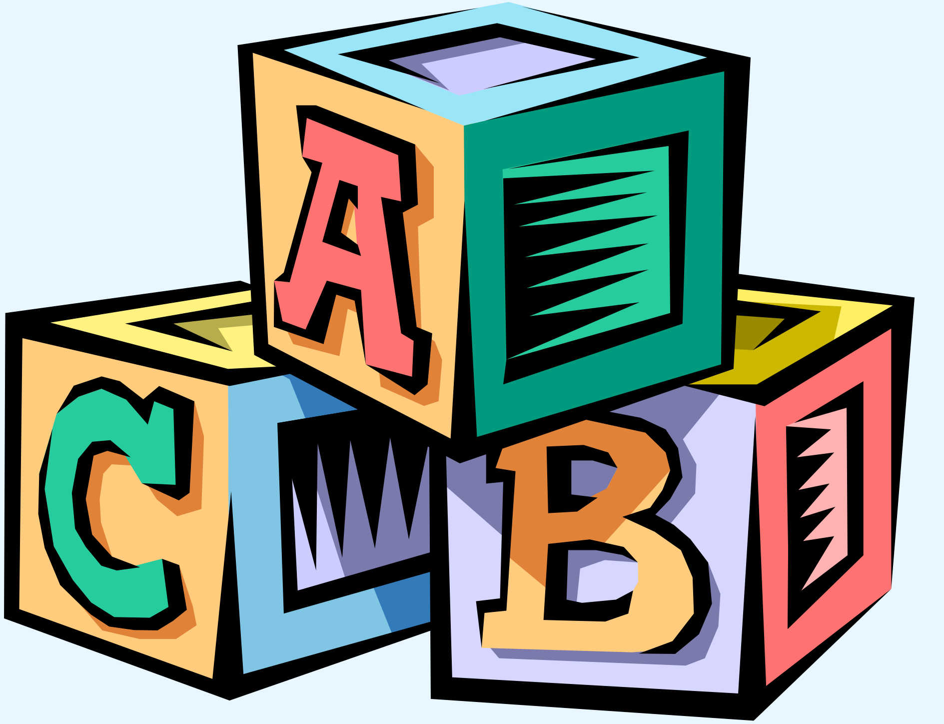
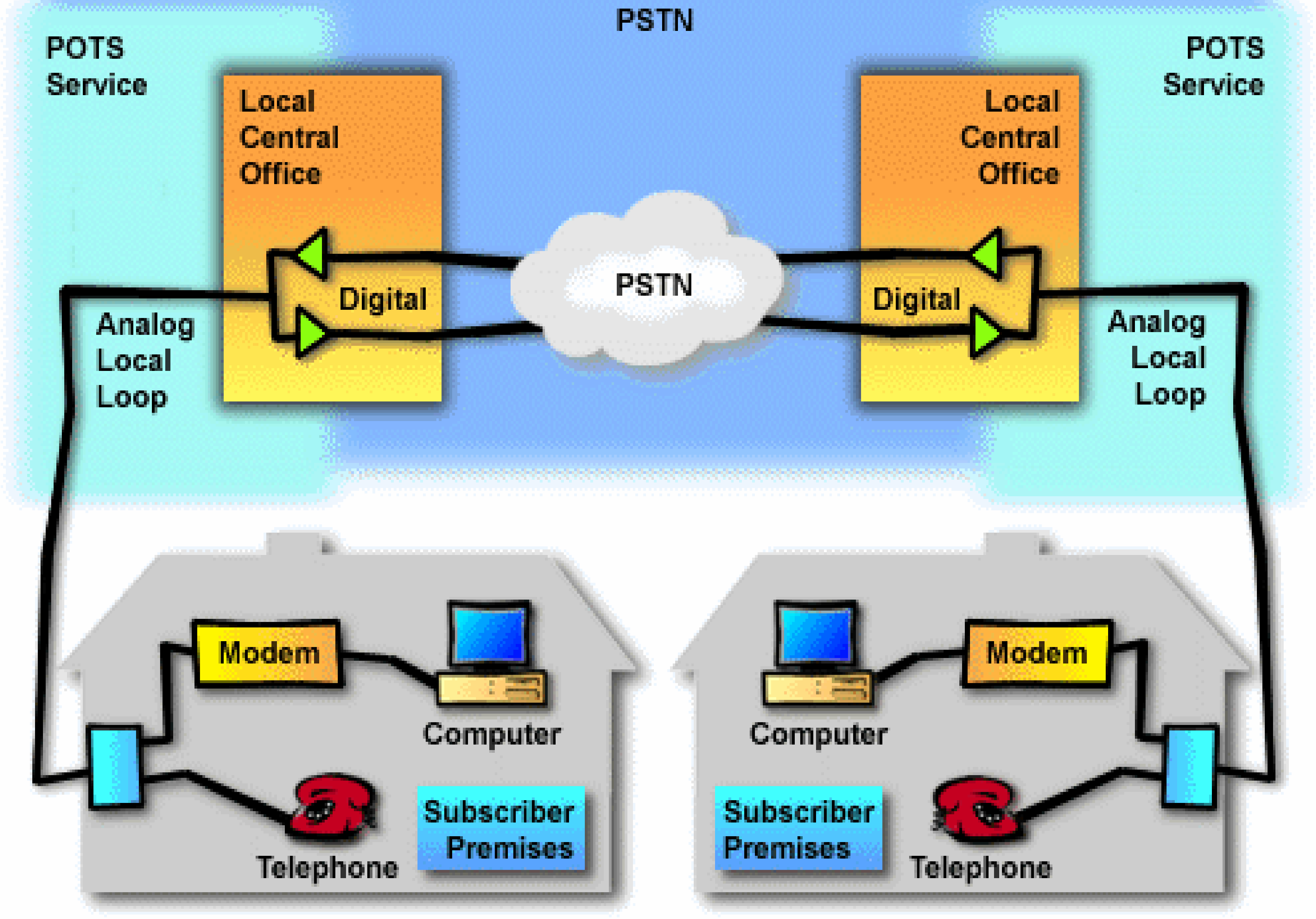
Your Direct Connection

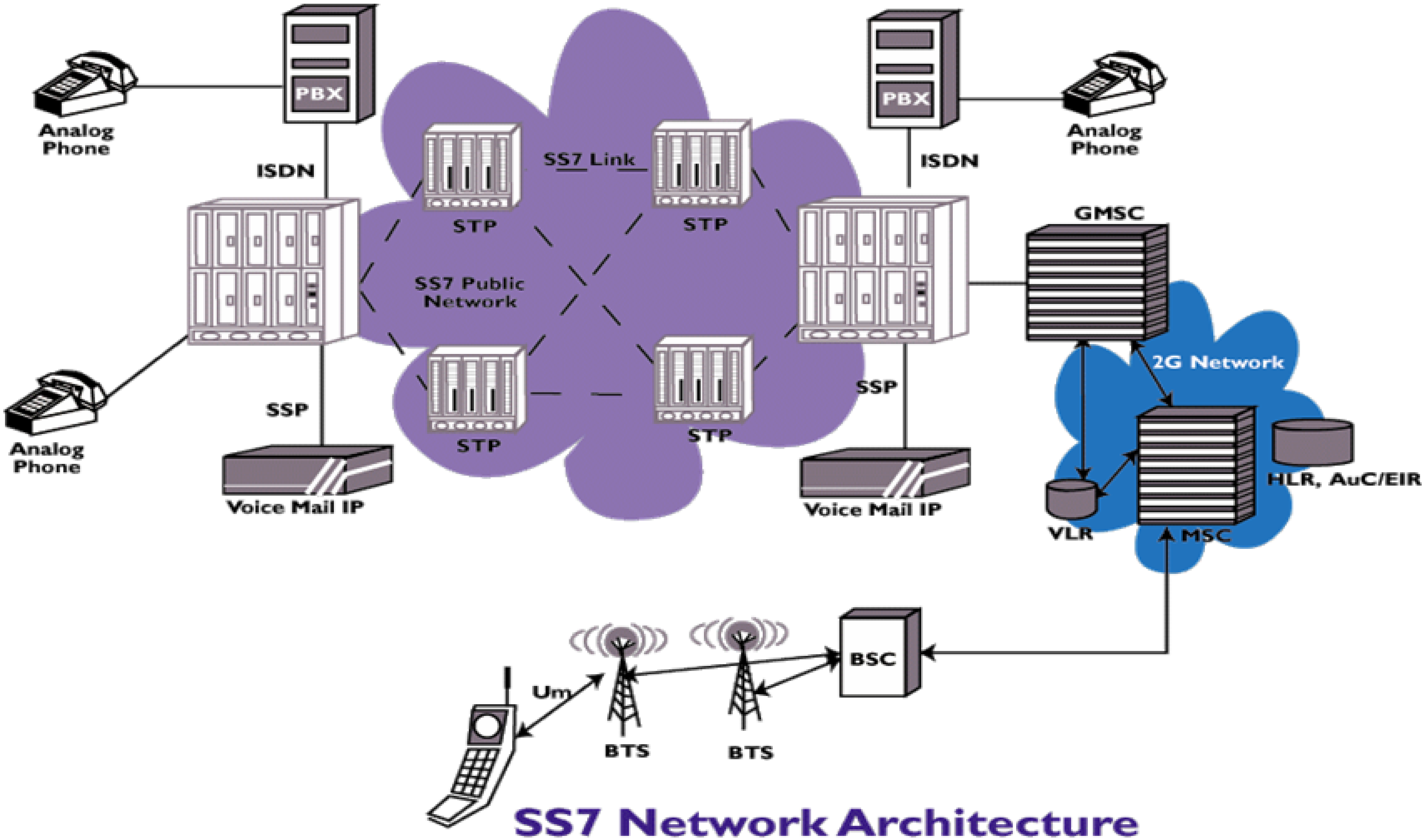
VoIP Basics



Phone Network



Typical SS7 Network Architecture

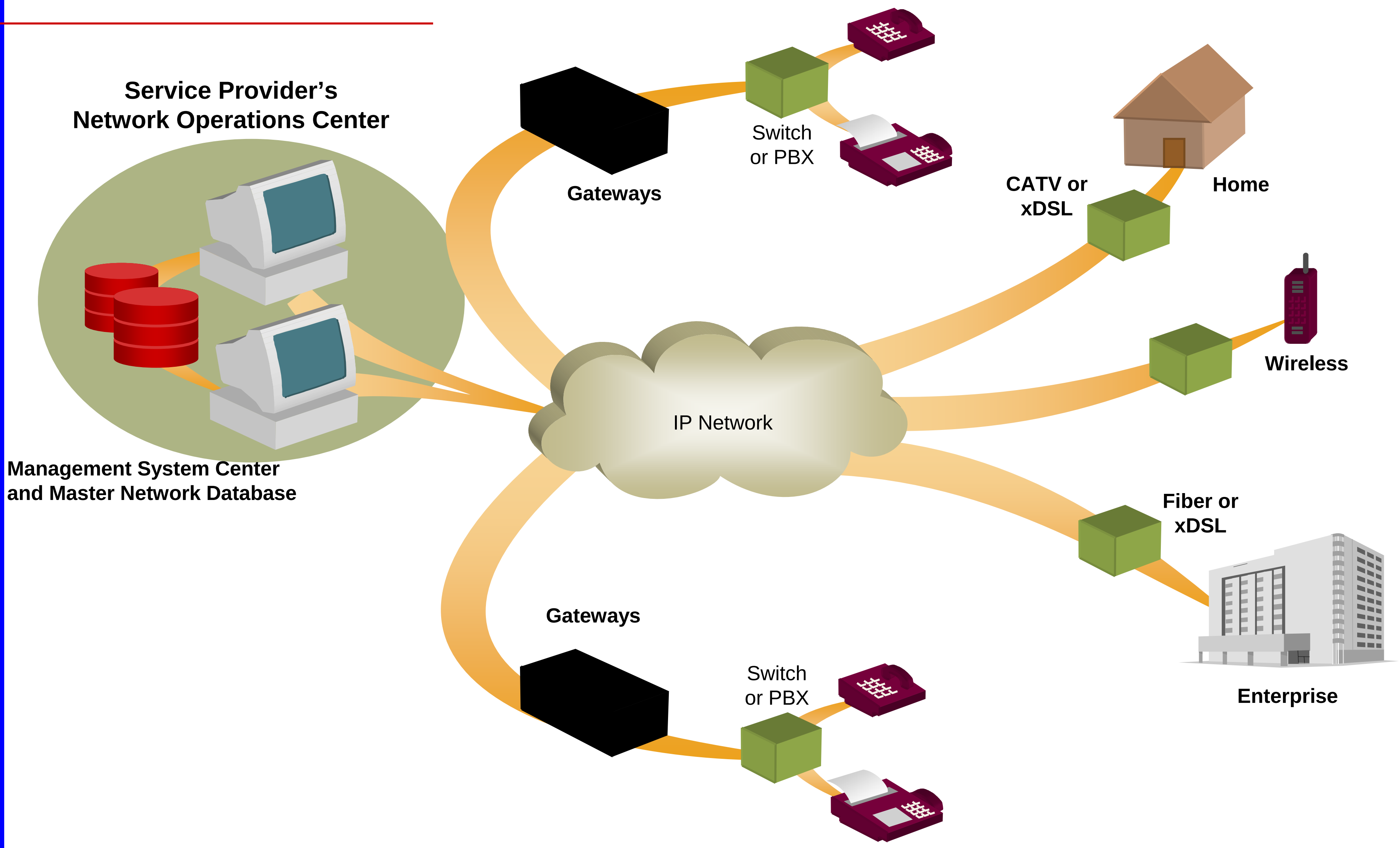


What is VoIP ?

(or IP Telephony)

Voice over IP (VoIP) is the transmission of digitized telephone calls over a packet switched data network (like the Internet) using the Internet Protocol rather than the traditional circuit switched voice Networks as a physical media.

VoIP Network



VoIP Calls

Normal telephone calls – have two parts :-

- Call control (setting up/tearing-down the call)
  - VoIP - call “signaling” uses one of many protocols
- Voice (speech) Communications
  - VoIP - uses a standard voice IP “streaming” protocol

Some VoIP Signaling Protocols

Conversion of a TDM (traditional) telephone signaling system (like ISDN or SS7) to a VoIP system :-

- ITU-T H.323
- IETF SIP RFC 3261
- IETF Megaco RFC 3525/ITU-T H.248
- IETF MGCP RFC 3435
- IETF SigTrans RFC 2719
- **Some use TCP as transport protocol**

Voice Streams

Converting the speech stream from a TDM (traditional) telephone system to a VoIP compatible system :-

- IETF's RTP (Real Time Protocol) for voice stream
- IETF's RTCP (Real Time Control Protocol) for voice stream control
- **Uses UDP as transport protocol**

Streaming Protocol

Packet make up is :-

Media Header, i.e. Ethernet
Internet Protocol (IP) Header
User Datagram Protocol (UDP) Header
Real Time Protocol (RTP) Header
Voice Sample(s)

➢60 bytes (Approx.) overhead

i.e. :-
<Media header – i.e. Ethernet><IP Header><UDP Header><RTP Header><Voice Sample(s)>

Nyquist

Nyquist devised a sampling system that is used when turning analogue voice into digital voice :-

- Analogue voice frequencies are between 400 Hz and 4.4 KHz … i.e. a bandwidth of 4 KHz approx.
- Nyquist defined a sample rate of 2 * maximum frequency to be sampled, i.e 2 * 4 KHz = 8 KHz
- Each digital sample of the analogue voice signal is 8 bits
- SO … the bandwidth needed for an analogue voice stream when converted to digital is 8 bits * 8 KHz = 64 Kbit per voice channel (i.e. per call)

Voice Compression

ITU-T defined digital voice compression/decompression (codec) systems :-

- G.711 (64 Kbit bandwidth) A-law (world) or μ-law (JPN/USA) uncompressed Pulse Coded Modulation (PCM)
- G.723.1 (5.3 Kbit or 6.4 Kbit bandwidth) Algerbraic-Code-Excited Linear-Prediction (ACELP) voice compression
- G.729A/B (8 Kbit bandwidth) Conjugate-Structure-Algerbraic-Code-Excited-Linear-Prediction (CS-ACELP) voice compression
  - G.729B includes speech activity detection for silence suppression
- Higher bit rates available from G.726 and G.728, etc. which may suit other languages than any of the Latin based are also available.

Sample packing

- UDP allows for the negotiation of packet sizes
- More than one voice sample in an IP packet :-
  - Increases efficiency
  - Lowers bandwidth requirements
- BUT … increases delays as there is time to collect more than one sample for each IP packet.

Jitter

What is “network” jitter …

- Transmitted VoIP packets may not appear at an appropriate frequency, sequence or gapped due to variable network delays (latency)

SO …

- buffer received VoIP to deliver frequently
- replace “missing” or “out of sequence” packets with duplicate of last packet

BUT …

- Look out for increased delay

Echo Cancellation

ITU-T G.165 (analogue) & G.168 (digital) define standards for echo cancellation

- The sending end buffers the sent speech and compares it with incoming speech.
- If they are the same, it's an echo, so remove it.
- These definitions usually include a sending buffer time/length called the "tail".

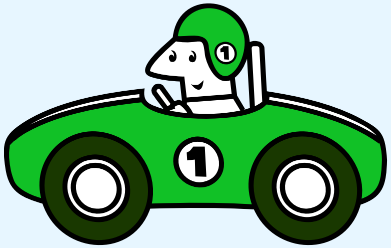
Call Bandwidth

Remember, IP bandwidth is cheaper than TDM bandwidth !!!

- Using G.729A at 8 Kbit without silence suppression and packing two voice sample per IP packet can achieve bandwidths of 25 Kbit per call
- Using G.729B at 8 Kbit packing 4 voice samples per IP packet can achieve bandwidths of 8 -12 Kbit per call
- Using G.723.1 at 6.4 Kbit without silence suppression and packing one voice sample per IP packet can achieve bandwidths of 12 - 16 Kbit per call
- Using G.723.1 at 6.4 Kbit with silence suppression and packing 4 voice samples per IP packet can achieve bandwidths of 4 - 6 Kbit per call

Your Direct Connection

VoIP - Questions



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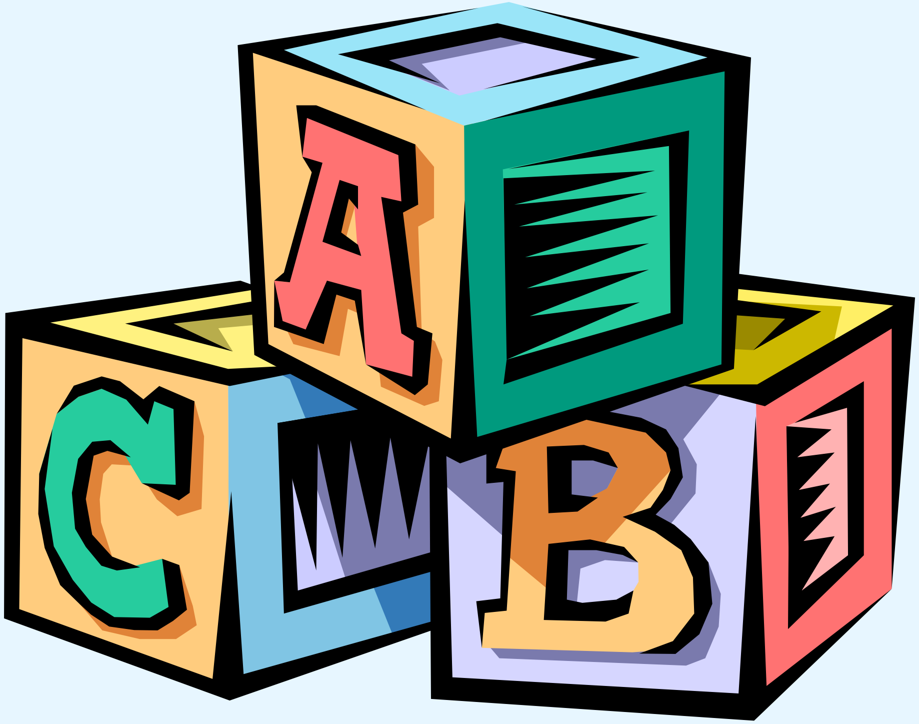
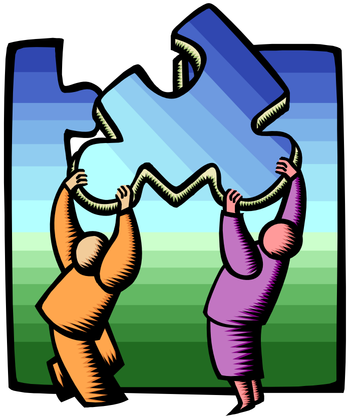
H.323 Basics



Elements of an H.323 System

- Terminals
- Multipoint Control Units (MCUs)
- Gateways
- Gatekeeper
- Border Elements

Referred to as “endpoints”

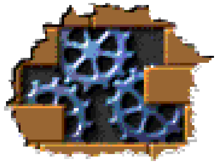


H.323 Terminals

Terminals are a mandatory component of H.323 and may comprise of :-

- Telephones
- Video phones
- IVR devices
- Voicemail Systems
- “Soft phones”

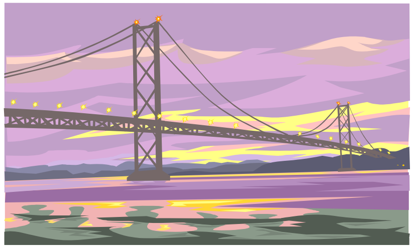
H.323 MCUs



Multipoint Controller Units (MCU’s) are :-

- Responsible for managing multipoint conferences (two or more endpoints engaged in a conference)
- Formed by a Multipoint Controller (MC) that manages the call signaling and may optionally have Multipoint Processors (MPs) to handle media mixing, switching, or other media processing

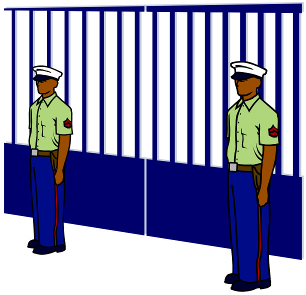
H.323 Gateways



Gateways are optional H.323 components and can be described as :

- A Gateway is composed of a “Media Gateway Controller” (MGC) and a “Media Gateway” (MG), which may co-exist or exist separately
  - The MGC handles call signaling and other non-media-related functions
  - The MG handles the media
- Gateways interface H.323 to other networks, including the PSTN

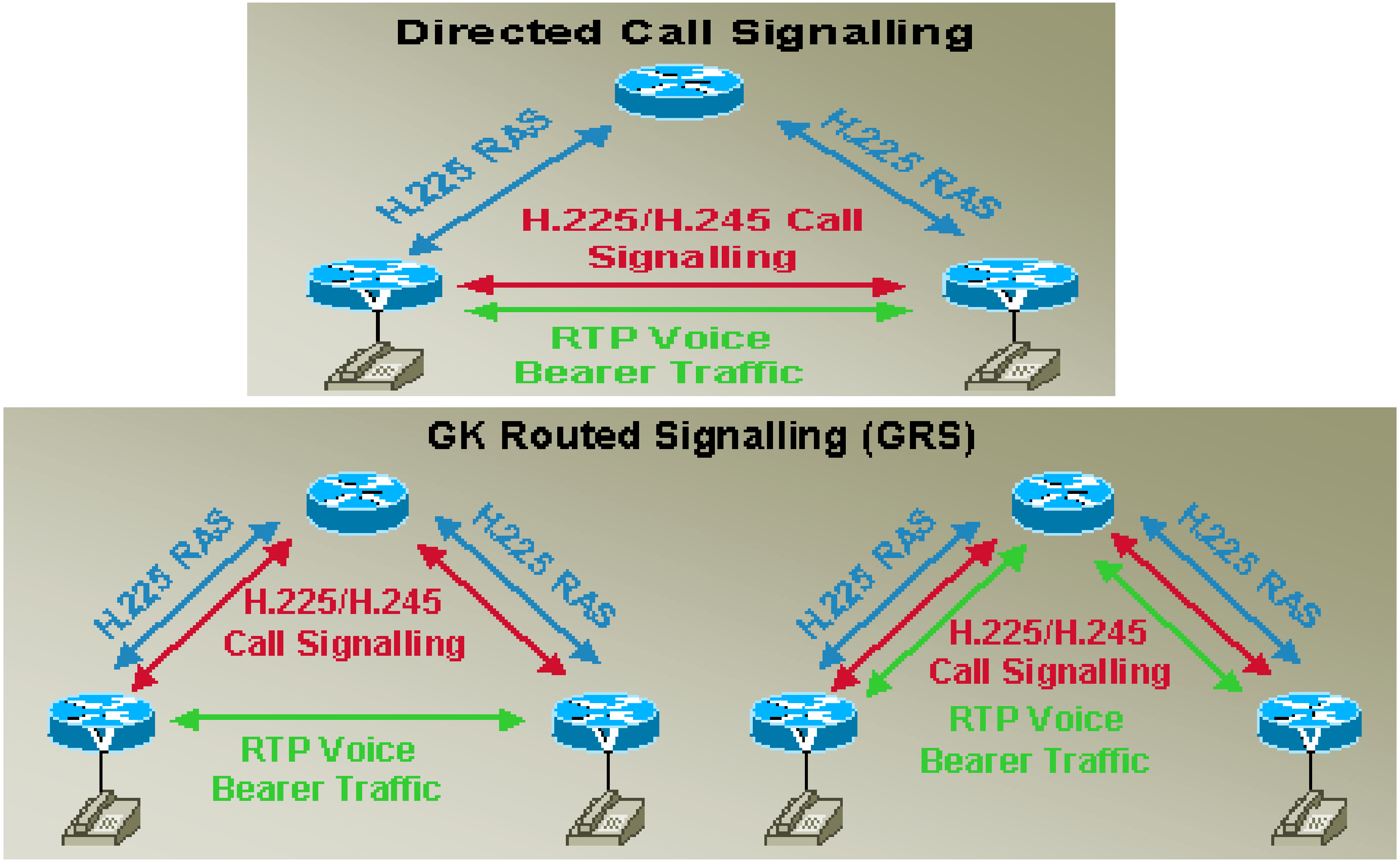
H.323 Gatekeeper



GateKeepers are optional H.323 components and may be described as :-

- Being used for admission control, address resolution and bandwidth management
- Allowing calls to be placed directly between endpoints (Direct Endpoint Model) or routing the call signaling through itself (Gatekeeper Routed Model).

H.323 GK Signaling



H.323 Border Elements

- Border Elements, which are often co-located with a Gatekeeper, exchange addressing information and participate in call authorization between administrative domains
- Border Elements may aggregate address information to reduce the volume of routing information passed through the network
- Border elements may assist in call authorization/authentication directly between two administrative domains or via a clearinghouse

H.323 Protocols

- ITU-T H.323 is a “umbrella” specification describing how the various pieces fit together
- Timeline is = ITU-T H.323 v.1 in 1996, v.2 in 1998, v.3 in 1999, v.4 in 2000
- H.225.0 defines the Q.931 based call signaling and communication between endpoints (Call Signaling) and the Gatekeeper (RAS) functions
- H.245 is the conference control protocol which handles post-connect signaling (e.g. T.38 Fax & out-of-band DTMF) and defines media stream features for connectivity

H.323 Protocols (cont)

- IETF RFC defined RTP/RTCP is used to manage the audio streaming and control
- H.450.x is a series of supplementary service protocols (QSig/ISDN, IP Centrex, etc)
- T.120 defines multimedia conferencing
- H.235 defines security and encryption systems

H.225/H.245 & G.711 support is mandatory in H.323 although G.723.1 (5.6 K & 6.4 K) support is "preferred" when bandwidth is of concern.

FastStart (Fast Connect)

- Fast Connect is a procedure introduced to speed up connections by proposing channels in the Setup, rather than going through the H.245 procedures
- For most point to point calls that use Fast Connect, H.245 is only necessary for DTMF relay or T.38 FAX
- How does a slow-start device negotiate codec's with a Fast-start device ?

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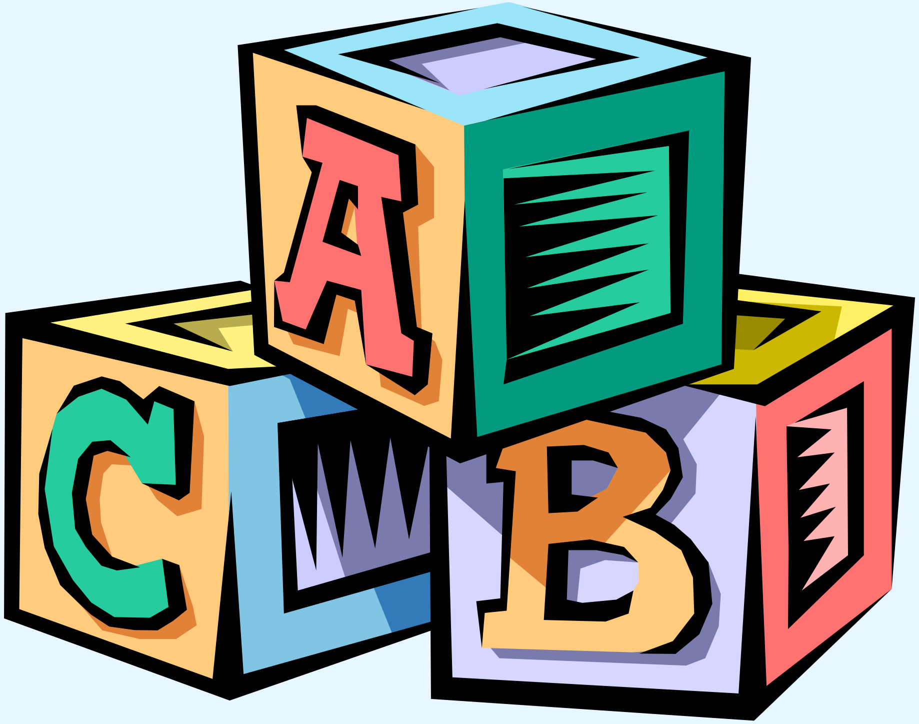
H.323 - Questions

SIP Basics



Session Initiation Protocol (SIP)

- The Internet Engineering Task Force's (IETF's) standard for multimedia conferencing over IP.
- The MMUSIC* Working Group

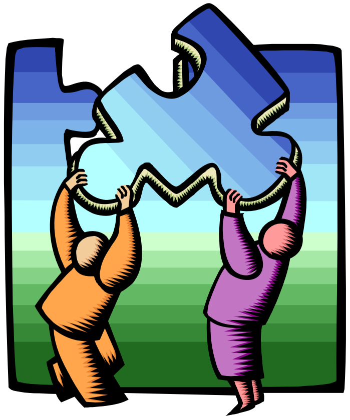
  * Multiparty Multimedia Session Control
- ASCII-based, application-layer control protocol
- Used to establish, maintain, and terminate calls between two or more end points.

SIP Capabilities

- ***Determines location*** - Supports address resolution, name mapping, and call redirection.
- ***Determines media capabilities*** - Via Session Description Protocol (SDP)
- ***Determines availability*** - Returns a message indicating why target was unavailable.
- ***Establishes sessions*** - Also supports mid-call changes (e.g. Adding another end point , codec changes, etc.)
- ***IP Centrex features*** - Handles transfer & hold

Elements of a SIP System

- SIP is a peer-to-peer protocol.
- The peers in a SIP session are called User Agents (UAs).



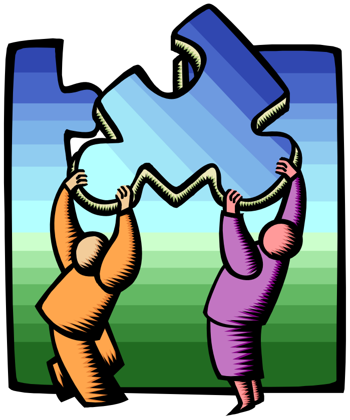
SIP Components

- User Agent Client (UAC) - A client application that initiates the SIP request
- User Agent Server (UAS) - A server application that contacts the user when a

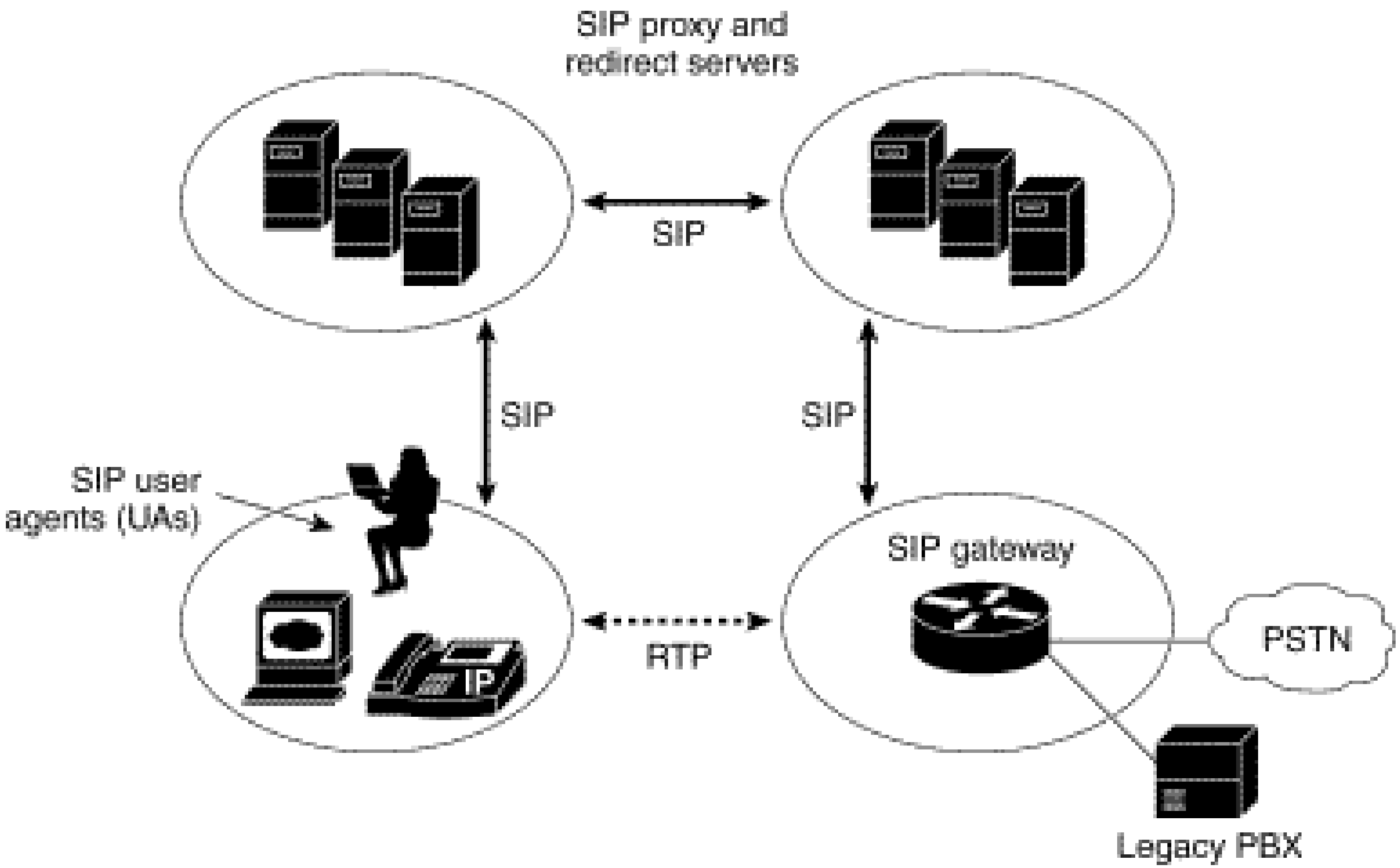
  SIP request is received and that

  returns a response on behalf of

  the user.



SIP Architecture



SIP Proxy Servers

- **Receives requests & forwards on client's behalf.**
- **Functions Include:**

  - Authentication

  - Authorization

  - Routing

SIP Redirect Servers

- **Provides client with next hop(s) information.**
- **Client will contact the next hop server or UAS directly.**

SIP Registrar Servers

- **Processes requests from UAC's for registration of their current location.**
- **Often co-located with a redirect or proxy server.**

SIP Message Requests

SIP uses six types of requests:

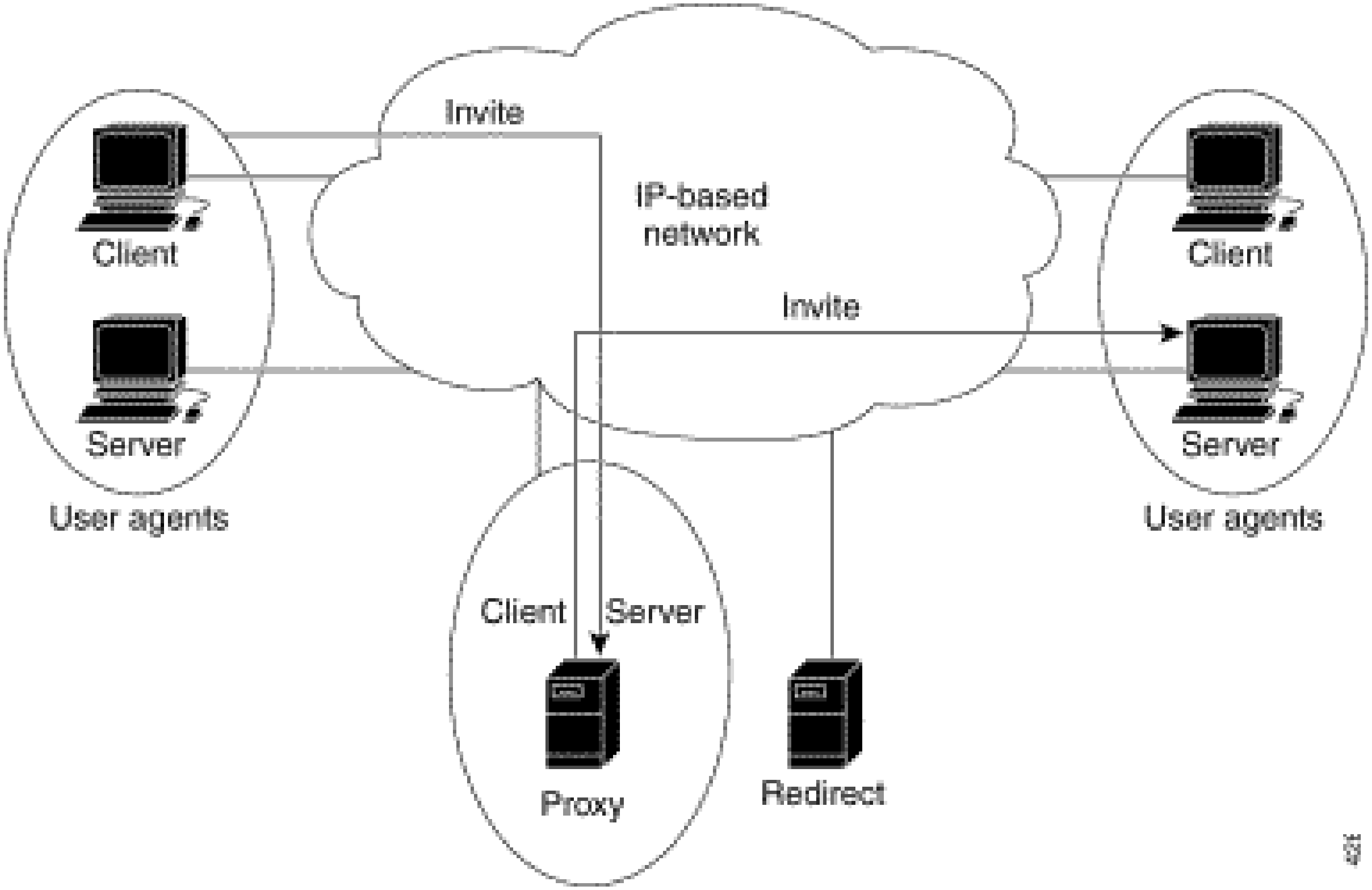
- **INVITE** - Indicates a user or service is being invited to participate in a call session.
- **ACK** - Confirms that the client has received a final response to an INVITE request.
- **BYE** - Terminates a call and can be sent by either the caller or the callee.
- **CANCEL** - Cancels any pending searches but does not terminate a call that has already been accepted.
- **OPTIONS** - Queries the capabilities of servers.
- **REGISTER** - Registers the address listed in the “To” header field with a SIP server.

SIP Message Responses

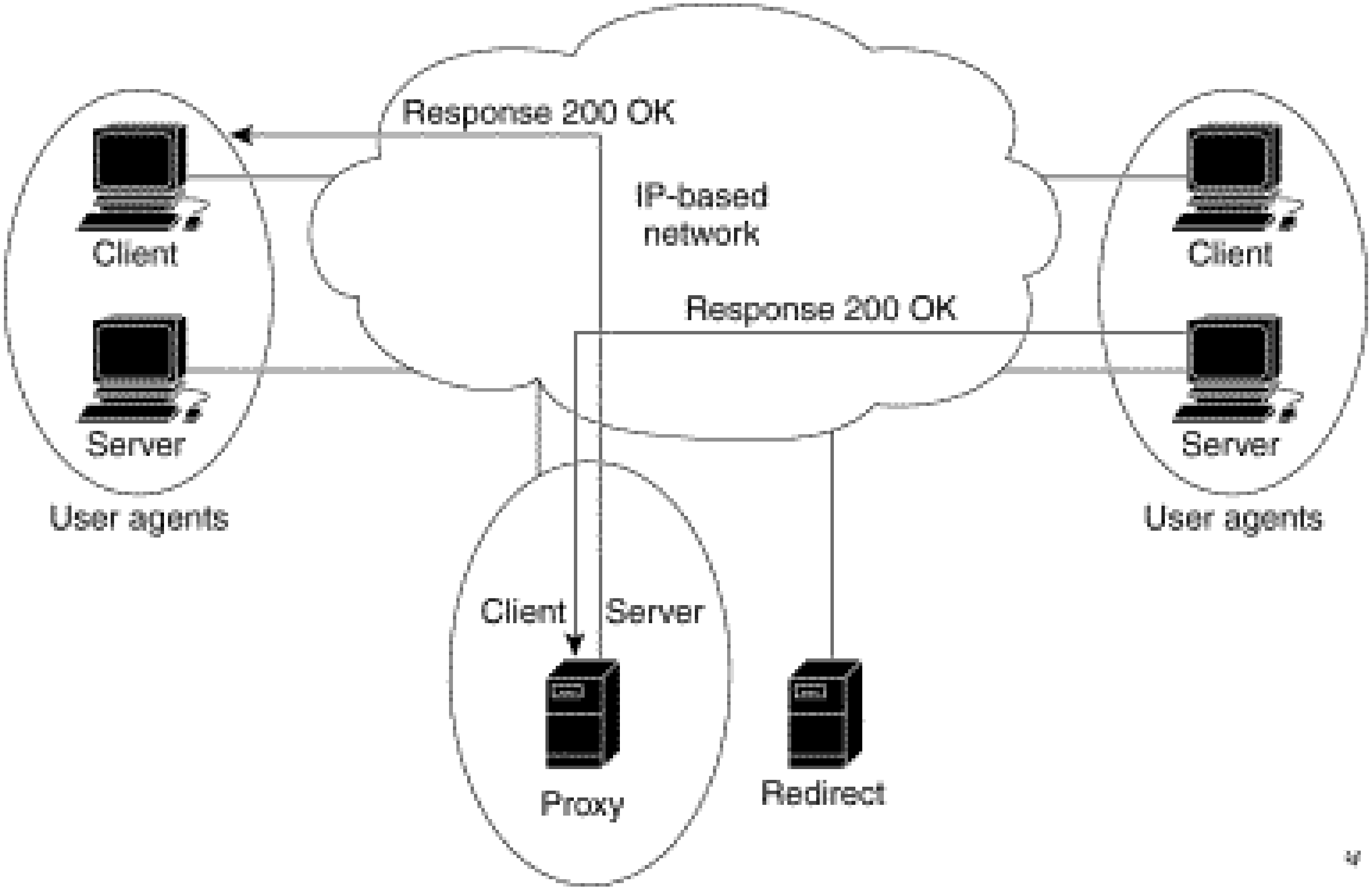
SIP uses six types of response :

- **SIP 1*xx*** - Informational Responses
- **SIP 2*xx*** - Successful Responses
- **SIP 3*xx*** - Redirection Responses
- **SIP 4*xx*** - Client Failure Responses
- **SIP 5*xx*** - Server Failure Responses
- **SIP 6xx** - Global Failure Responses

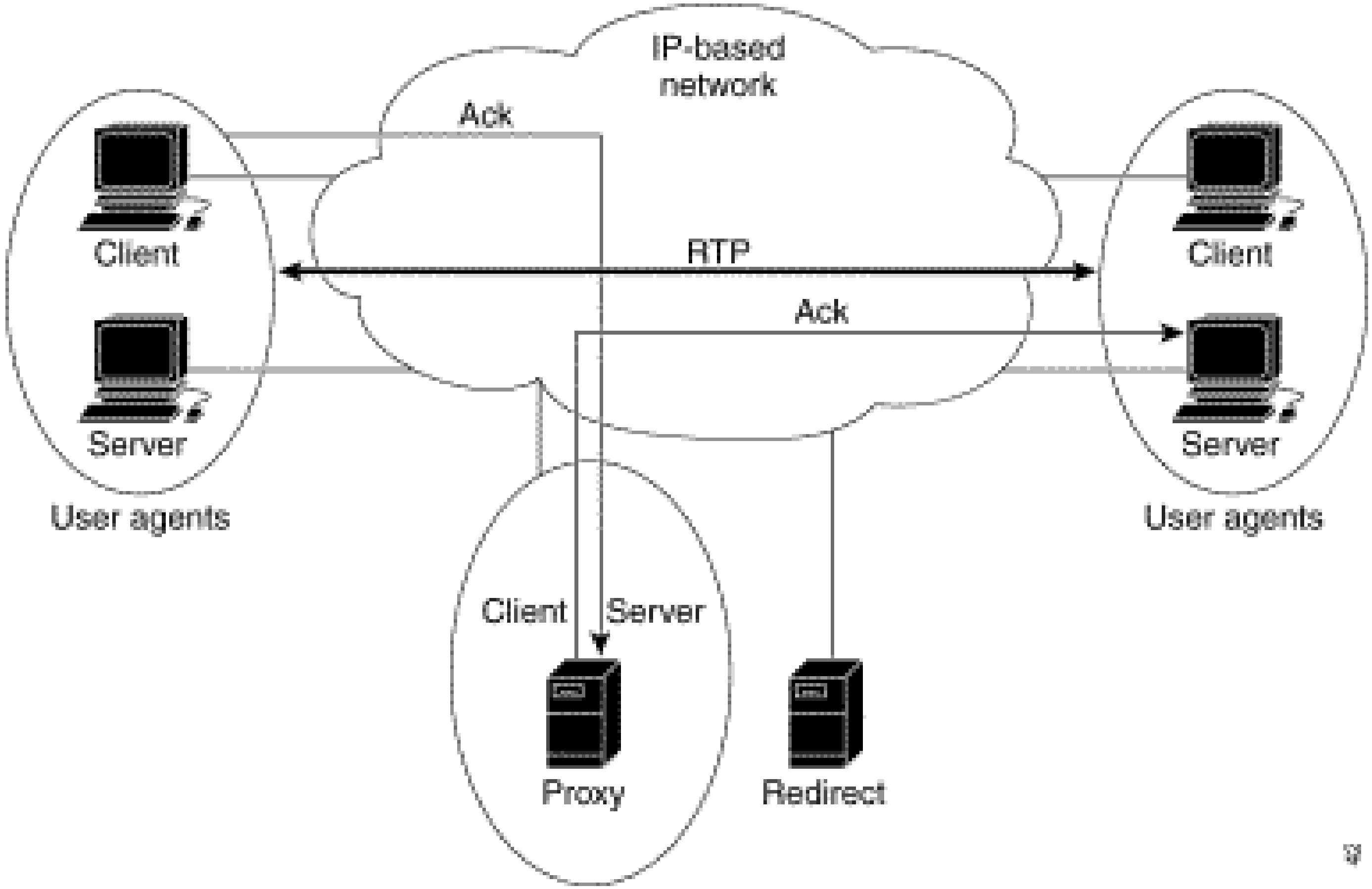
SIP Request Through a Proxy Server



SIP Response Through a Proxy Server

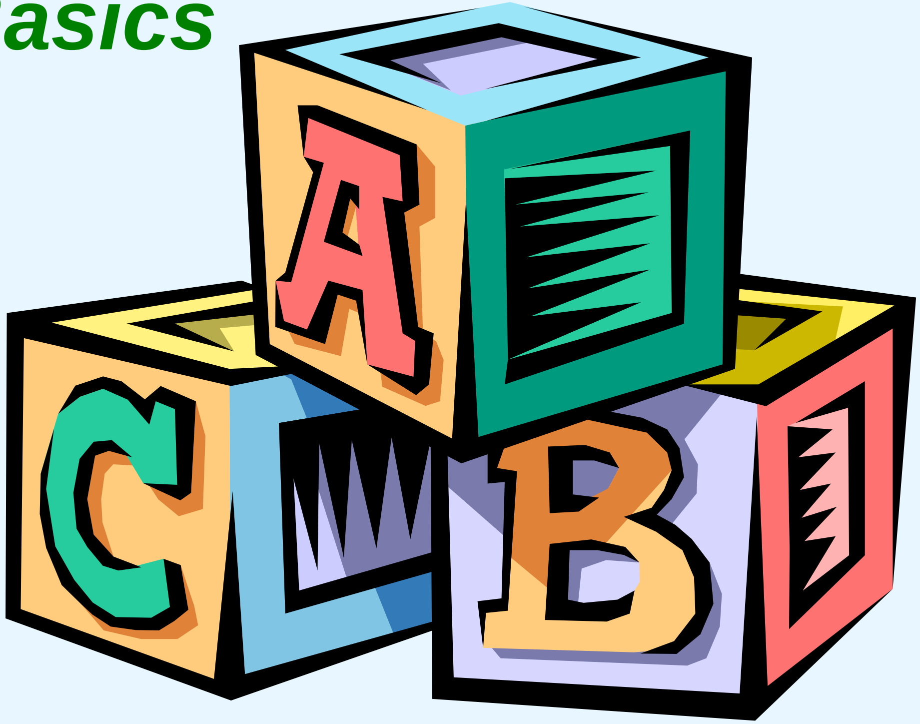


SIP Session Through a Proxy Server



SIP - Questions

Session Controller Basics



What is a Session Controller

A VoIP device to provide "any-to-any" vendor independent VoIP endpoint connectivity whilst also providing translation services, etc,
i.e. H.323 GateKeeper/SIP Proxy server.

Session Controller

- A breed of networking technology that provides ISO layer 5 routing and control to manage real-time traffic flows in IP networks.
- Technology addresses issues of:
  - Session routing
  - Signaling interoperability
  - Network security
  - Call admission control (CAC)
  - Service quality
- Session Border Control is a subset of this functionality localized at the network edge

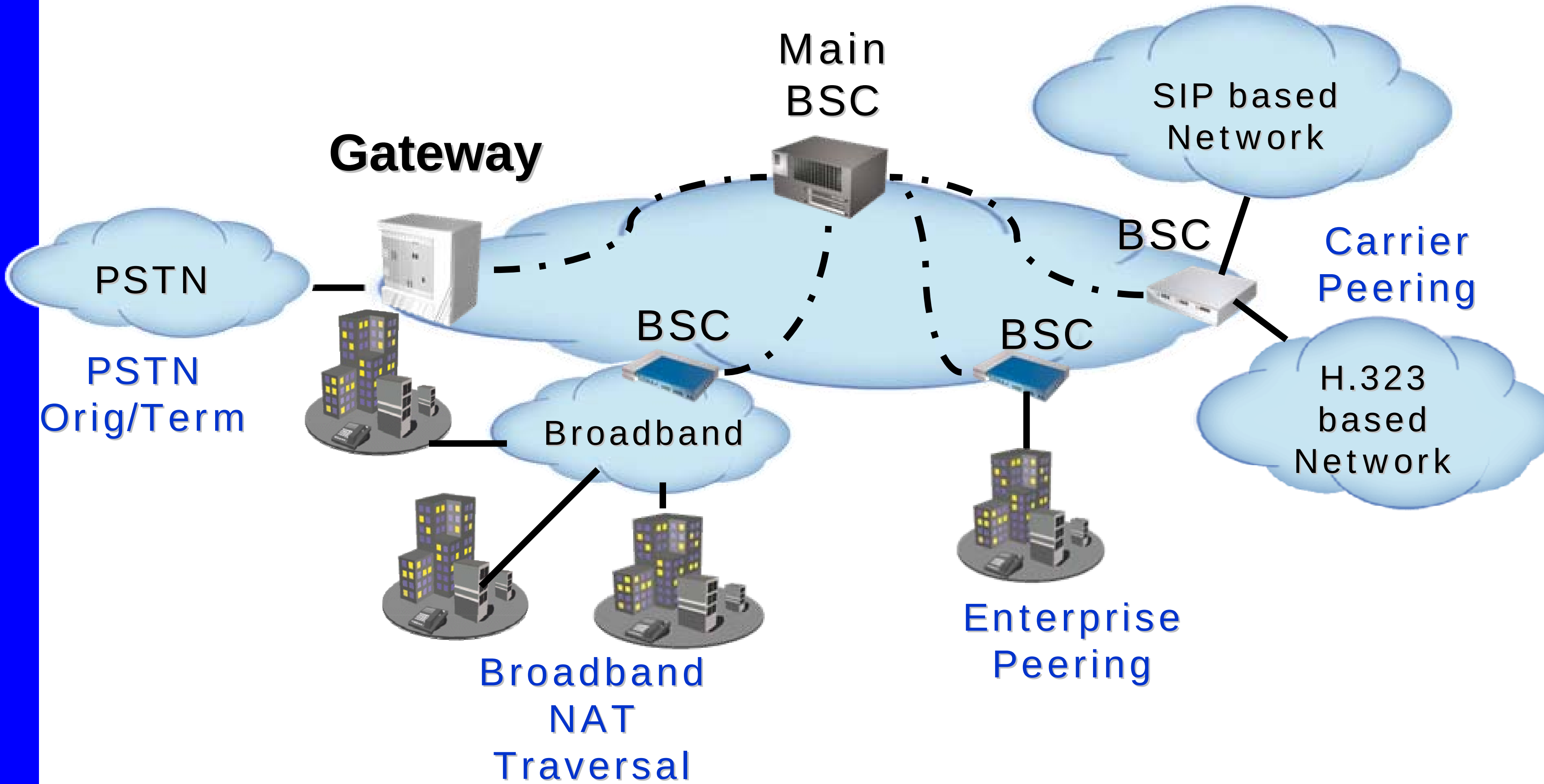
SC Market Trends

- The Session Controller is a key facility for the deployment of real-time services such as voice and video.

- VoIP Peering is happening today between international voice carriers.

SIP/H.323 interworking and VoIP CAC are key enablers within both applications.

SC Based Architecture



Session Controllers

- Border Session Controller (BSC)
  - Signaling based device deployed in the network
  - Programmable route engine
  - Centralized CDR collection
  - Call Admission Control on a network/local basis
  - Deployed to manage interconnects (both media and signaling)
  - Signaling Interoperability and interworking
  - NAT and Media Routing for network security