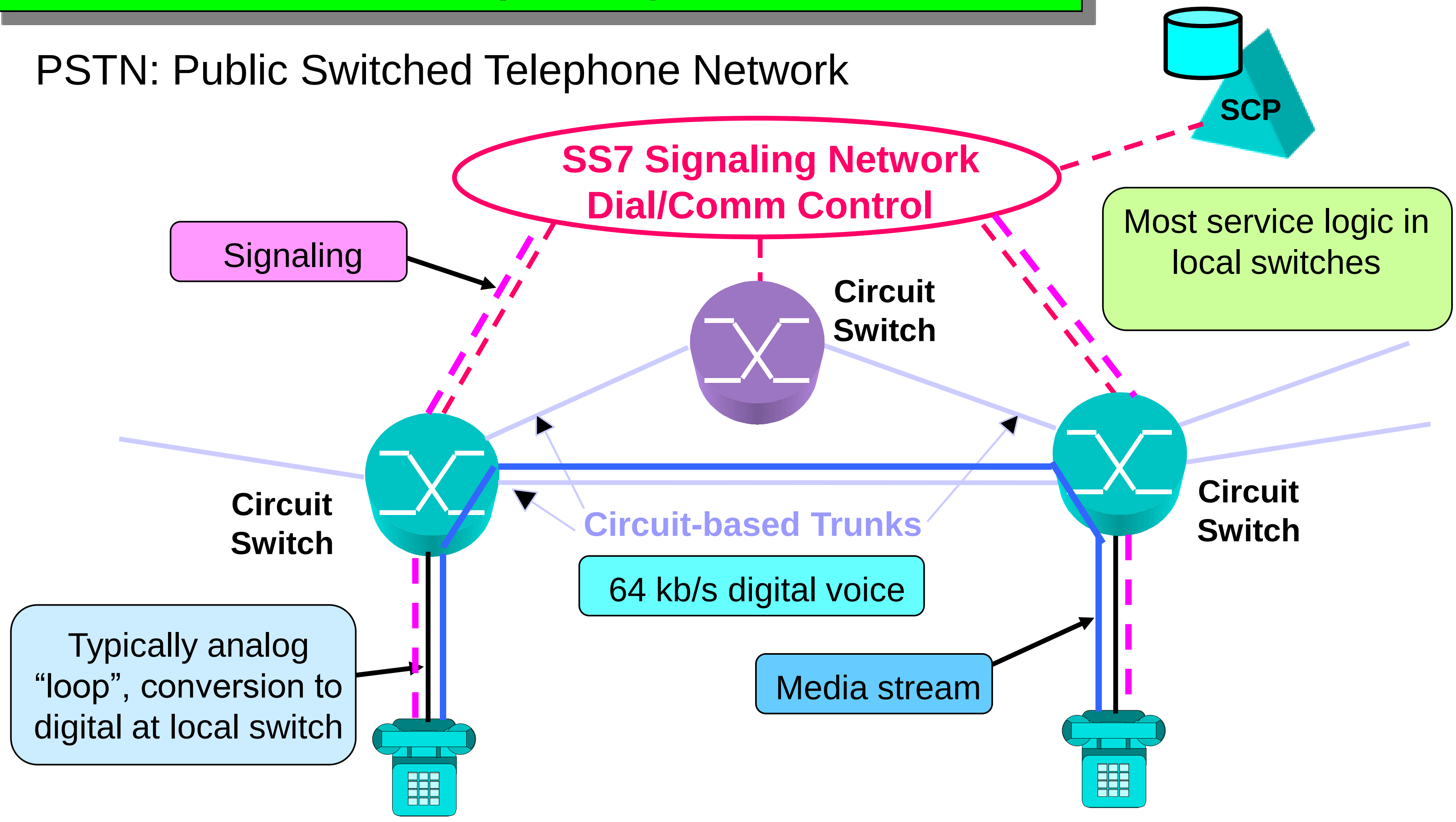
Media Communications
Internet Telephony and Teleconference

- Scenario and Issue of IP Telephony
- Scenario and Issue of IP Teleconference
- ITU and IETF Standards for IP Telephony/conf.
- H.323 Standard Series for IP Multimedia Comm.
- T.120 Standard Series for Data Conferencing
- SIP/SDP (Session Initiation/Description Protocol)

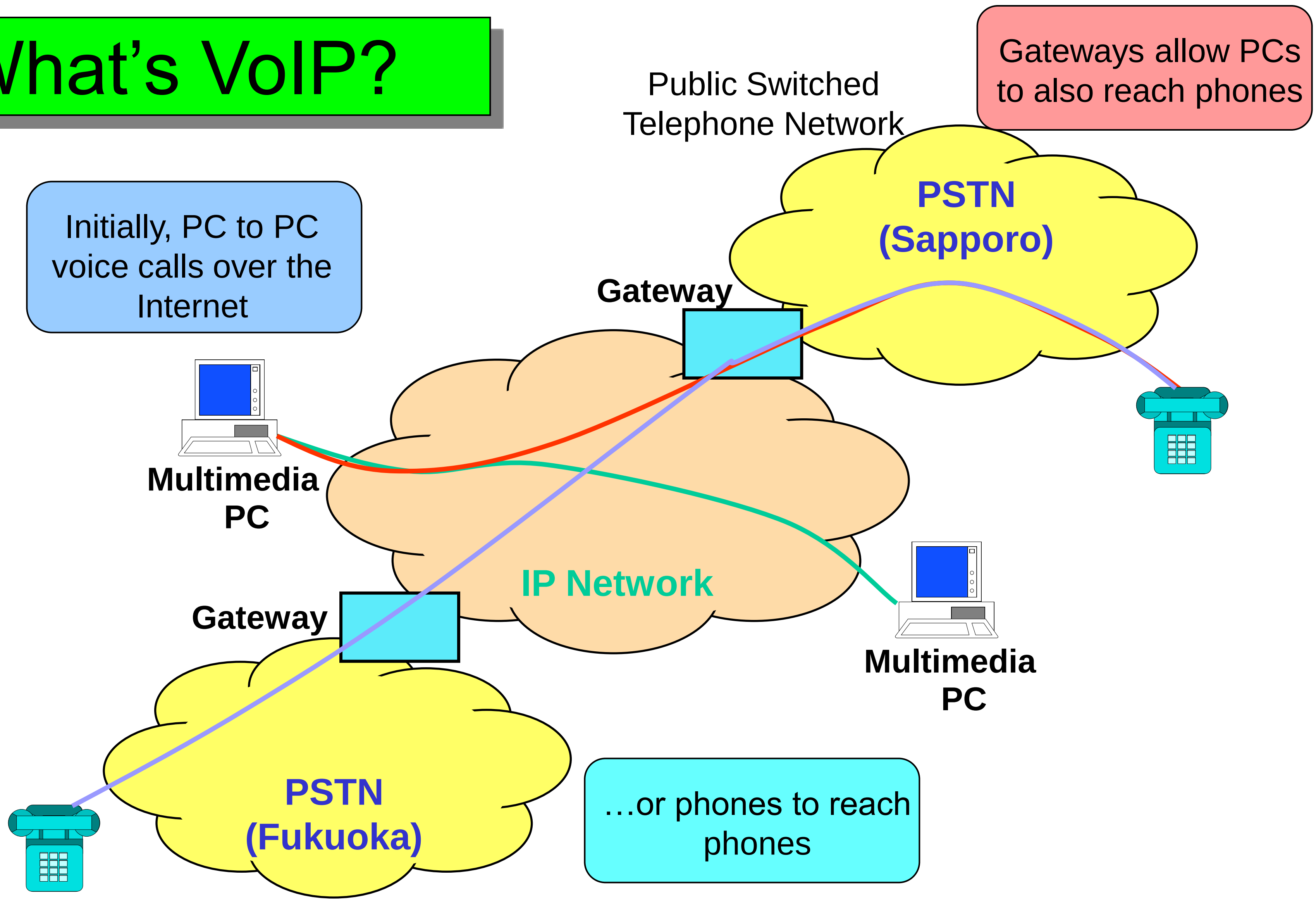
Traditional Telephony over PSTN

PSTN: Public Switched Telephone Network



- Different pair of telephones travels over a parallel/separate links
- Features: High voice quality, low bandwidth efficiency, inflexible

What's VoIP?


Public Switched
Telephone Network
Gateways allow PCs to also reach phones
Initially, PC to PC voice calls over the Internet
PSTN (Sapporo)
Gateway
Multimedia PC
IP Network
Gateway
Multimedia PC
PSTN (Fukuoka)
…or phones to reach phones

Packet-based Network (IP Network)

The data transmission method in computer communication is conceptually similar as the postal system. A large data stream will be divided into relatively small blocks, called packet, before transmission. Each packet is transmitted individually and independently over networks → **Packet-based Communication/Network**

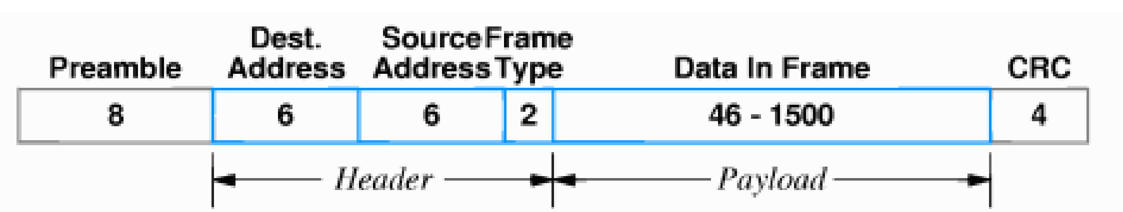
Original data stream: 10011…01 01001…11 … … … … … … … … … 10100…10

10011…01 | 01001…11 | • • • | 10100…10

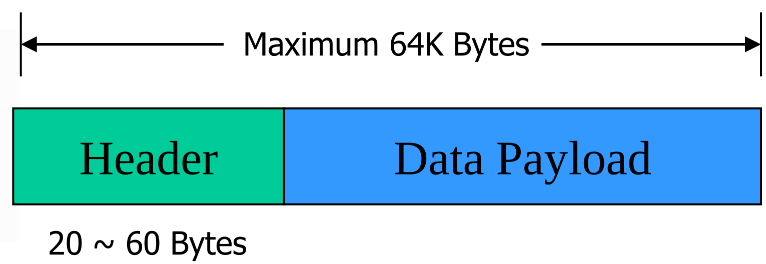
1st block | 2nd block | | Nth block

C-data 10011…01 | C-data 01001…11 | • • • | C-data 10100…10

1st packet | 2nd packet | | Nth packet

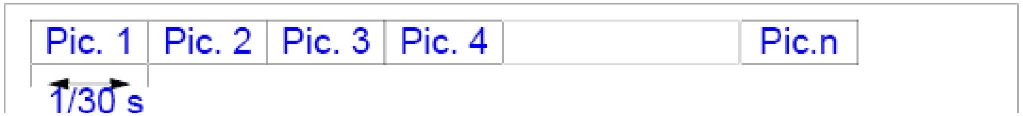


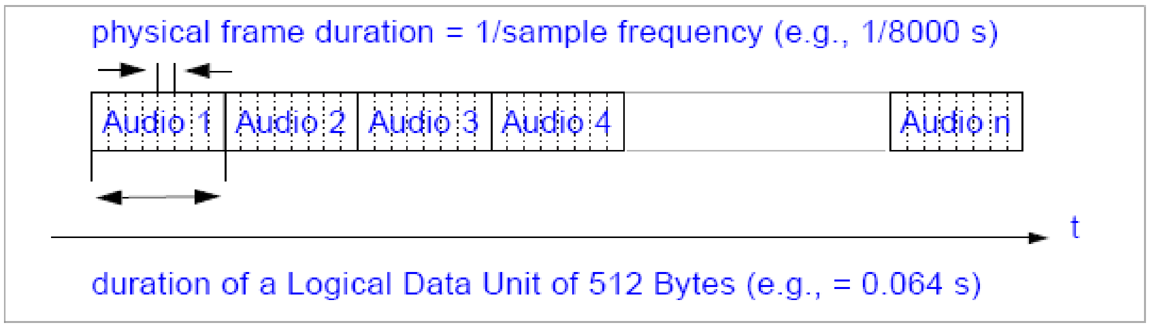
Ethernet Packet

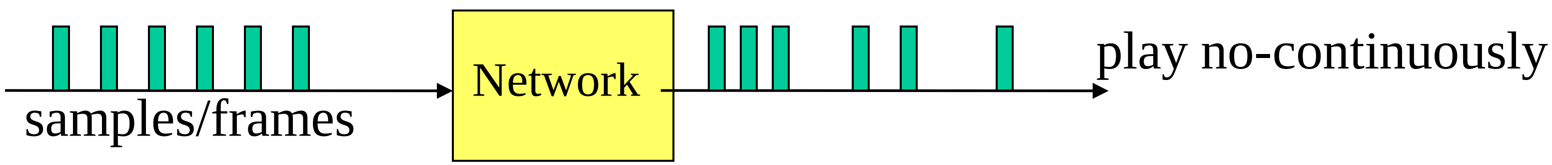


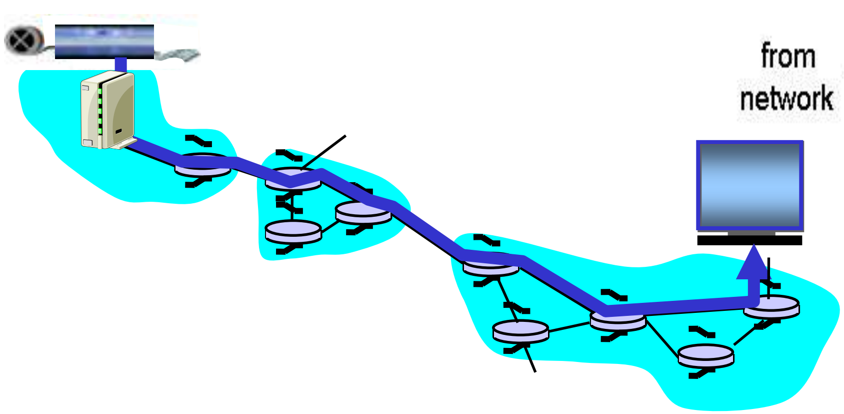
Internet Packet

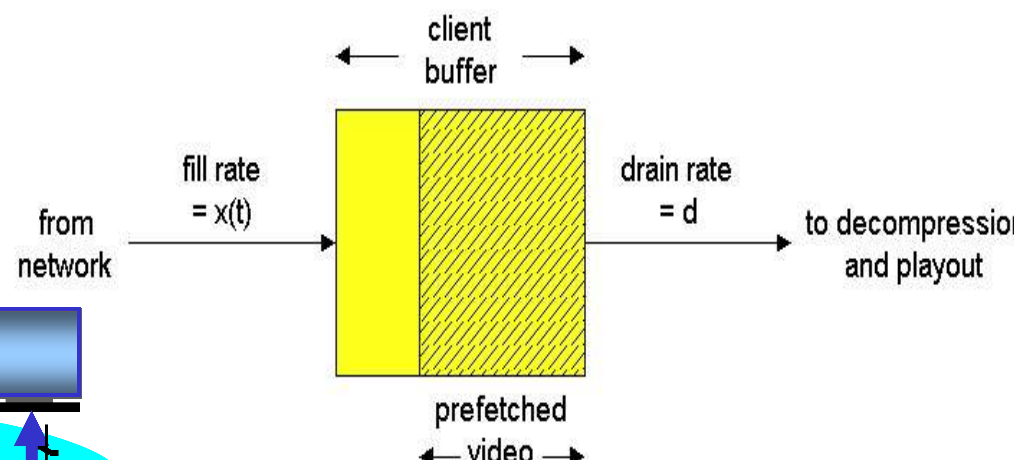
Temporal Relations in Video and Audio







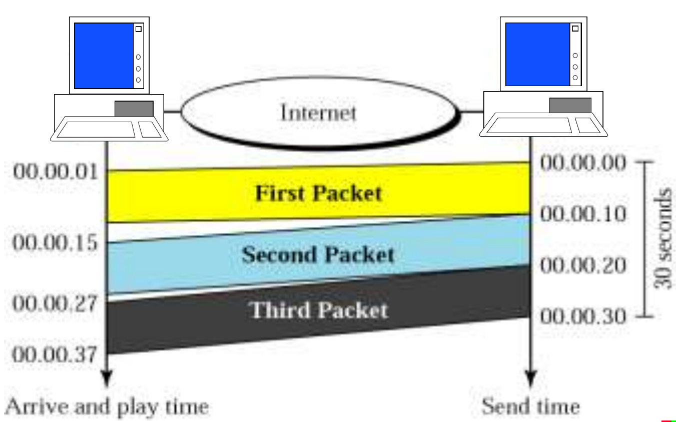


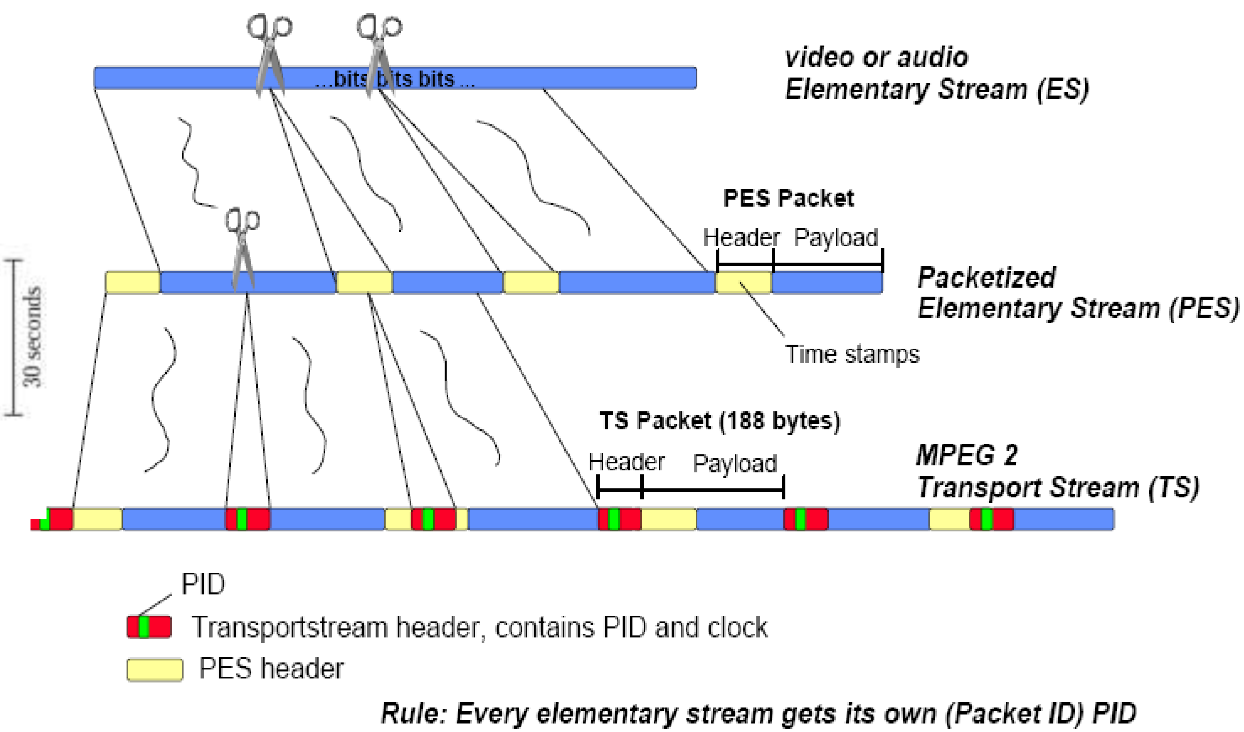


VoIP Basic Features and History

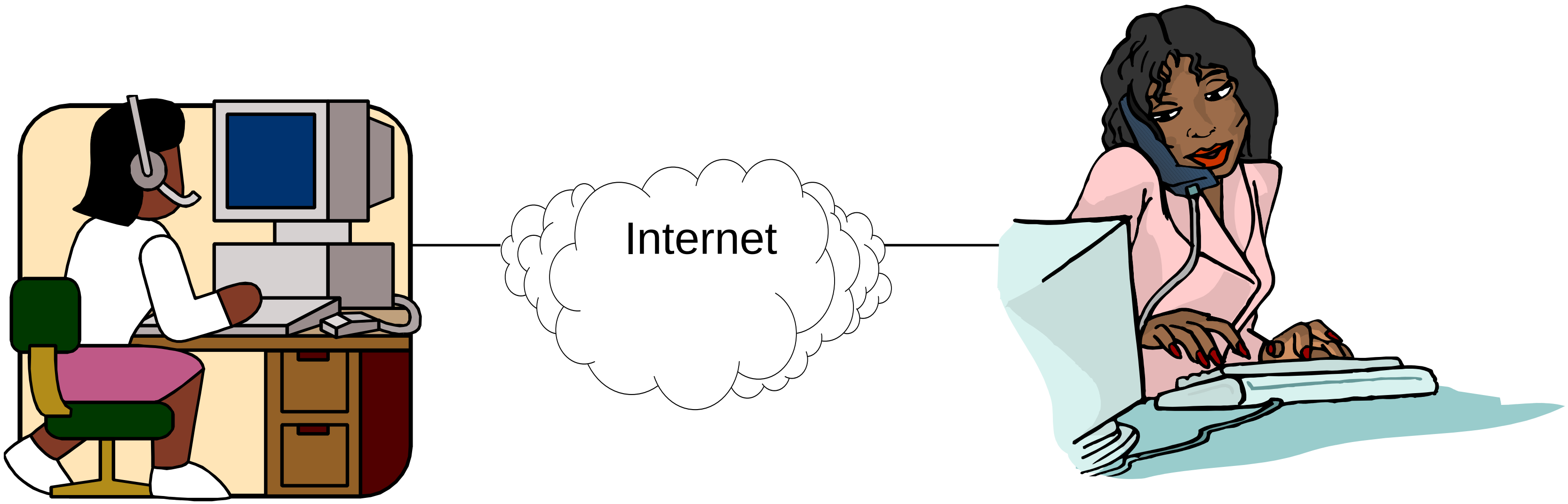
- Internet telephony, also called Voice over IP (VoIP), refers to using the IP network infrastructure (LAN, WLAN, WAN, Internet) for voice communication.
  IP (Internet Protocol) transmission unit: *packet*
- First product appeared in February of 1995:
  - Internet Phone Software by Vocaltec, Inc., “free” long distance call via PC
  - Software compressed the voice and sent it as IP packets.
- Other software/products soon followed → **NetMeeting, Skype, Gphone, …**

- Delay & jitter





Scenario 1: PC to PC



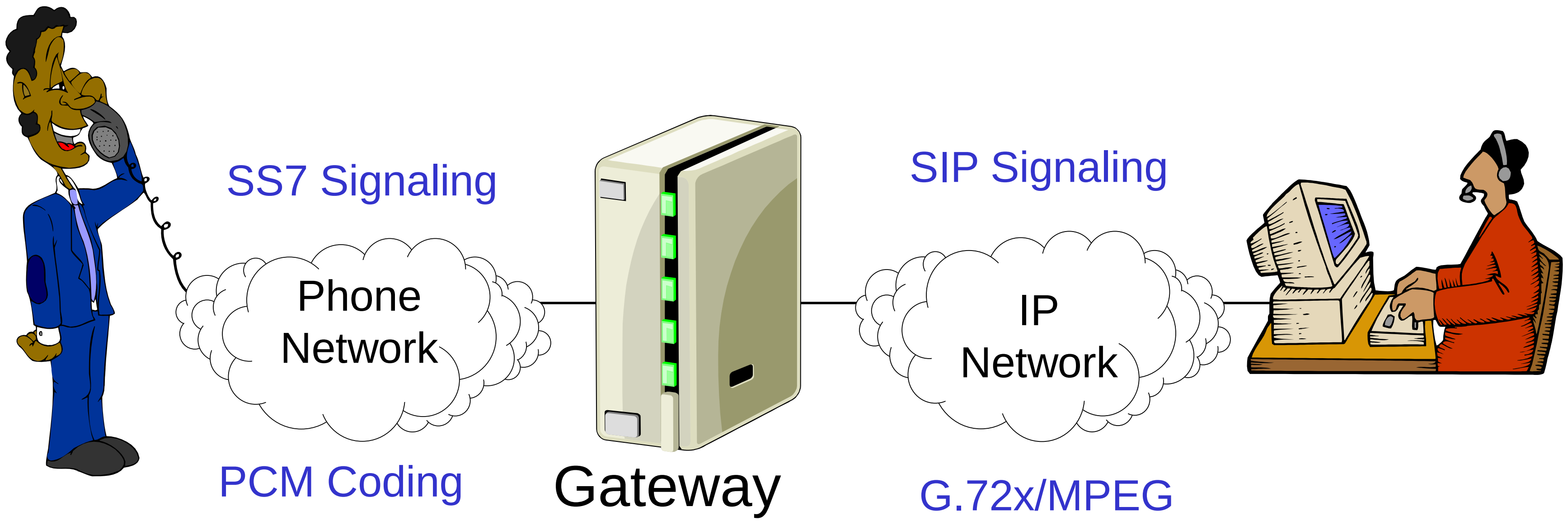
- Issues:
  - Addressing, i.e., VoIP phone number
  - Call admission, setup, control, release, etc
  - IP network related: delay, jitter, packet loss, out-of-order
  - Transmission overhead: Headers
  - Small delay
    → Small packet size

• •	RTP Header	UDP Header	IP Header	• •	Voice data

Total > 100 bytes Can't be large for voice delay

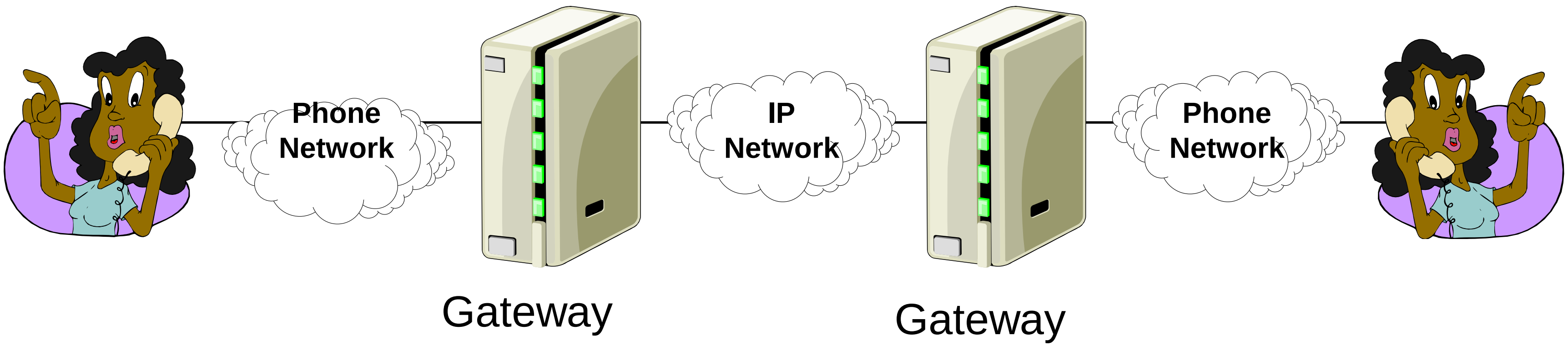
Voice data rate: 1~8KBytes/Second
or 8~64Kbps (bits-per-second)

Scenario 2: PC to Phone



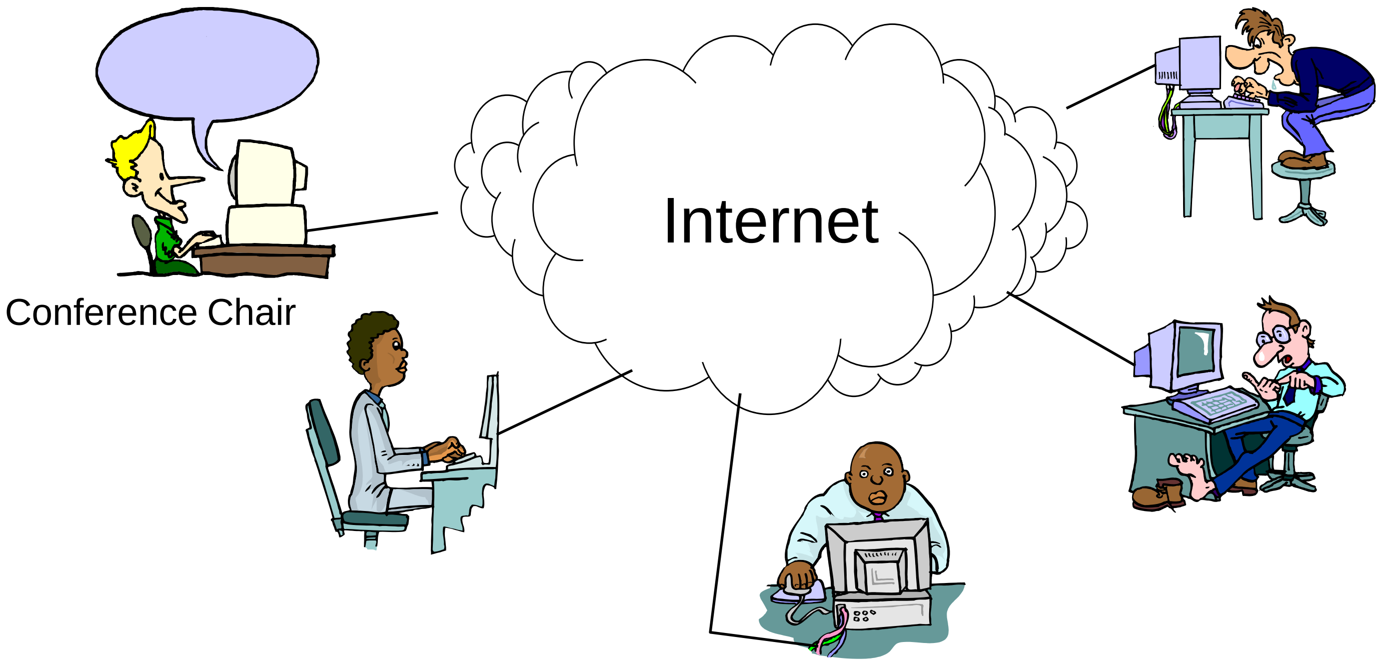
- A Gateway is needed to connect the PSTN to the IP network:
  - Signaling conversion
  - Format conversion

Scenario 3: Phone to Phone



- Gateways will connect the phone network to the IP network.
- The IP Network can be a dedicated backbone or intranet (to provide guaranteed QoS) or can be the Internet (no guarantees …)
- The phone network can be a company PBX (Private Branch Exchange) or carrier switches

What is Internet Teleconference

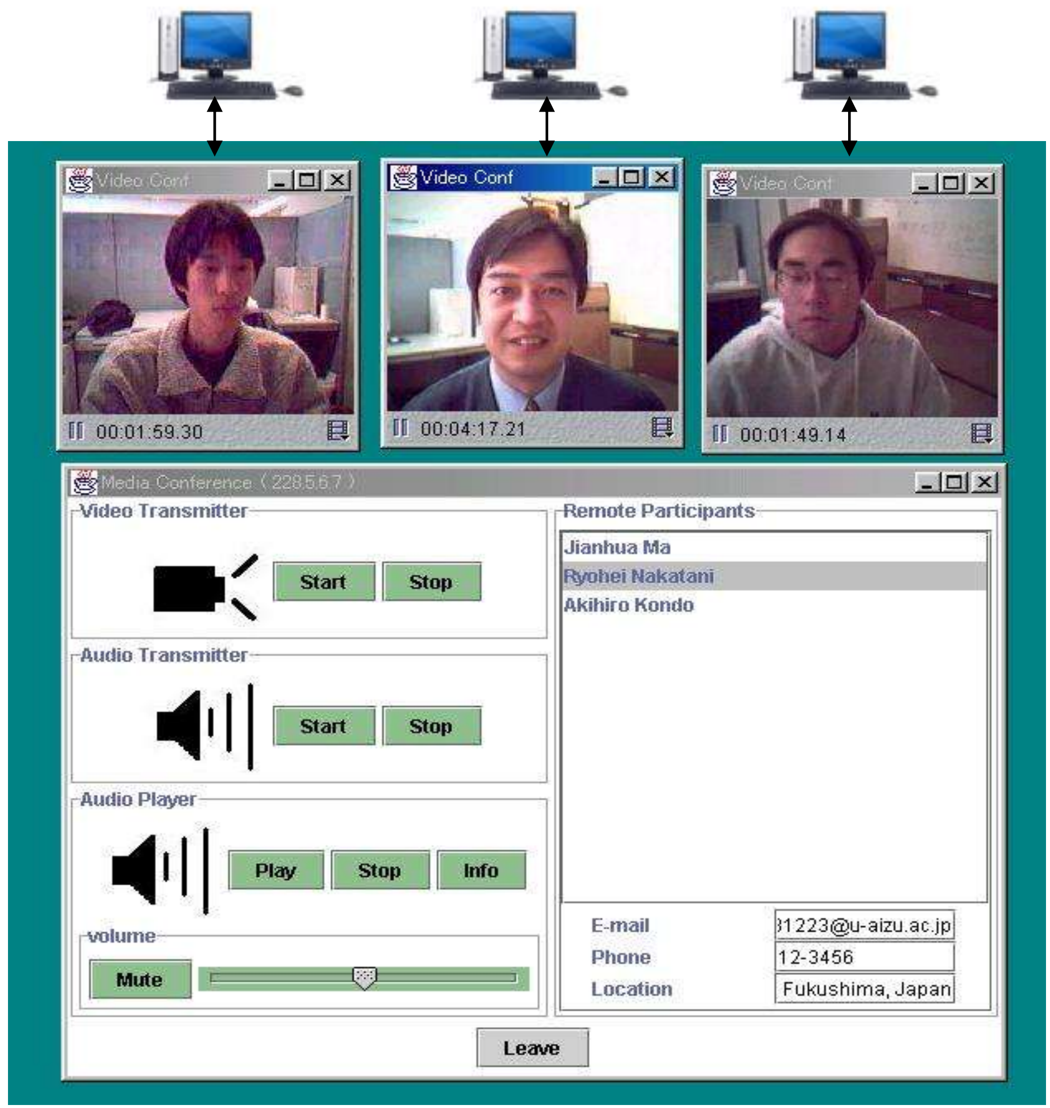


Internet teleconference: A group of people communicate each other via voice, video and/or other data over the Internet

- Conference initiation, start, join, leave, end, control, etc.
- Sending audio/video data from one-to-many (multicast)
- Sharing other conference data (data conferencing) among all participants
- Synchronization and network delay, jitter, packet loss, …

Example of Audiovisual Conference

NetMeeting



What is Data Conferencing?

Data conferencing is a virtual connection between two or more computers where:

- All computers in the conference display a common graphical image of text, graphics or a combination of both.
- Each computer in the conference displays any changes to the common image in near real time.
- Participants have ability to interact with the displayed document
- WYSIWIS: What You See Is What I See

Presentation (group broadcast)

- Broadcast event where a single presenter's electronic presentation is distributed to multiple remote computers.

Collaboration (group meeting)

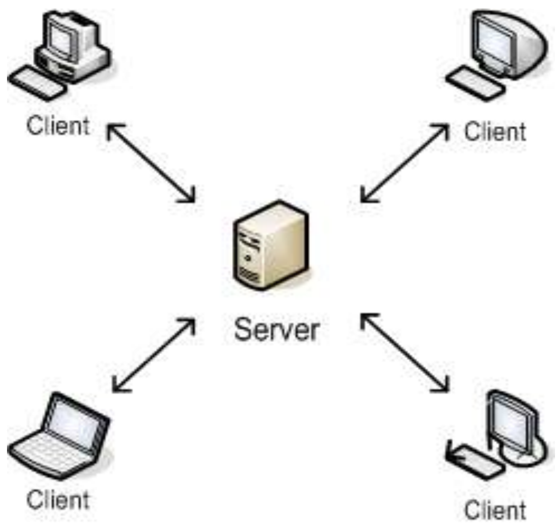
- Everyone can talk, operate, …
- Usually involves a small conference of 3-10 participants
- Two types of Collaboration: Whiteboarding & Application Sharing

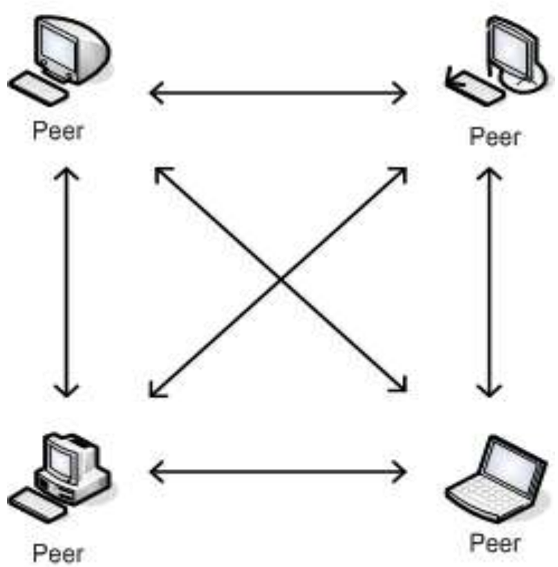
Example of Data Conferencing: VCR

Example of Tele-Conference Rooms

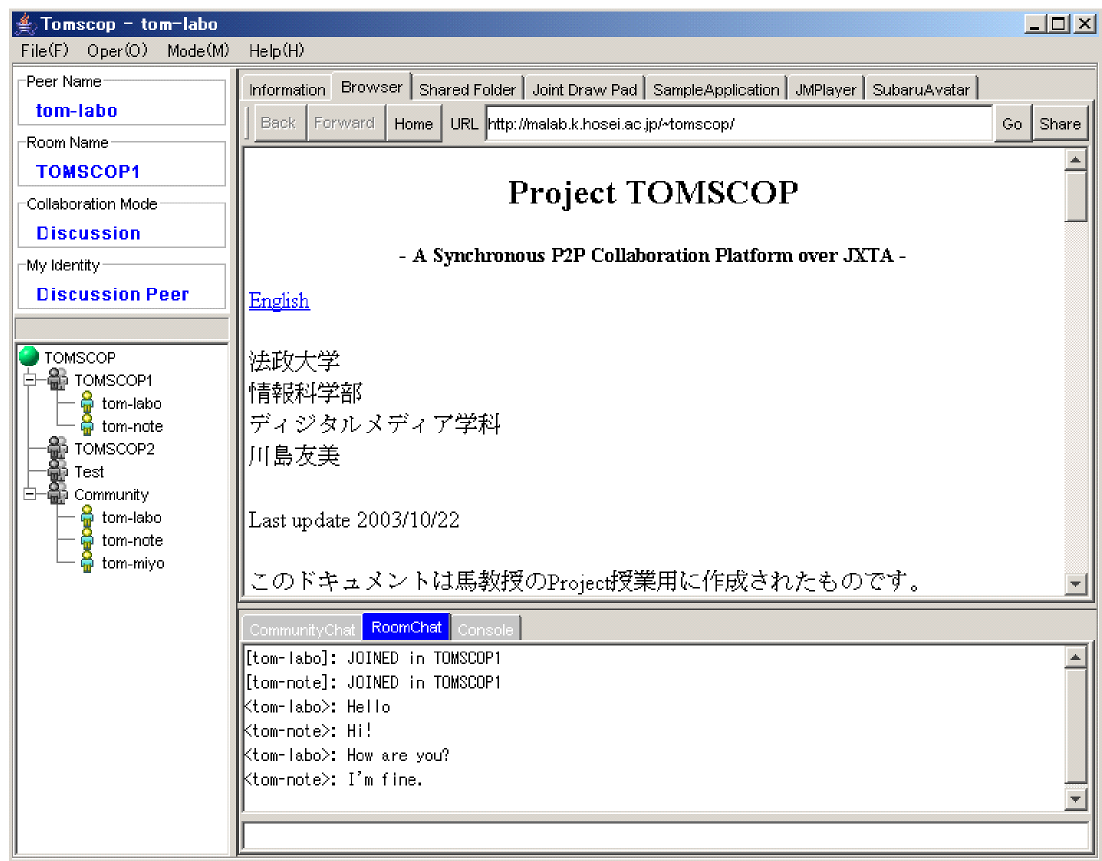
Control Center

Server-Client & P2P Communication Models

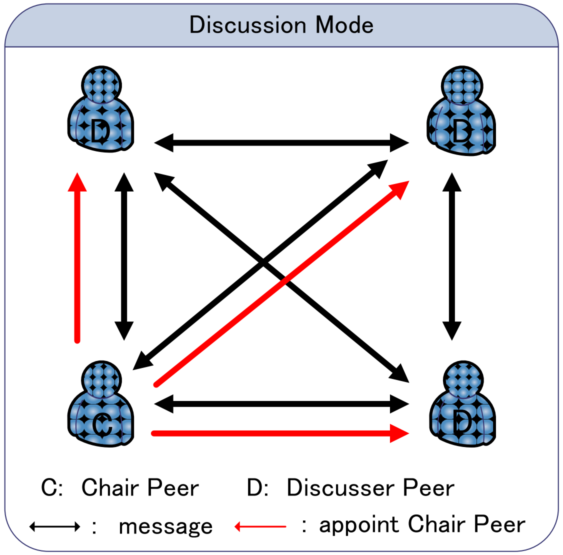


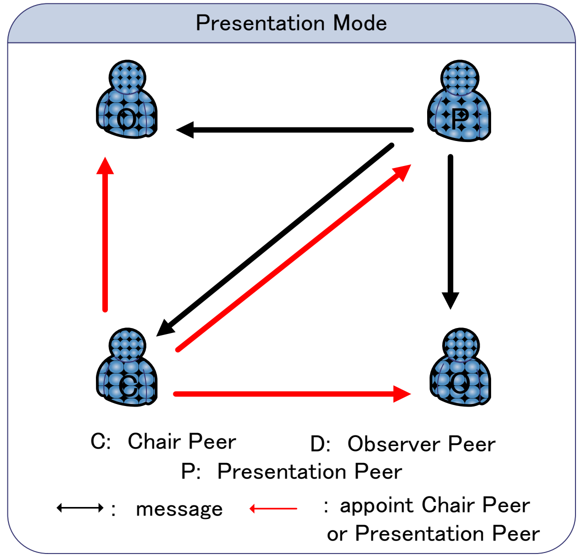


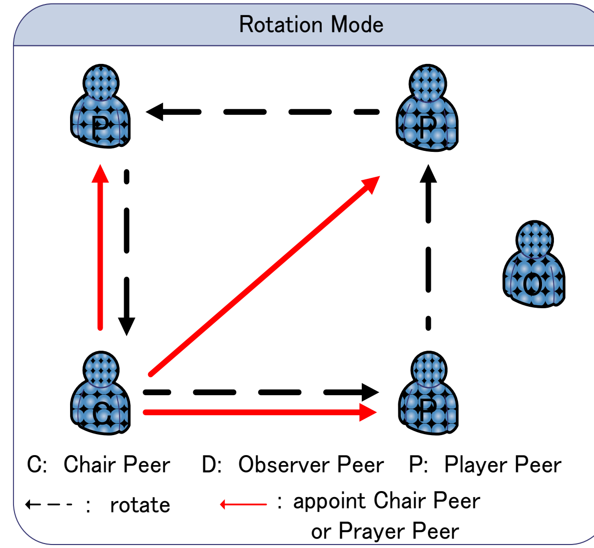
- Client/Server model: TANGO, Habanero, **VCR**
  Problem: load, cost, system down
- Peer-to-Peer model: DSC, Groove, **TOMSCOP**
  Problem: difficulty of peer/group management

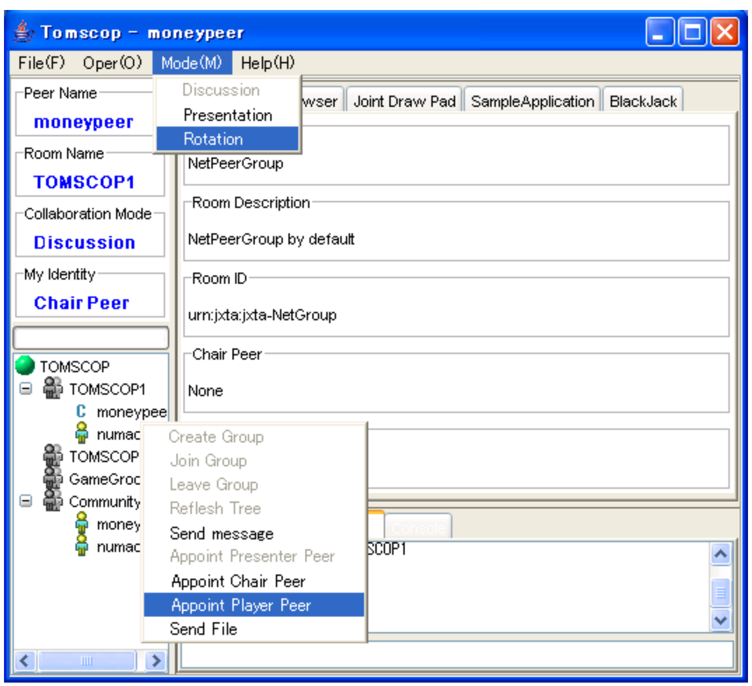


Peer Identity and Collaboration Modes







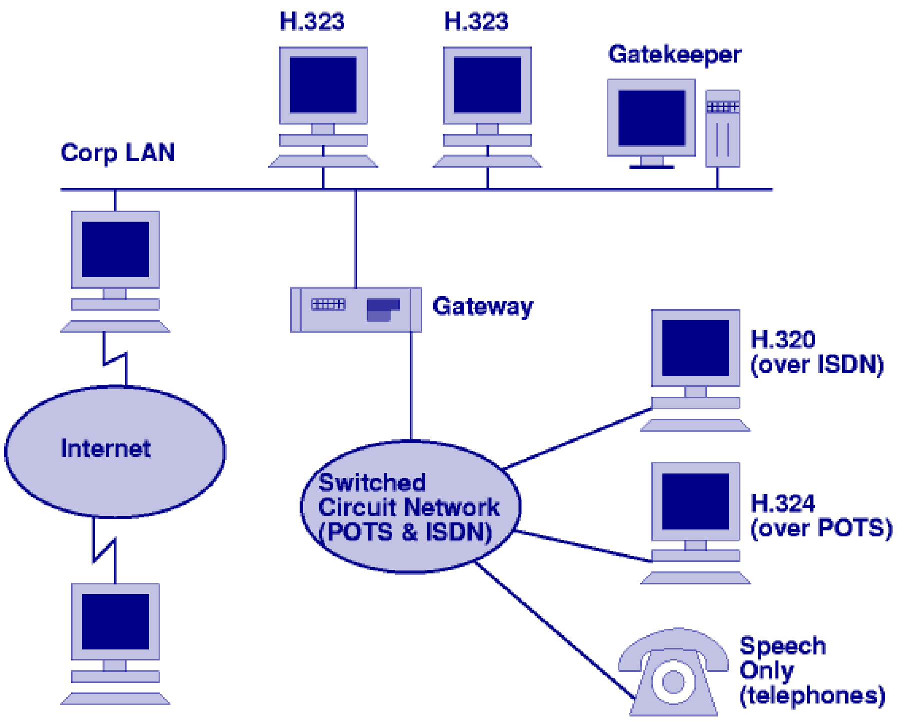


Typical Standards: H.323 & SIP

- Self-developed communication software/middleware
- Implementations of Internet telephony and conference can use two types of popular standards
  - **H.323** standards from ITU (1996, 1st Version)
    * Adopt some protocols (RTP/RTCP) from IETF
    * More implementations
    * Very complex
    * Poor interoperability between vendors
  - **SIP** standards from IETF (1998, 1st Version)
    * Similar functions as H.323
    * Relatively easy because of textual natural instead of binary
    * Better interoperability
    * Under going and improvement, e.g., security

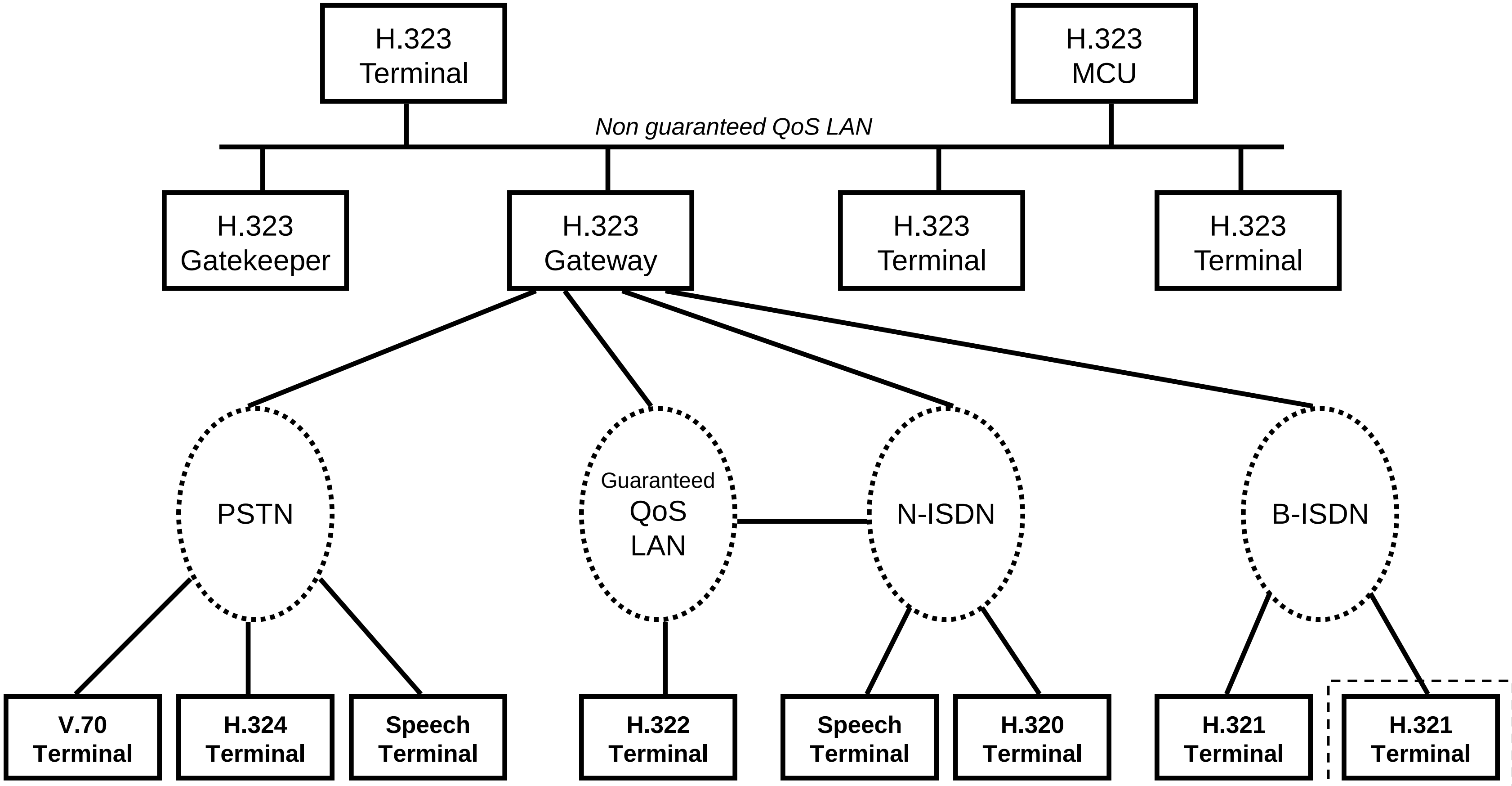
H.323 History

- H.323 is a product of ITU-T Study Group 16.
- Version 1: "*visual telephone systems and equipment for LANs that provide a nonguaranteed quality of service (QoS)*" was accepted in October 1996.
  - Focus on multimedia communication in a LAN
  - No support for guaranteed QoS
- Version 2: "*packet-based multimedia communications systems*" was driven by the Voice-over-IP requirements and was accepted in January 1998.
- Version 3 was accepted in September 1999 and has minor incremental features (caller ID, …) over version 2.
- Version 4 was accepted in November 2000 and has significant improvements over version 3.



H.323 System

H.323 Entities: Terminal, Gatekeeper, Gateway, MCU (Multipoint Control Unit)



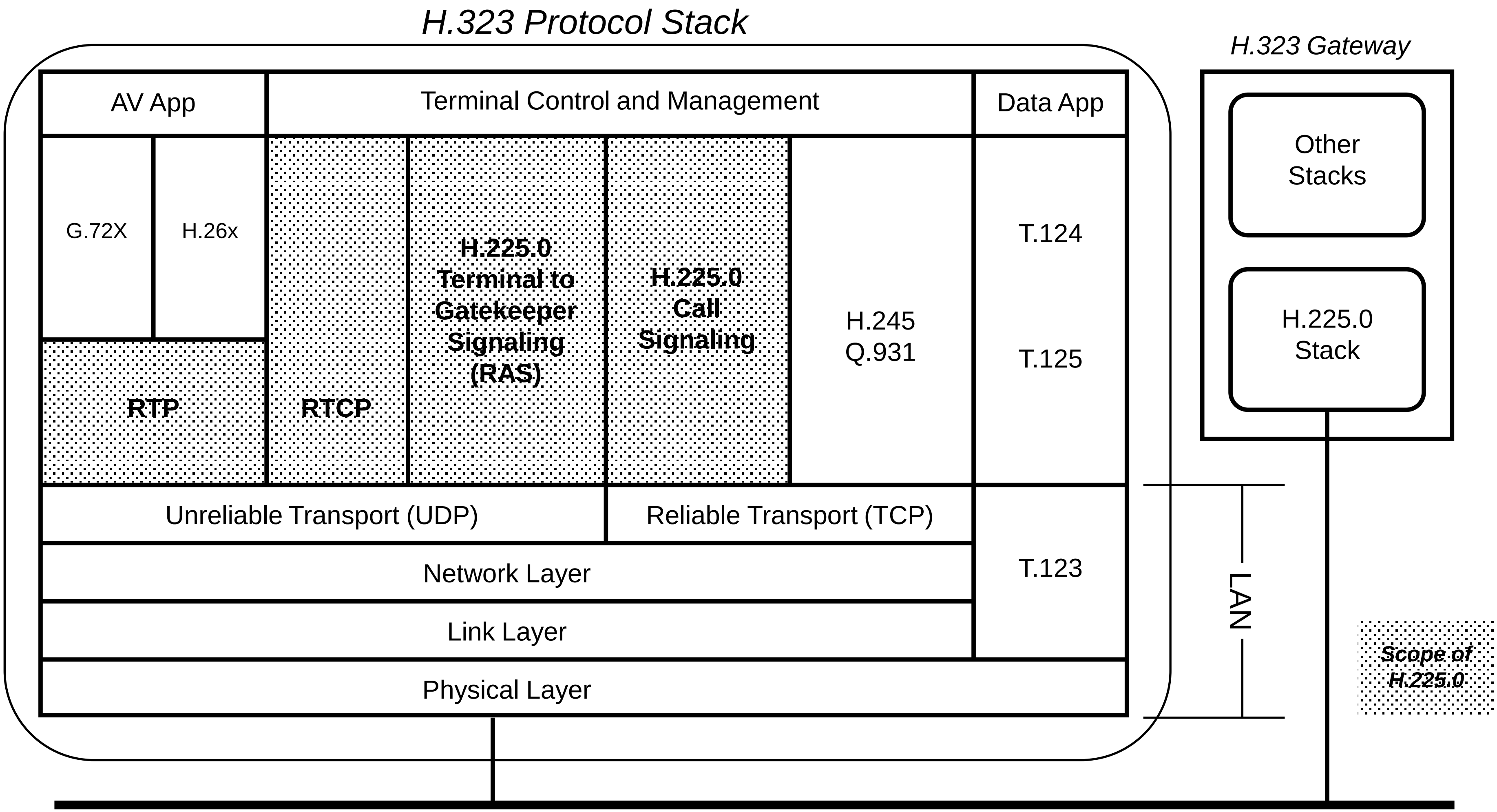
- H.310 (B-ISDN)
- H.320 (N-ISDN)
- H.321 (ATM)

- H.322 (GQOS-LAN)
- H.324 (GSTN), H.324/M (mobile phone, 1998)
- V.70 (DSVD - Digital Simultaneous Voice & Data)

H.323 Entities

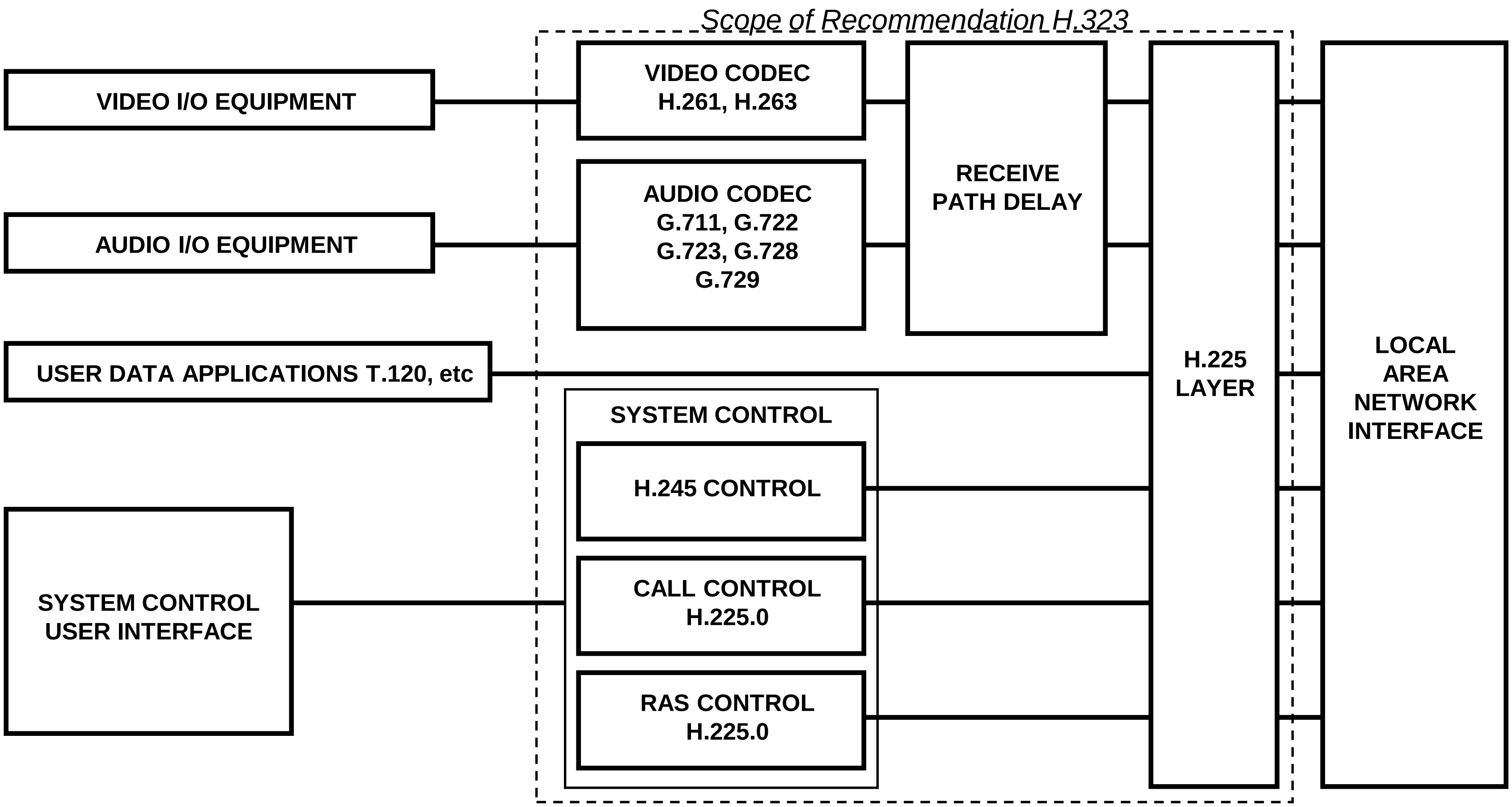
- **Terminal**
  - An endpoint on the LAN which provides for real-time, two-way communications with another H.323 terminal, Gateway, or MCU
  - May provide audio, video, and/or data
- **Gatekeeper**
  - Provides address translation and controls access to the LAN
  - Performs bandwidth management
- **Multipoint Control Unit (MCU)**
  - Provides the capability for 3 or more terminals and Gateways to participate in a multipoint conference
- **Gateway**
  - Provides for real-time, two-way communication between H.323 terminals on a LAN and other ITU terminals on a wide-area network or another H.323 Gateway

H.323 Protocol Stack



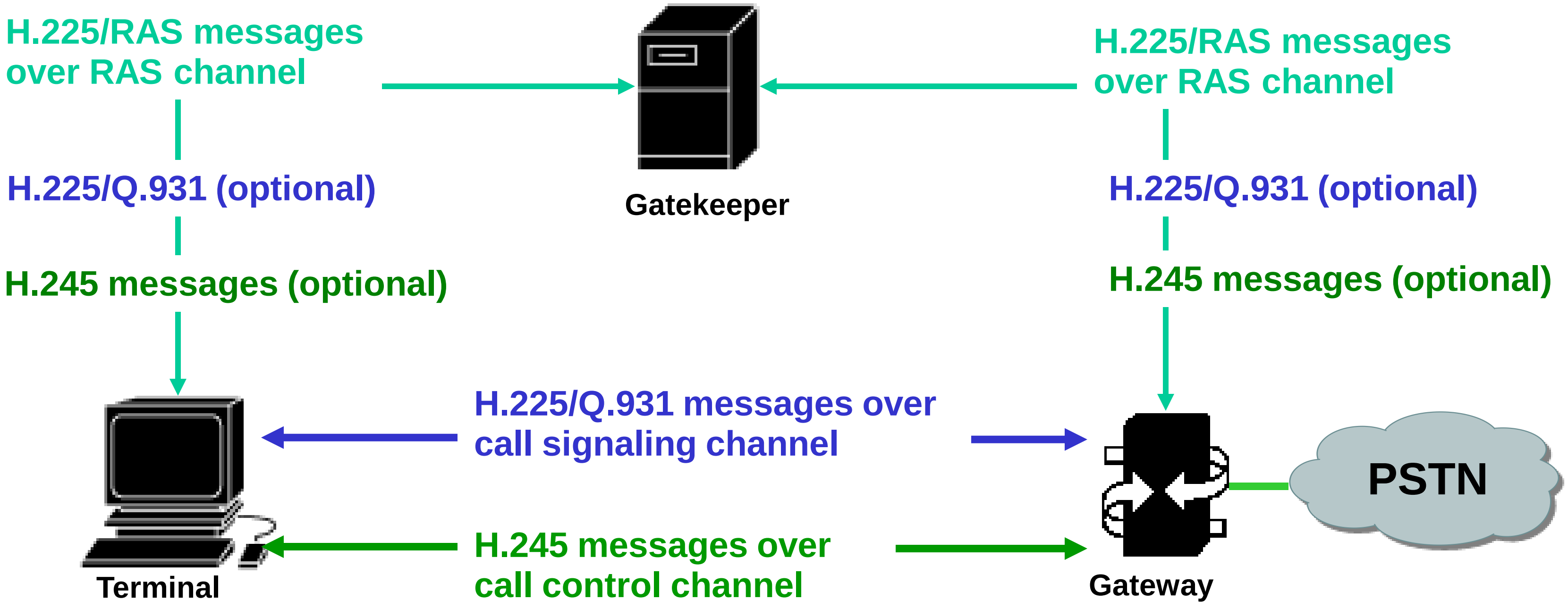
RAS: Registration, Admission, Status

H.323 Terminal

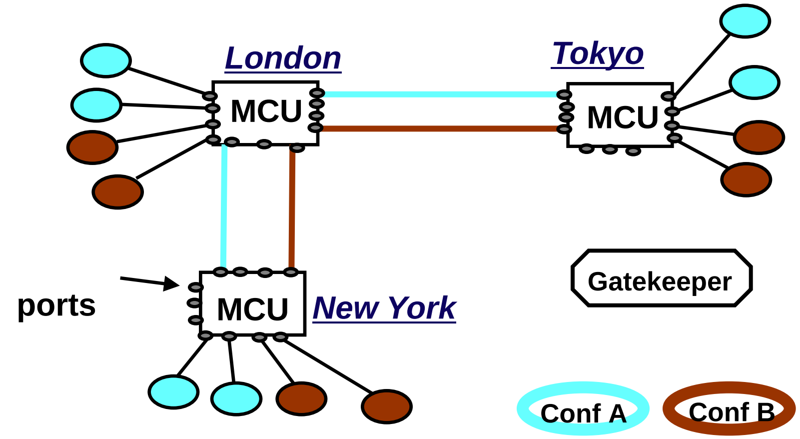


Gatekeeper

- Provides the following services:
  - Address translation between Transport Addresses and Alias Addresses
    # Transport Addresses: LAN IP Address + TSAP Identifier (port number)
    # Alias Addresses: phone number, user name, email address, etc.
  - Admission control based on authorization, bandwidth, or other criteria
  - Dynamic bandwidth control during a conference
- Transport address for the H.245 Control Channel is exchanged on the Call Signaling Channel

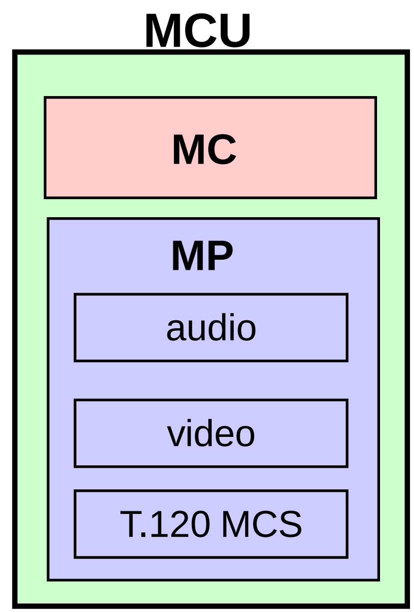


Multipoint Entities & MCU

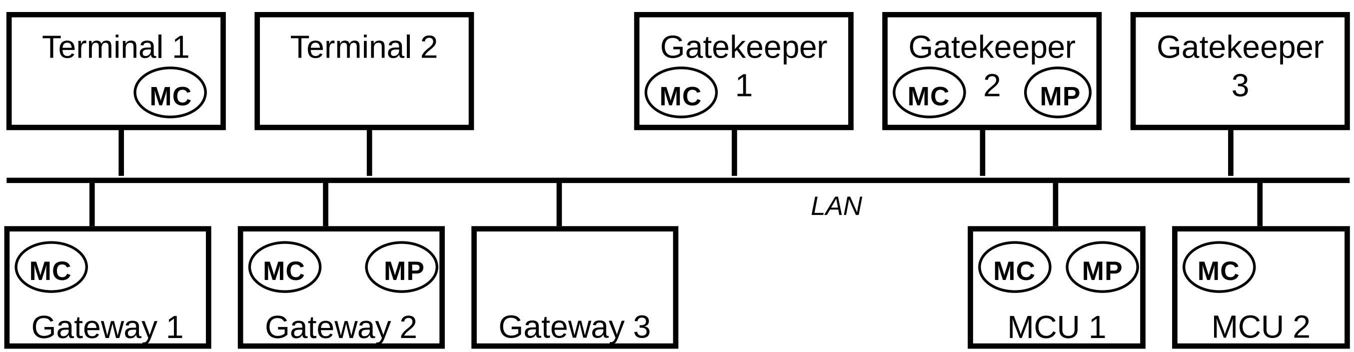


MC: Multipoint Controller, **MP**: Multipoint Processor

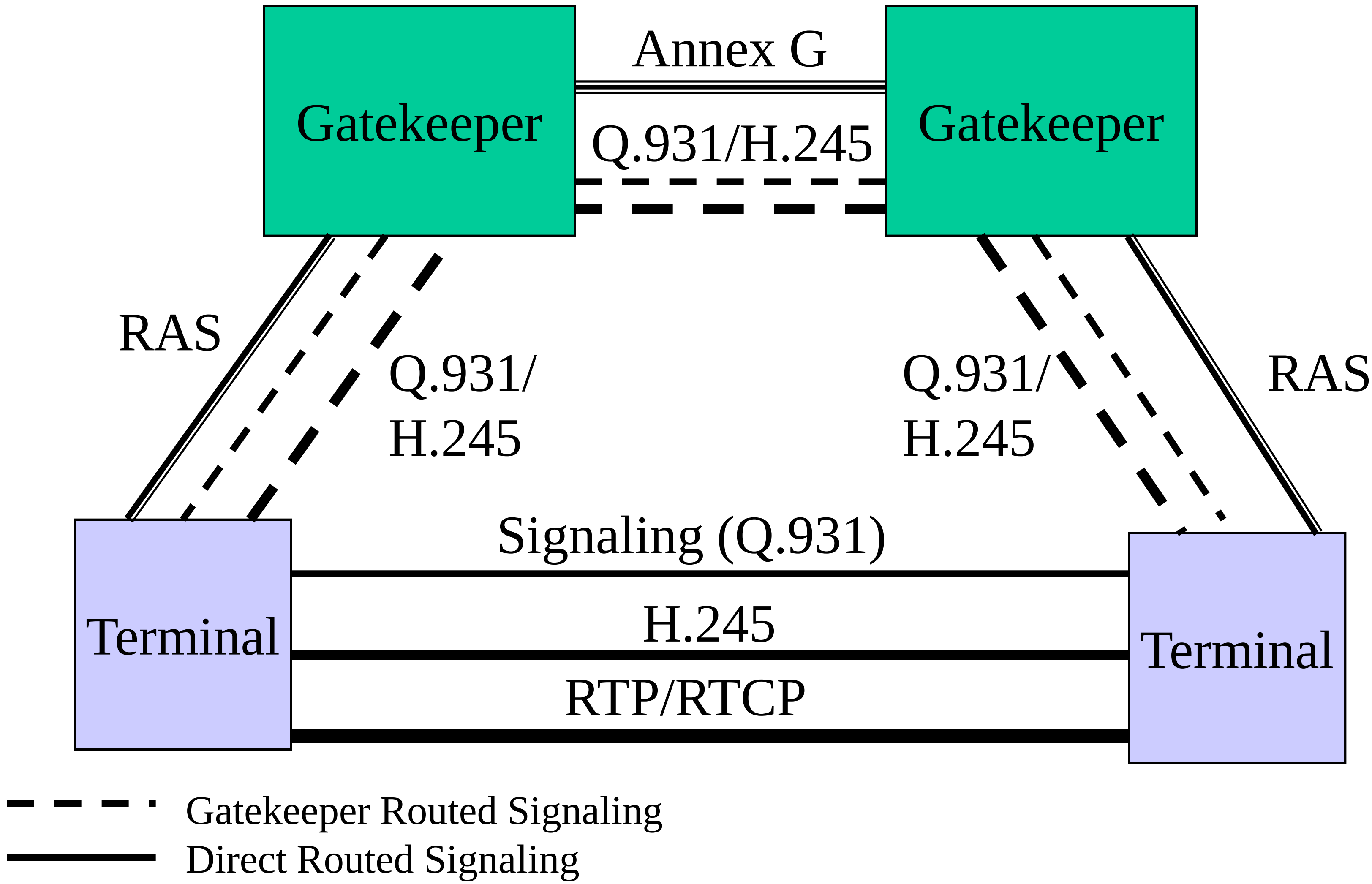
- **MC** performs capability exchanges with each endpoint and determines the media format used in a conference
  - Assigns terminal numbers to each endpoint in the conference
  - Maintains a list of all conference participants
- **MP** is used for processing of audio/video/data streams in a centralized or hybrid multipoint conference



Note: - *MC/MP* may be co-located with a Gateway or Gatekeeper
- Gateway, Gatekeeper and MCU may be a single device

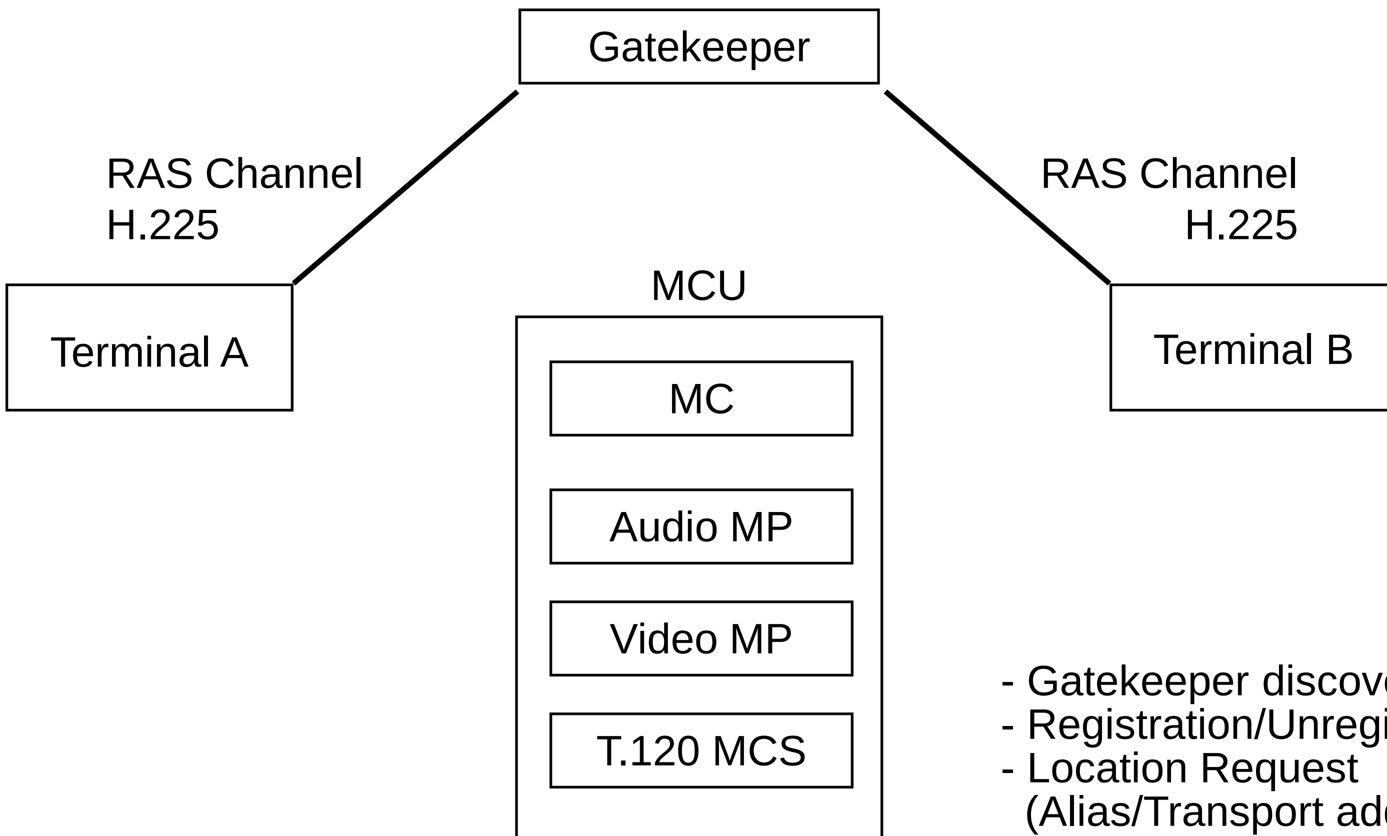


H.323 Basic Protocols for VoIP



H.323 VoIP Call Setup Procedures (1)

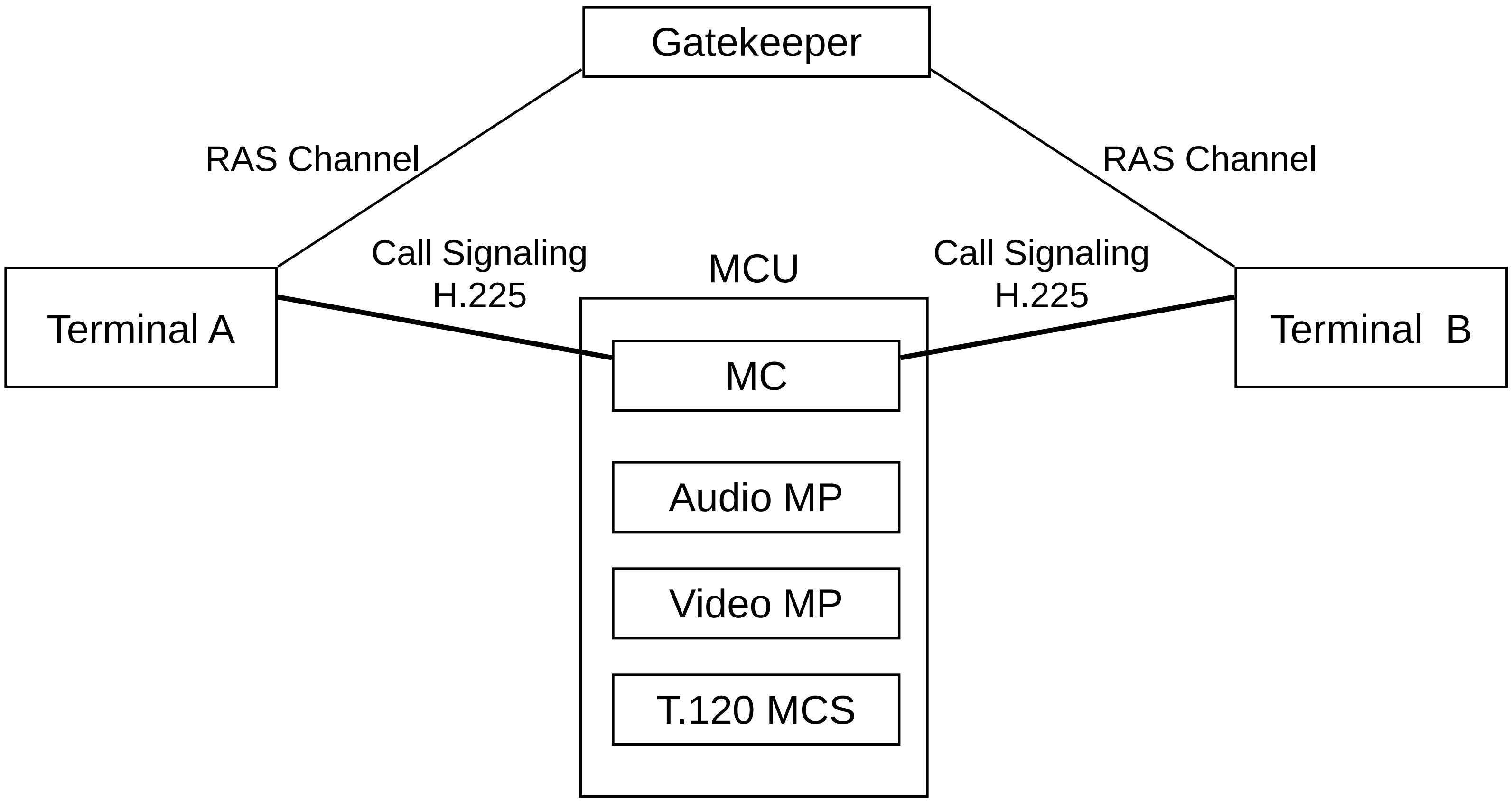
- Step 1: Endpoint - Gatekeeper communication



- Gatekeeper discover
- Registration/Unregistration
- Location Request
  (Alias/Transport address lookup)
- Admission control
- Bandwidth changes
- Status Request

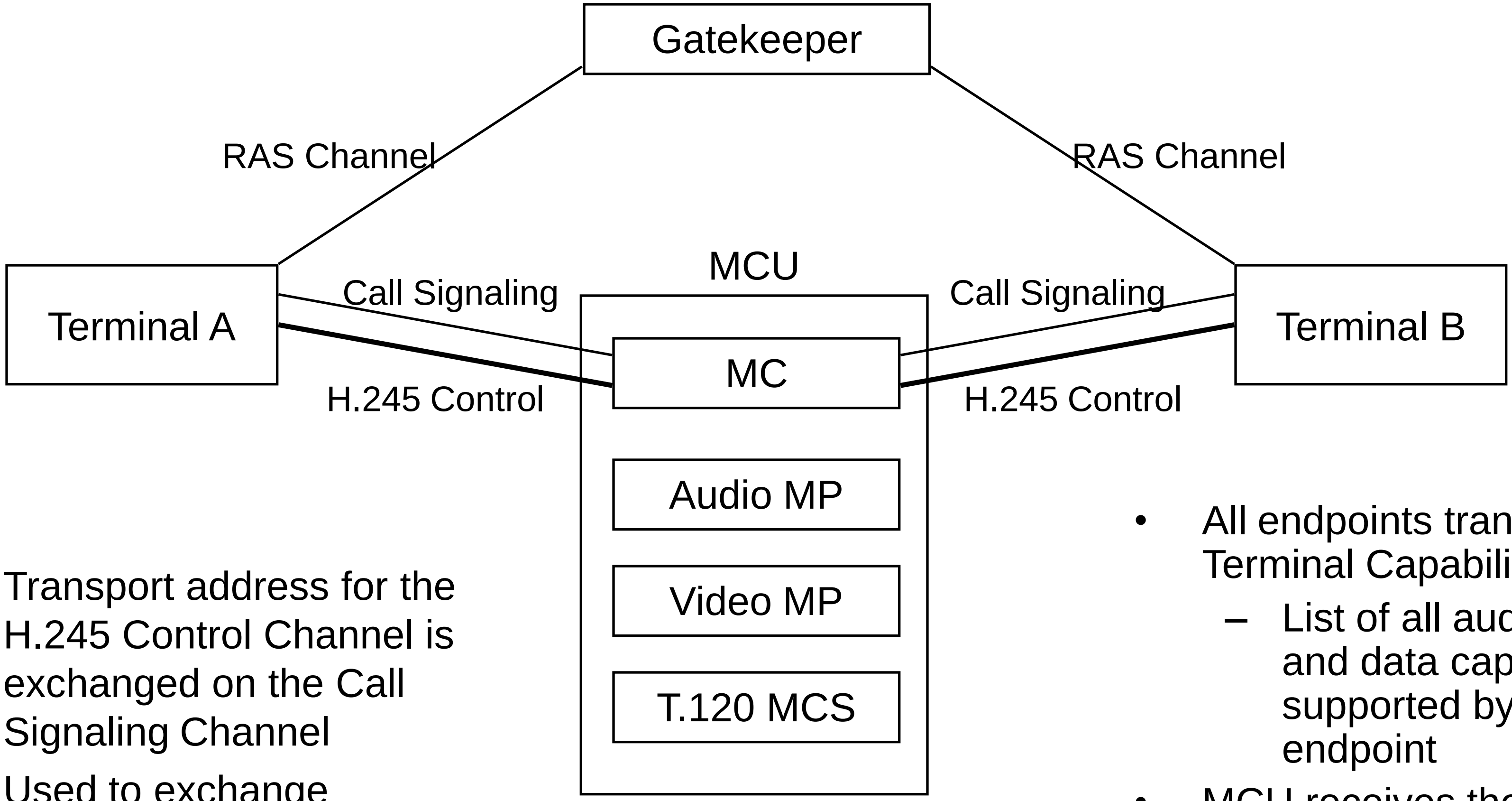
H.323 VoIP Call Setup Procedures (2)

- Step 2: Setup initial connection with the MCU using the Call Signaling Channel via gatekeeper



H.323 VoIP Call Setup Procedures (3)

- Step 3: Setup H.245 Control Channel with the MCU

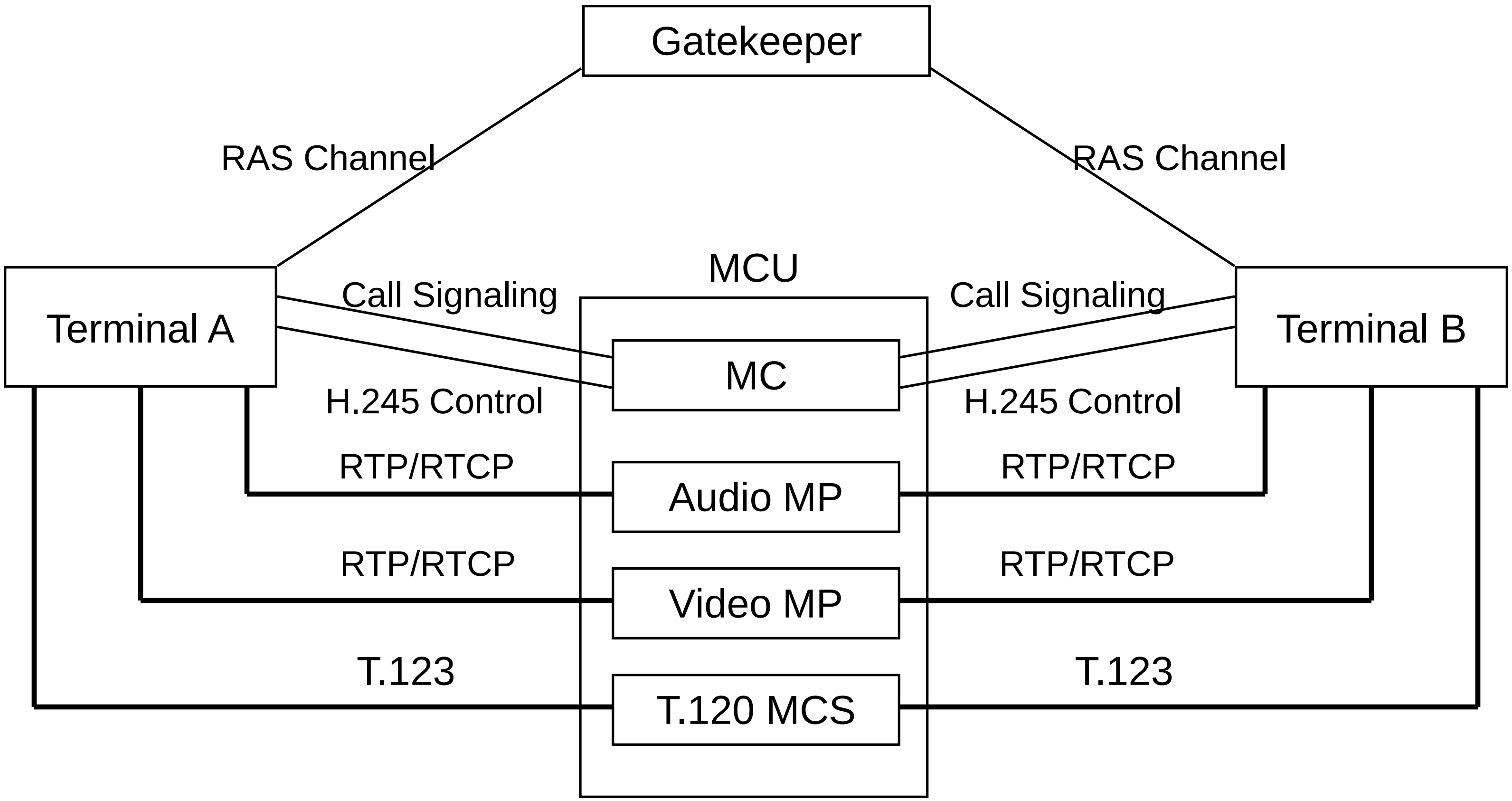


- Transport address for the H.245 Control Channel is exchanged on the Call Signaling Channel
- Used to exchange capabilities, create logical channels, and exchange multipoint commands

- All endpoints transmit a Terminal Capability Set
  - List of all audio, video, and data capabilities supported by the endpoint
- MCU receives the capabilities and determines the Selected Communication Mode (SCM)

H.323 VoIP Call Setup Procedures (4)

Step 4: Setup additional logical channels for audio/video/data



T.120 Multipoint Data Conferencing

- T.120 defines multipoint data communications standards in a multimedia conferencing environment
- Provides mechanism to identify the participating nodes and exchange information
- Enables multiple simultaneous conference handling and participation
- Consists of a set of protocols:

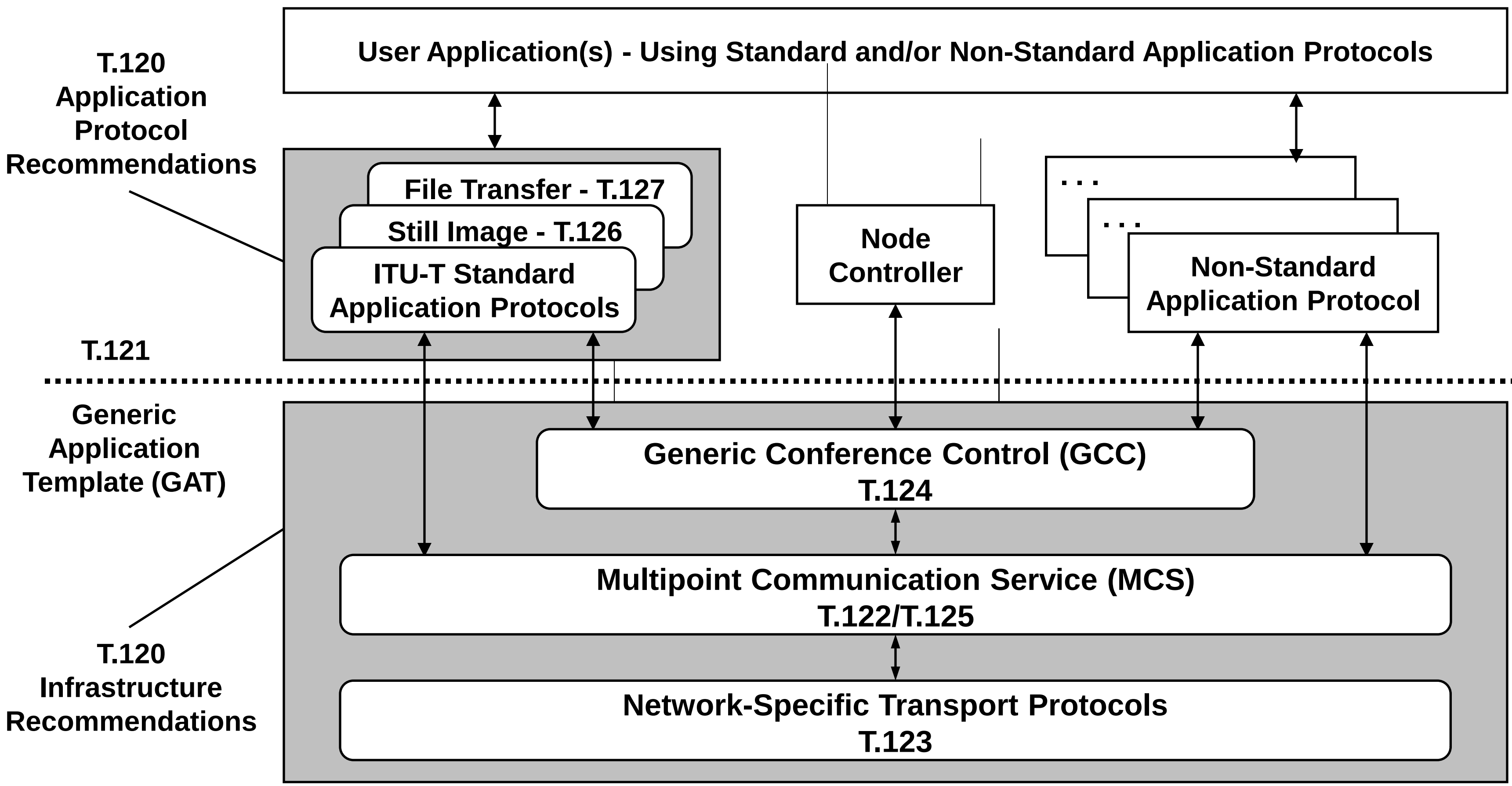
<u>Core Protocols:</u>

- ➢ T.123: Transport Protocol
- ➢ T.124: Generic Conference Control (GCC)
- ➢ T.125/T.122 Multipoint Communication Service (MCS)

<u>Optional Protocols</u>

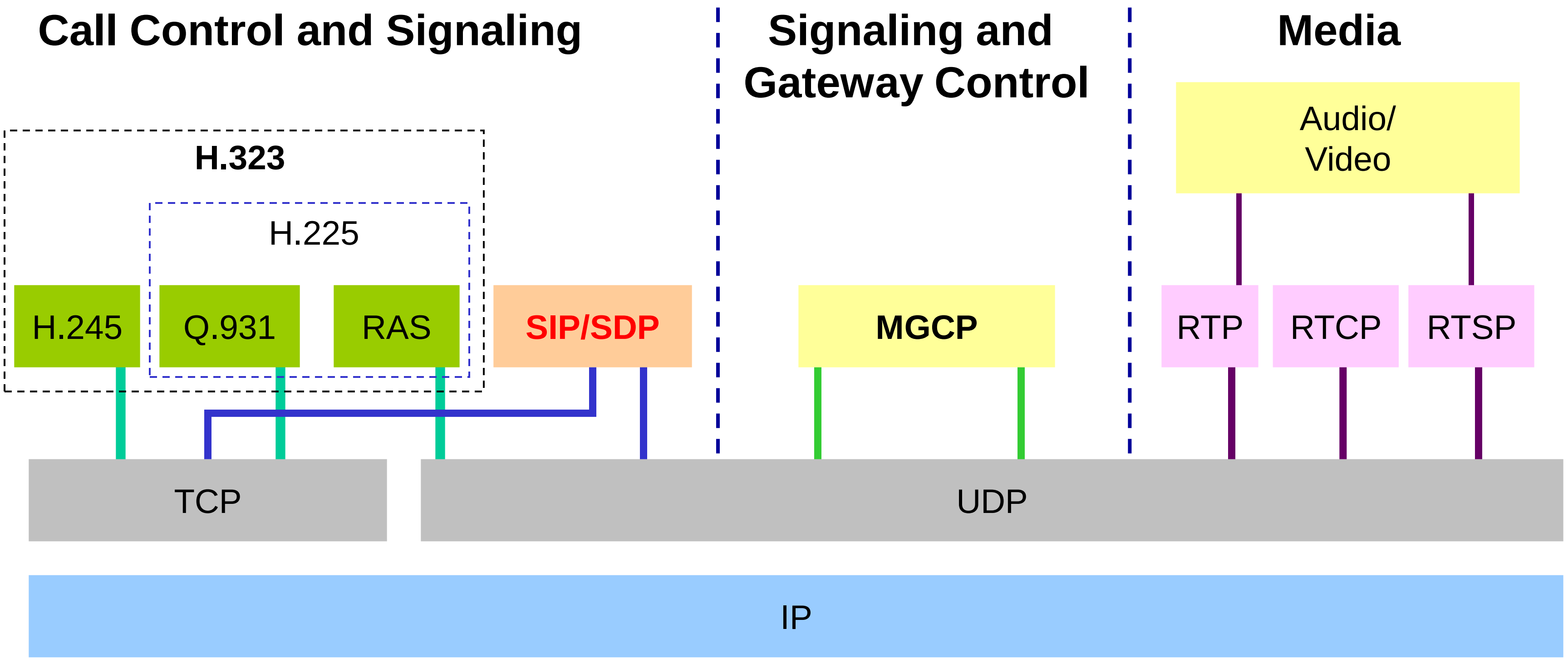
- ➢ T.121: Generic Application Template (GAT)
- ➢ T.126: MultiPoint Still Image and Annotation Protocol (NSIA)
- ➢ T.127: Multipoint Binary File Transfer Protocol (MBFT)
- ➢ T.128: Application Sharing (AS)

T.120 System Model



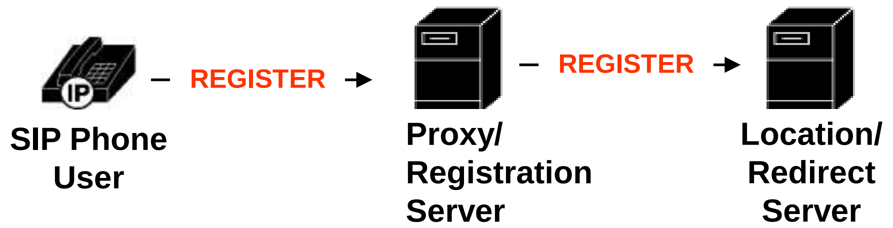
Alternative: SIP/SDP

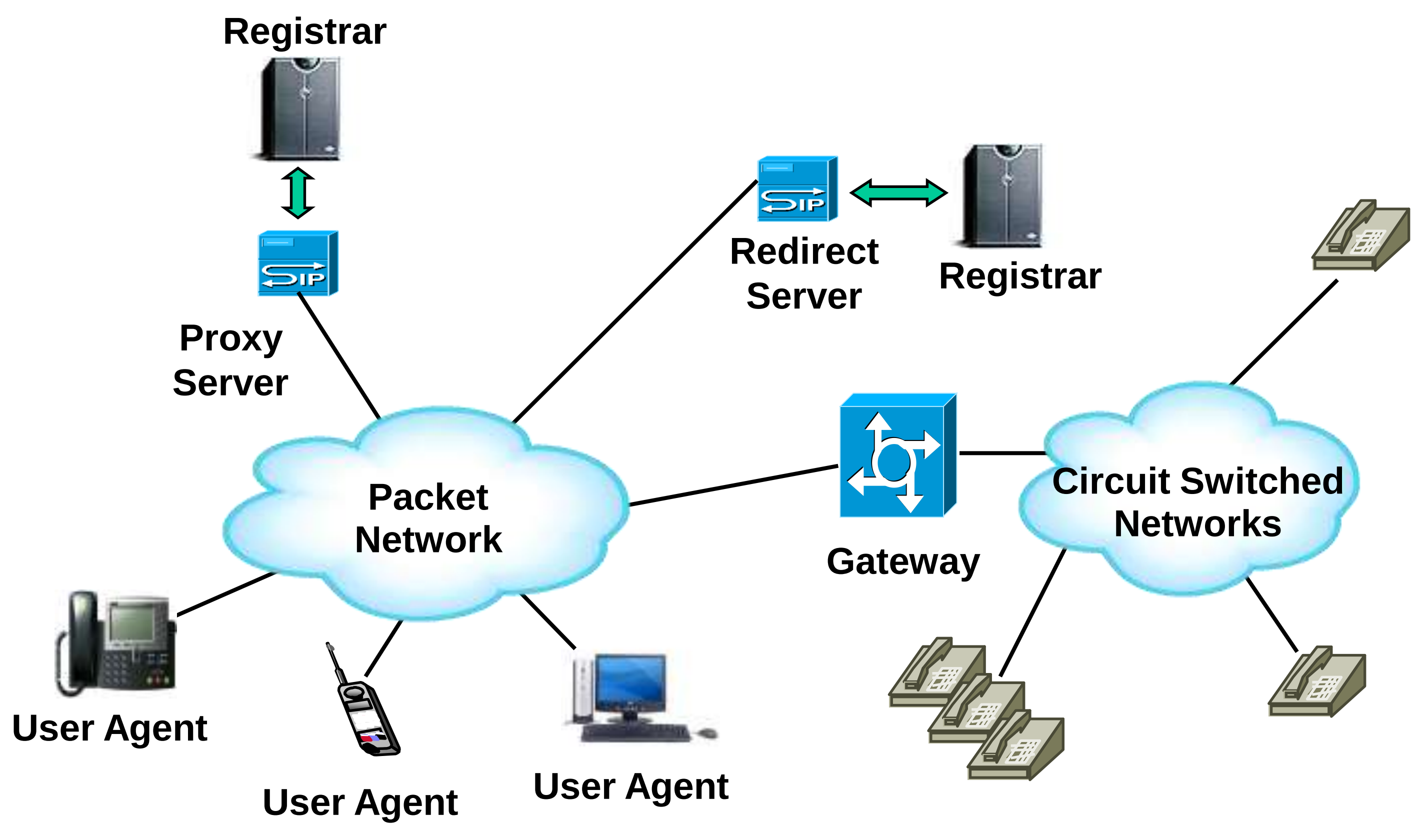
- The **Session Initiation Protocol** (**SIP**, RFC 2543) has been proposed as an alternative to H.323
- SIP is capable of negotiating a call
- SDP is used to describe capabilities: media, coding, protocol, address/port, crypto key
- Media still runs over RTP
- Each has merits and demerits, but quite similar



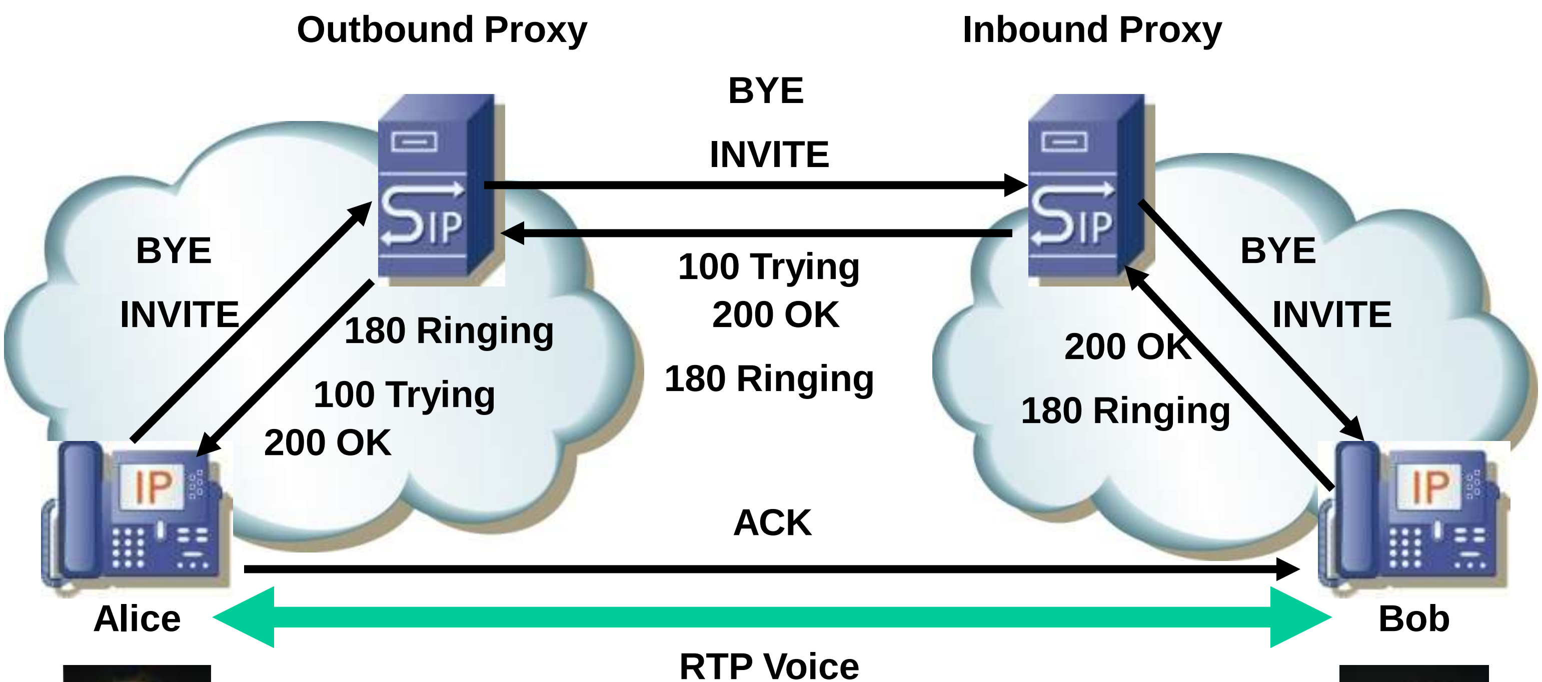
SIP Entities and Architecture

- H.323 terminal → SIP user agent
- H.232 gatekeeper
  → SIP server: proxy, registrar, redirect
- H.232 gateway → SIP gateway





SIP Call Flow



Alice Calls Bob
Is Bob there?

No. I need Bob.

Thanks. Bye.

Hello.

Steve answers Bob’s phone

Sorry, no, can I help you

SIP Detailed Call Setup and Teardown

```
INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bK776asdhds
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710@pc33.atlanta.com
CSeq: 314159 INVITE
Contact: <sip:alice@pc33.atlanta.com>
Content-Type: application/sdp
Content-Length: 142
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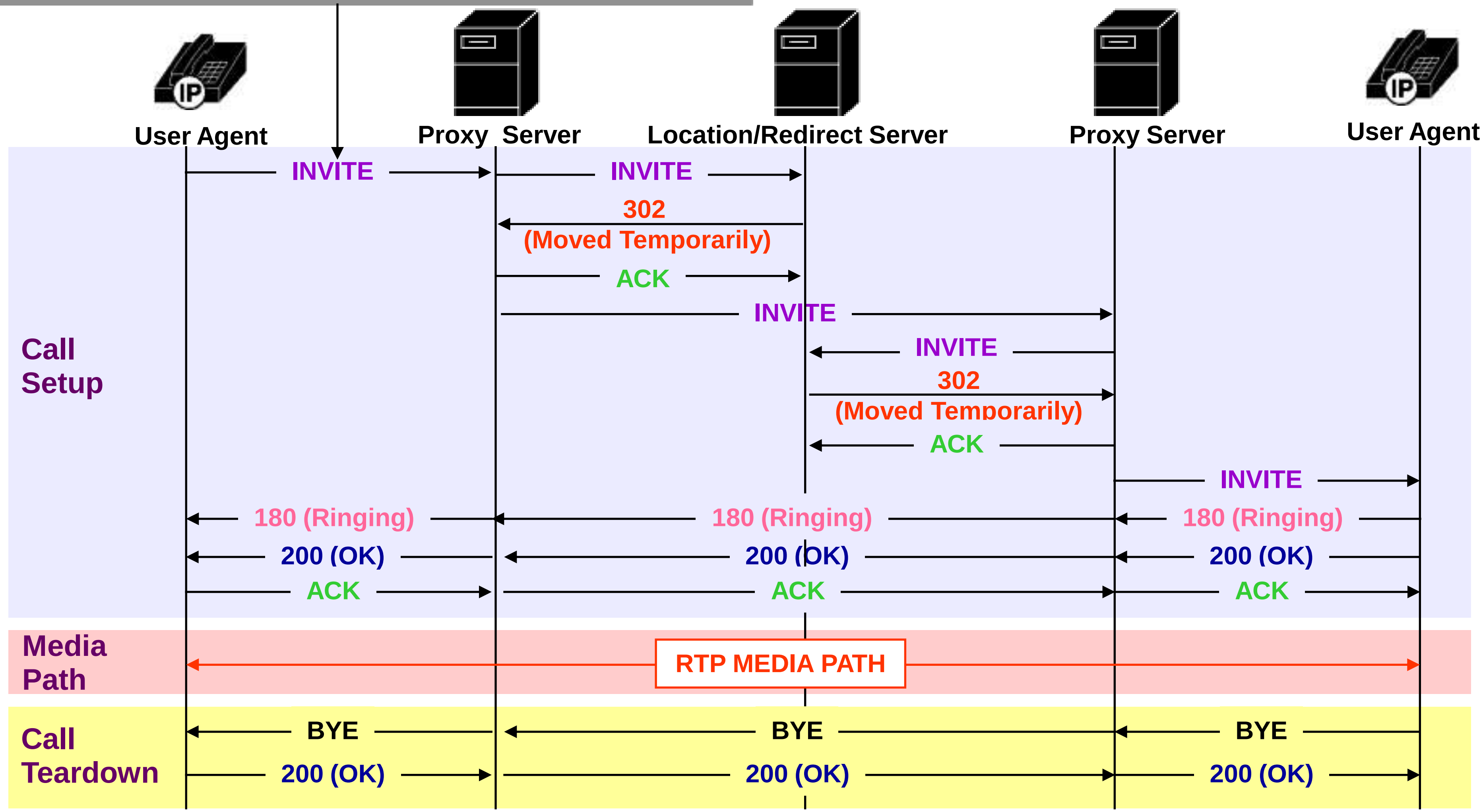
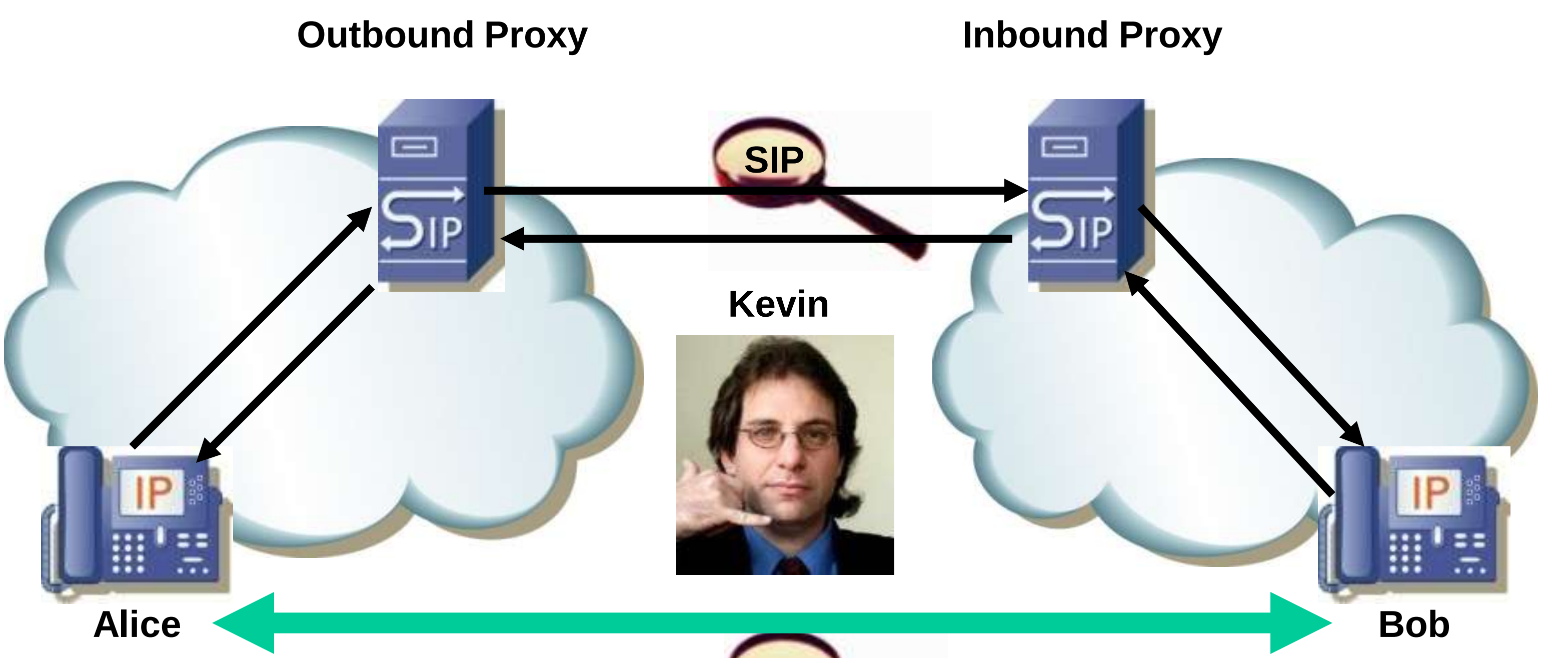


VoIP Communication Security



Yak 

 Yak

- DTMF intercept
- IM snooping
- Call pattern analysis
- Number harvesting
- Network discovery

- Voice reconstruction
- Fax reconstruction
- Video reconstruction

Demos of Skype for Phone Call and Tele-Meeting