

Voice over IP (VoIP)

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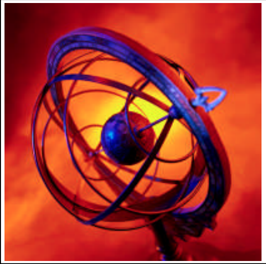
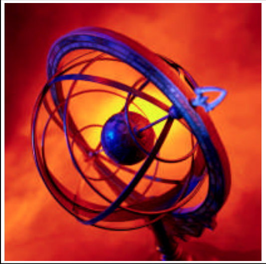


Presentation Outline

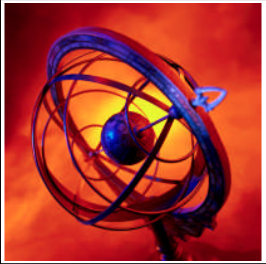
- Basic IP phone set up
- The SIP protocol



Learning Objectives

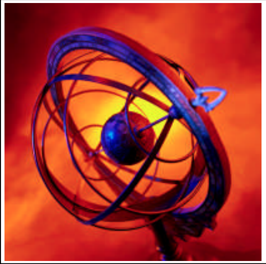
You will be able to:

- To understand the current state of the art in Internet Telephony.
- To be able to describe the key protocols involved in VoIP and their roles in this new technology.
- To be able to discuss the limitations of current VoIP technology.
- Identify key factors influencing the growth of Internet telephony services for global communications.



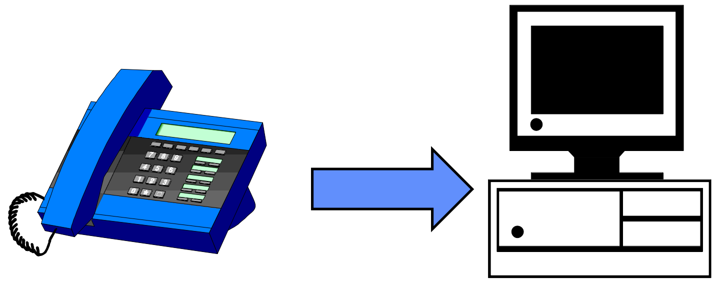
References

- Forouzan, “Data Communications and Networking”, 4th Edition
- Tanenbaum, “Computer Networks”, 4th Edition
- Cisco CCNA1 Module 10 - part 1
- Stallings, William 2000 ‘Data and Computer Communications’, Prentice Hall, Sixth Edition
- Russell, Travis 1997 ‘Telecommunications Protocols’, McGraw Hill



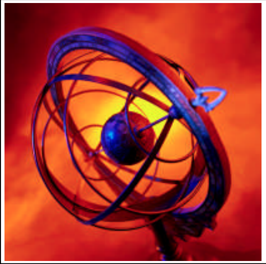
Introduction

- Many companies have seen advantages in minimising costs by transporting voice over IP networks.
- This has set the stage for standards development and the design of terminals and gateways and the rolling out of services on a global scale.
- Adding voice to packet networks generates many challenges:
  - interoperability
  - packet loss
  - delay
  - scalability
  - reliability
  - quality



Examples of Possible VoIP Applications

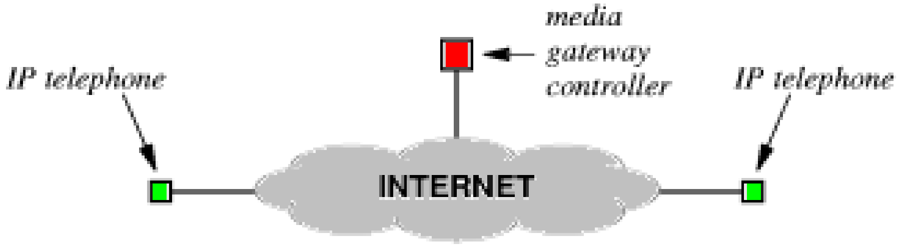
- **PSTN Gateways**
  - PC based telephone accessing a public network by calling a gateway at a point close to the destination to minimise long distance charges.
- **Internet aware telephones**
  - Enhancement of ordinary telephones to serve as an Internet access device as well as ordinary telephony. Directory services could be accomplished via the Internet.
- **Tie line replacement**
  - Intranet links could replace tie lines between company PBXs
- **Remote access from branch or home**
  - Small office could gain access to corporate voice, data and fax.
- **Voice calls from a mobile PC via the Internet**
- **Internet call centres**



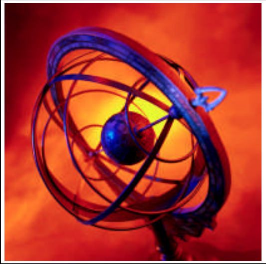
More on Challenges

- Voice quality has to be comparable to PSTN
- Underlying network must meet strict performance criteria including:
  - minimising call rejections
  - network latency
  - packet loss
  - disconnects
- Call control (signalling) must make the telephone calling process transparent so that the callers need not know the technology involved.
- PSTN/VoIP service interworking.
- System management and security and accounting and consolidated with PSTN OSSs

A Basic IP Telephone System



- **The simplest IP telephone system uses two basic components:**
  - ***IP telephone*: end device allowing humans to place and receive calls.**
  - ***Media Gateway Controller*: providing overall control and coordination between IP phones; allowing a caller to locate a callee (e.g. call forwarding)**



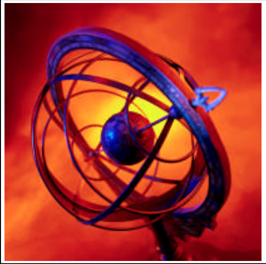
Voice over IP protocol families

- There are currently four families of VoIP protocols:
  - SIP,
  - H.323 and
  - H.248 based, and
  - SIGTRAN.
- Call Control
  - The call control protocols establish, modify and release connections.

Can we use TCP?

Note

TCP, with all its sophistication, is not suitable for interactive multimedia traffic because we cannot allow retransmission of packets.



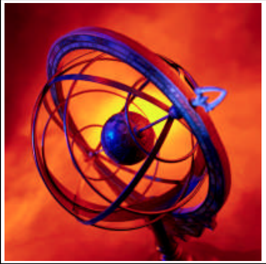
What about UDP?

Note

UDP is more suitable than TCP for interactive traffic. However, we need the services of RTP, another transport layer protocol, to make up for the deficiencies of UDP.

SIP family

- **Based on IETF specifications:**
  - **SIP: "Session Initiation Protocol" used for call control.**
  - **SDP: "Session Description Protocol" used to describe the "media" session inside of SIP and other protocols.**
  - **RTSP: "Real Time Streaming Protocol" used to set up streaming sessions.**
  - **Sigcomp: "Signalling Compression " a solution for compressing messages generated by application protocols such as SIP and RTSP.**



H.323 family

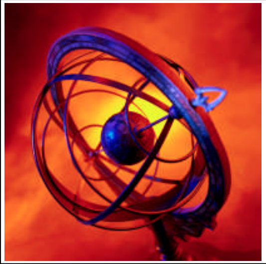
- **Based on ITU-T specifications:**
  - **H323: "Packet-based multimedia communications systems"**
  - **H225: “Call control protocol"**
  - **H235: "Security"**
  - **H245: "Media control protocol"**
  - **H450: "Supplementary Services"**
  - **Q.931: "ISDN user-network interface layer 3 specification for basic call control"**
  - **H223: Multiplexing protocol for circuit-based multimedia communications system**

H.248 family

- **Based on IETF specifications:**
  - **H248/MEGACO: "Media Gateway Control Protocol" H248.1 RFC3015(Version 1)**
  - **MGCP: "Media Gateway Control Protocol" RFC3435**

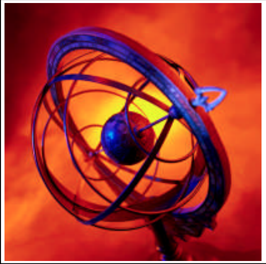
SIGTRAN family

- SIGTRAN is often used in connections towards PSTN.
- Proprietary protocols
  - Cisco proprietary protocols:
    - SKINNY: Terminal control protocol.
  - Digium proprietary protocols:
    - IAX2: A standalone VoIP protocol primarily developed for communication between Asterisk servers.
  - Nortel proprietary protocols:
    - UNISTIM: A stimulus protocol for IP phones.



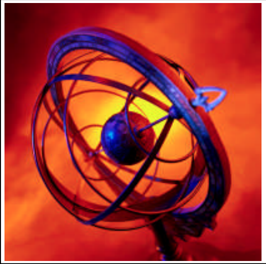
SIP: Session Initiation Protocol

- **Invented by the IETF.**
- **SIP defines three main elements that comprise a signalling system:**
  - **User Agent:** **IP phone or applications**
  - **Location servers:** **store information about a user's location or IP address**
  - **Support servers:**
    - *Proxy Server*: **forwards requests from user agents to another location.**
    - *Redirect Server*: **provides an alternate called party's location for the user agent to contact.**
    - *Registrar Server*: **receives user's registration requests and updates the database that location server consults.**



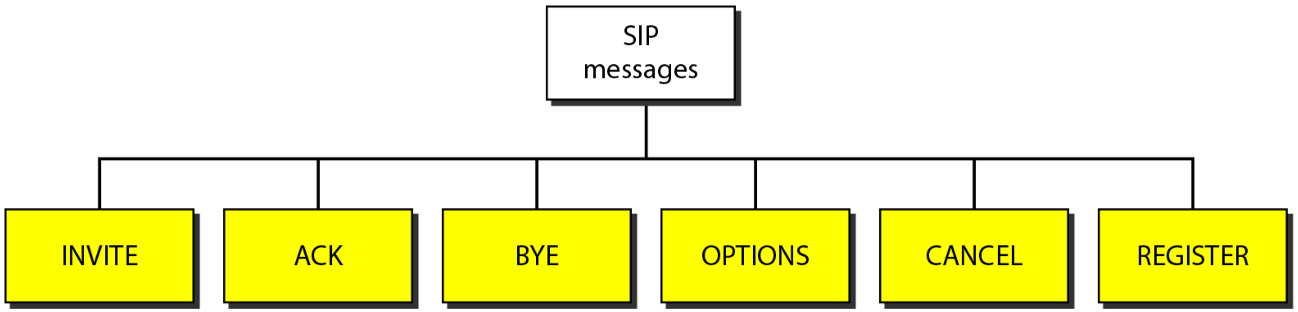
SIP Characteristics

- Operates at the Application Layer.
- Encompasses all aspects of signalling, e.g. location of called party, ringing a phone, accepting a call, and terminating a call.
- Provides services such as call forwarding.
- Relies on multicast for conference calls.
- Allows two sides to negotiate capabilities and choose the media and parameters to be used.
- SIP URI is similar to an email address. (With prefix "sip:") E.g. sip:bob@somewhere.com

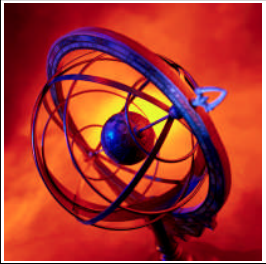


SIP Methods

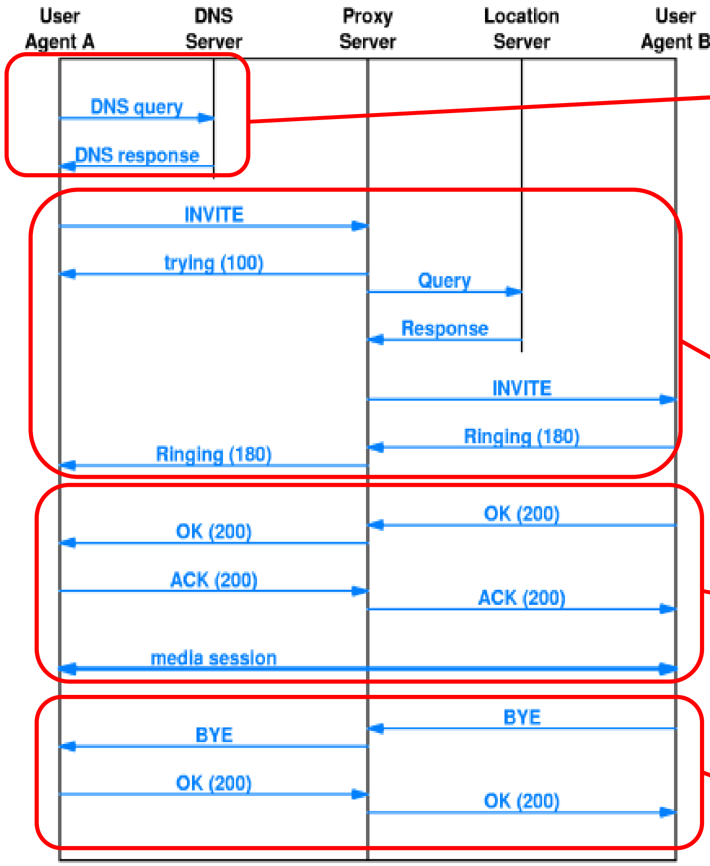
- **Six basic message types, known as *methods:***



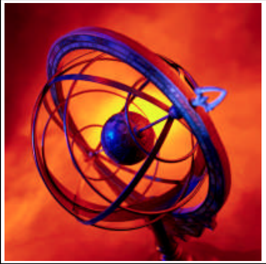
Method	Purpose
INVITE	Session creation: an endpoint is invited to participate in the session.
ACK	Acknowledgment response to INVITE.
BYE	Session termination: call is ended.
CANCEL	Pending request cancellation (no effect if request has been completed).
REGISTER	Registration of user's location (i.e., a URL at which the user can be reached).
OPTIONS	Query to determine capabilities of called party.



An Example SIP Session



- User agent A contacts DNS server to map domain name in SIP request to IP address.
- User agent A sends a INVITE message to proxy server that uses location server to find the location of user agent B.
- Call is established between A and B. Then media session begins.
- Finally, B terminates the call by sending a BYE request.

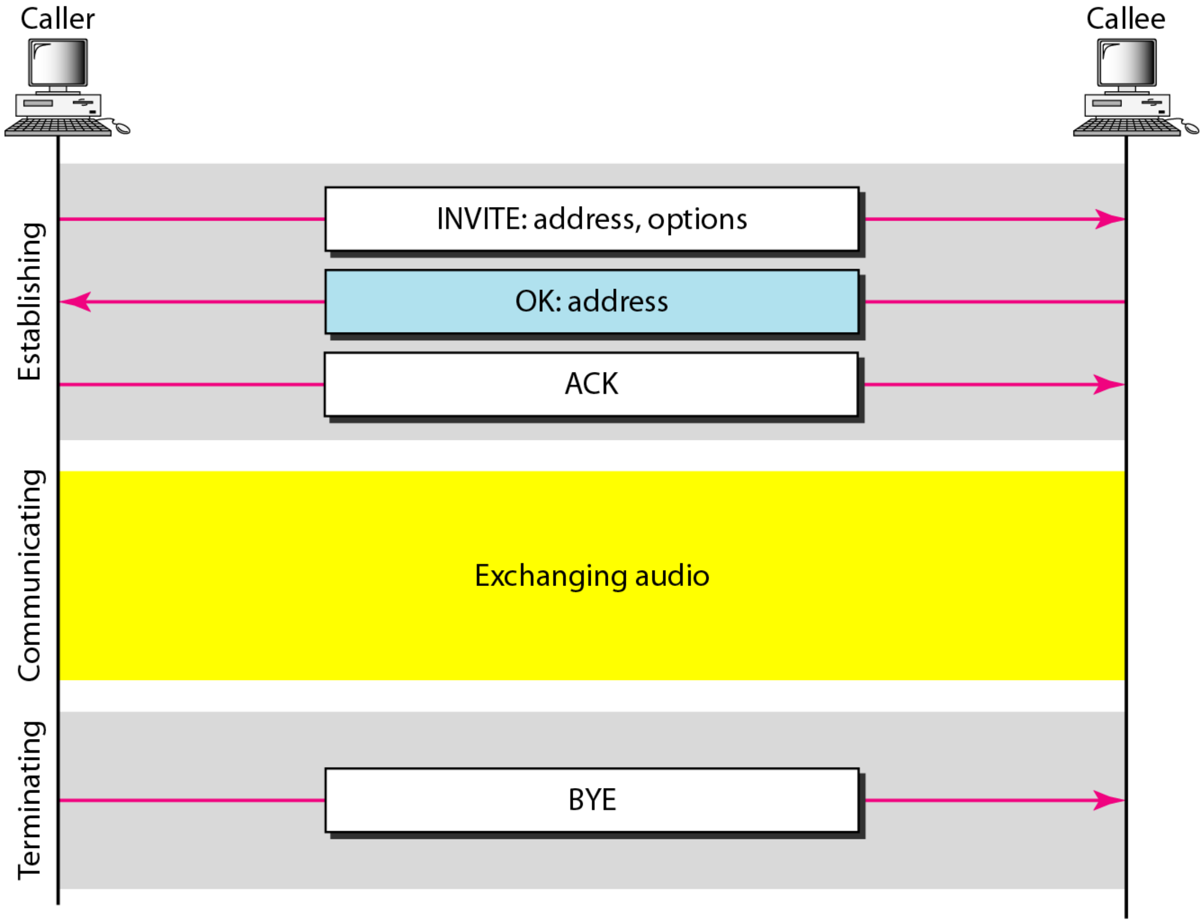


SIP Message Formats

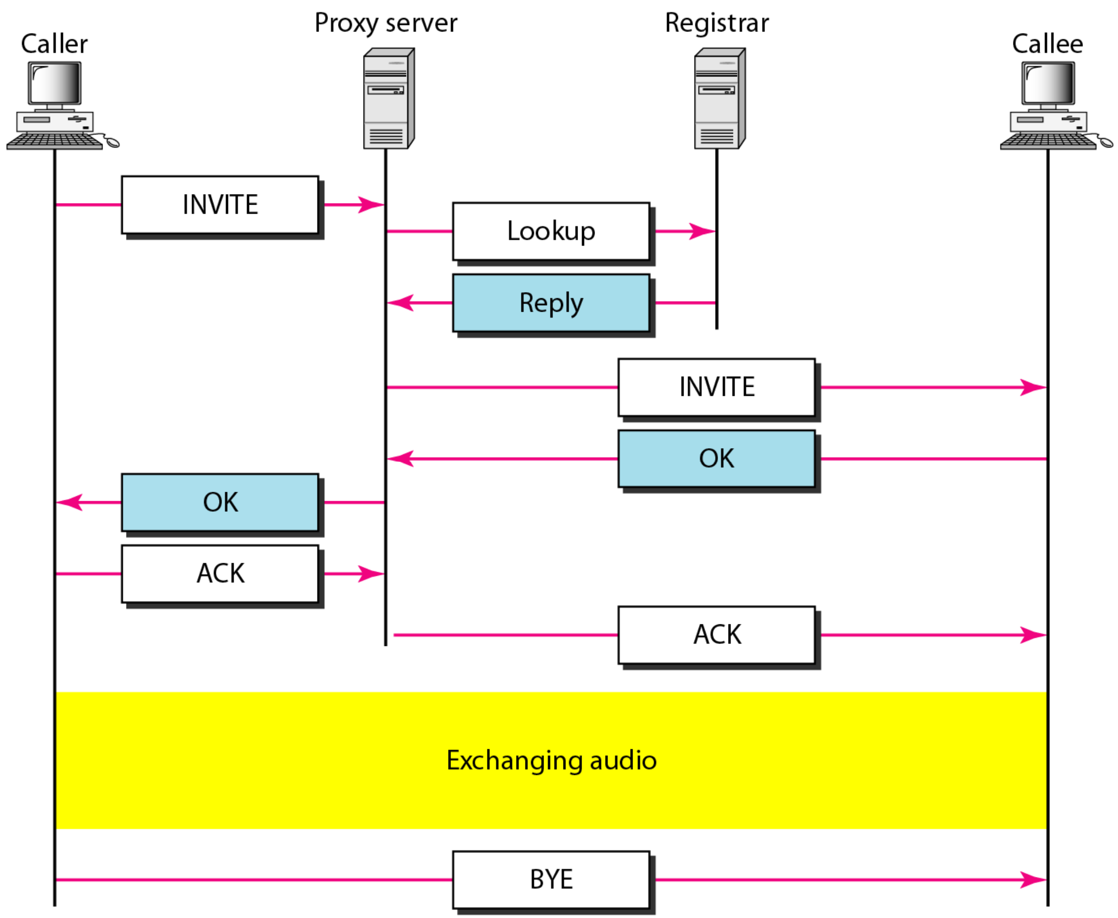
- **There are three basic SIP message formats, viz:**

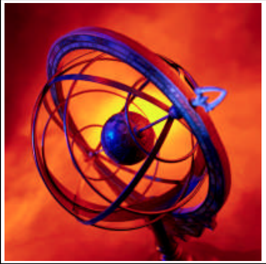
sip:bob@201.23.45.78	sip:bob@fhda.edu	sip:bob@408-864-8900
IPv4 address	E-mail address	Phone number

Simple SIP session



Tracking the Callee





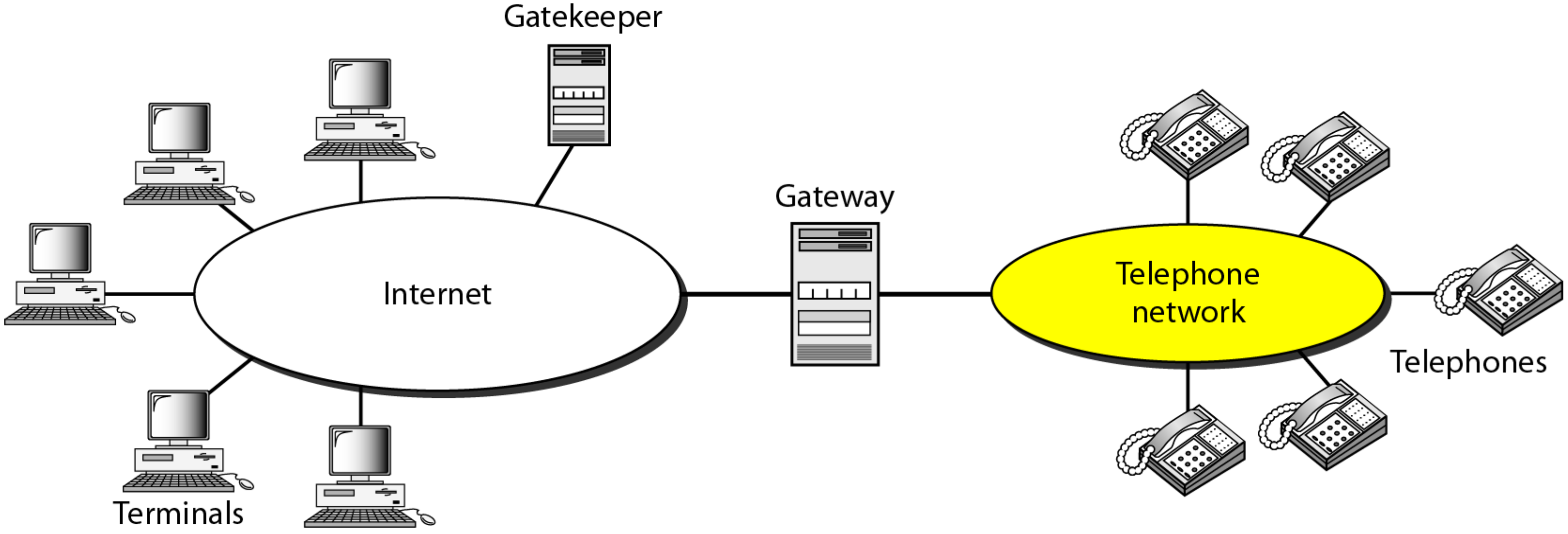
H323 Protocol - 1

- **Reference: ITU-T H.323**
- **Title: Packet-based Multimedia Communications Systems**
- **Conceived originally for multimedia conferencing on a LAN, but now extended to cover Internet Telephony.** (Revised in 1998)
- **Provides:**
  - **Call control**
  - **Conferencing functions**
  - **Call management**
  - **Capability negotiation**
  - **Supplementary services**

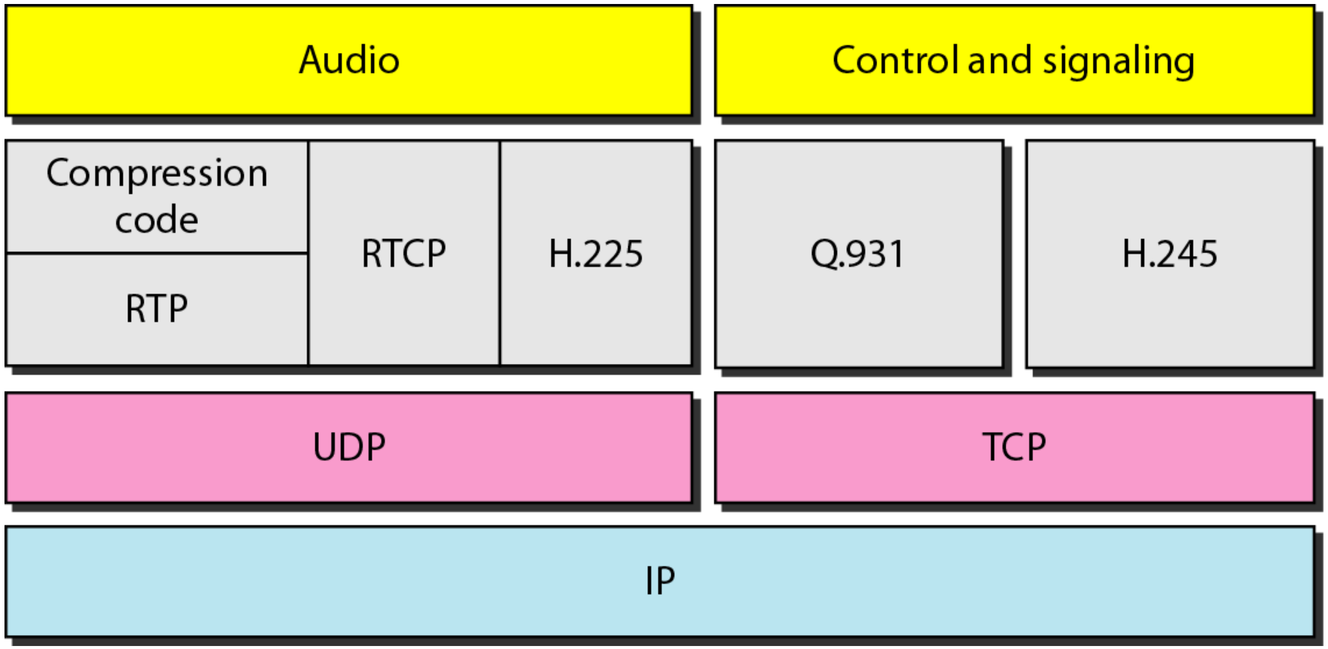
H.323 Protocol - 2

- ***This Recommendation describes terminals and other entities that provide multimedia communications services over Packet Based Networks (PBN) which may not provide a guaranteed Quality of Service. H.323 entities may provide real-time audio, video and/or data communications.***
- ***Support for audio is mandatory, while data and video are optional, but if supported, the ability to use a specified common mode of operation is required, so that all terminals supporting that media type can inter-work.***
- ***The packet based network over which H.323 entities communicate may be a point-to-point connection, a single network segment, or an internetwork having multiple segments with complex topologies.***

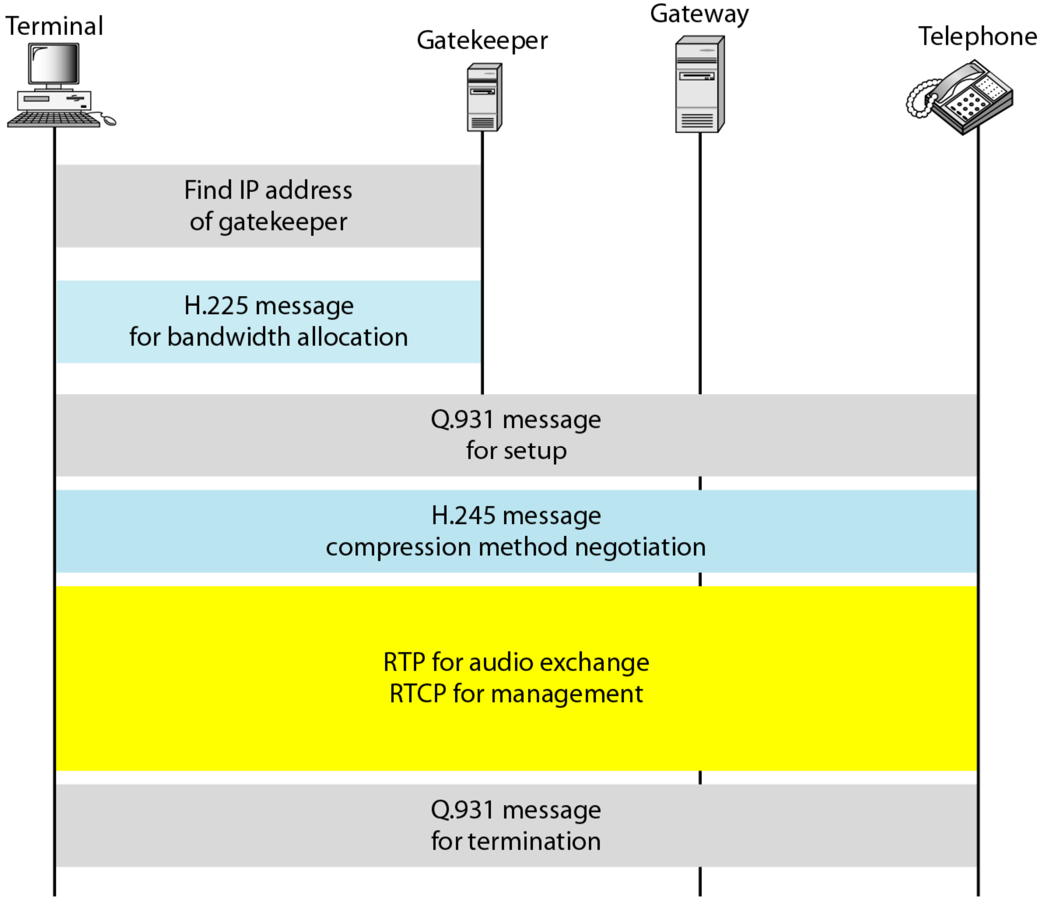
H.323 architecture



The H.323 protocols

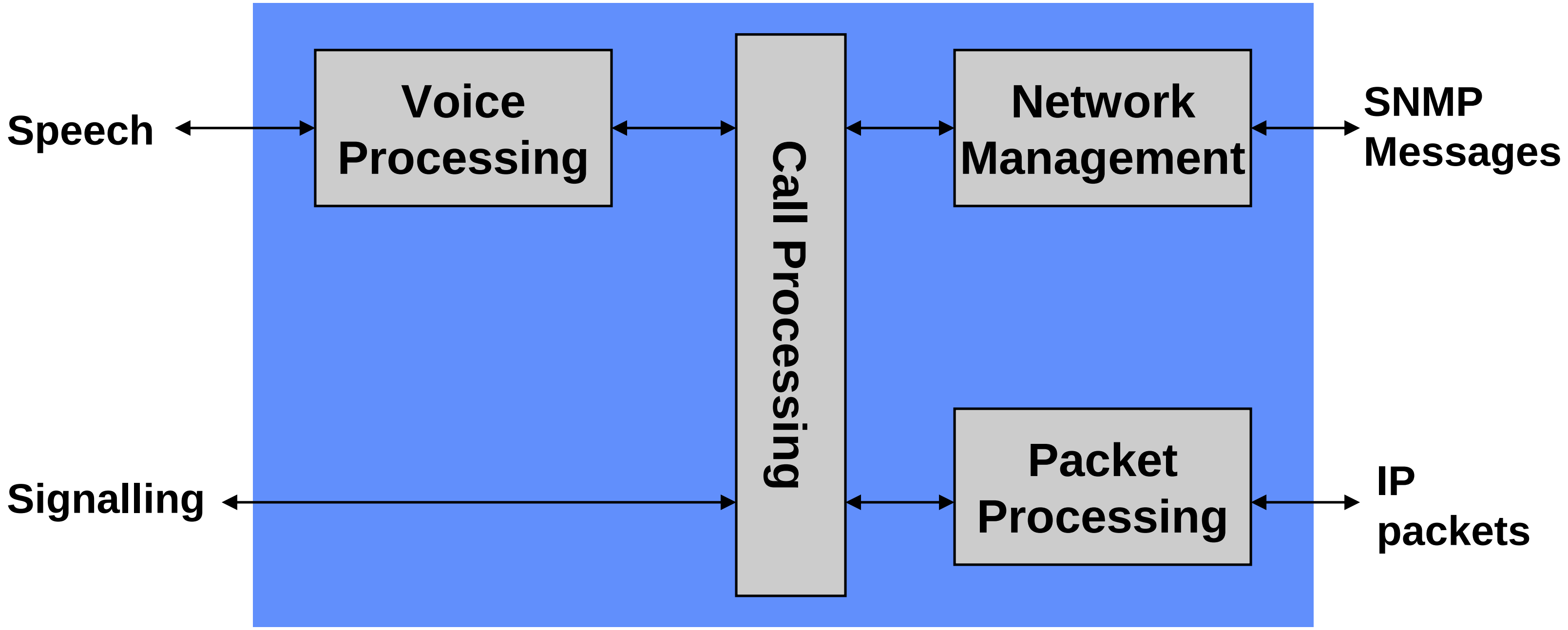


H.323 Example



Voice Gateway/Terminal Functions

- **The following picture shows the functional components of terminals that use the H.323 standards:**



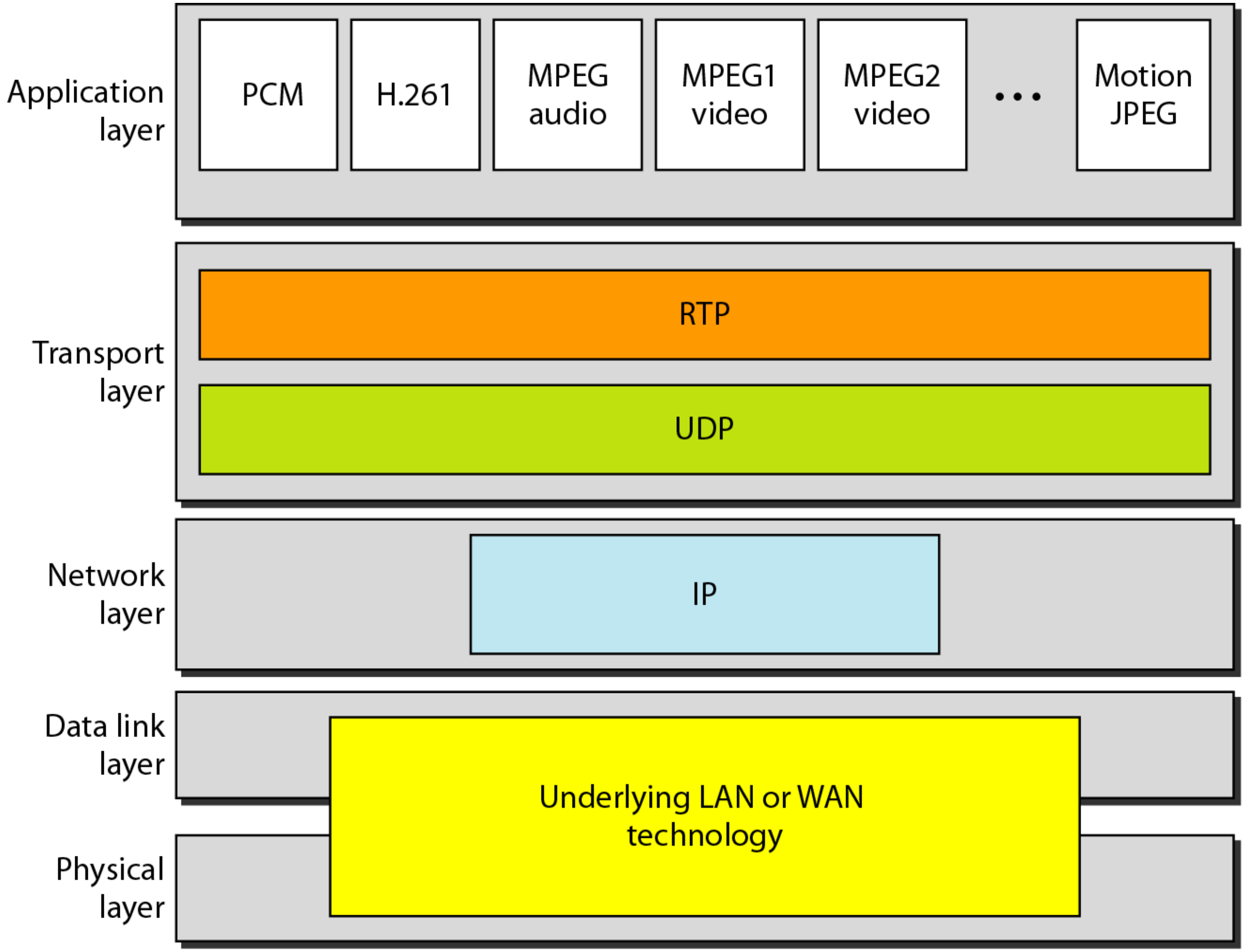
Transport

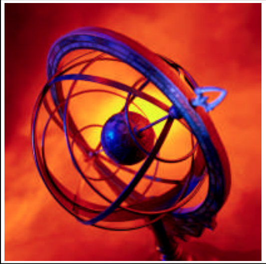
Both SIP and H323 use the same RTP/RTCP transport protocols:

- RTP: "RTP: A Transport Protocol for Real-Time Applications" RFC3550
- RTCP: "Real-time Control Protocol", used together with RTP
- T38: "Real-time facsimile (T.38), Fax-over-IP"

- Movement toward secure communications involve these protocols
  - SRTP: "The Secure Real-time Transport Protocol (SRTP)" RFC3711
  - MIKEY: "MIKEY: Multimedia Internet KEYing" RFC3830

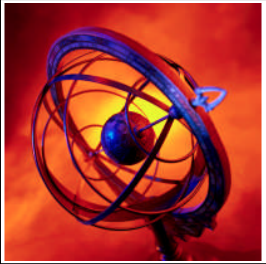
RTP - Real-time Transport Protocol - 1





RTP - Real-time Transport Protocol - 2

- Reference: RFC 1889
- A real time end to end protocol
  - Utilises existing transport layers for data that has real time properties.
  - Used by H.323
  - Takes the bitstream generated by the media encoder breaking it into packets, sending the packets over the network and recovering the bitstream at the receiver.
  - Plays a key role in Internet telephony since it is the component that moves the actual voice among the participants.
  - Signalling protocols provide the parameters for RTP transport.

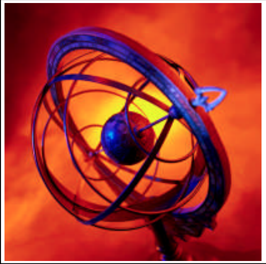


RTP - Real-time Transport Protocol - 3

- **Specific functions provided by RTP are:**
  - **Sequencing**
    - **Each RTP pack has a sequence number used for loss detection and compensation for reordering.**
  - **Intramedia synchronisation**
    - **Packets within the same stream can suffer different delays (jitter). Applications use playout buffers to compensate for this jitter. RTP provides timestamps to assist in this….**
  - **Payload Identification**
    - **Since network conditions may vary during a call, it may be necessary to change encoding dynamically. RTP contains a payload type identifier in each packet.**
  - **Frame Indication**
    - **Video and audio are sent in logical units called frames. It is used to mark B of Frame and E of Frame for upper layers.**

RTP - Real-time Transport Protocol - 4

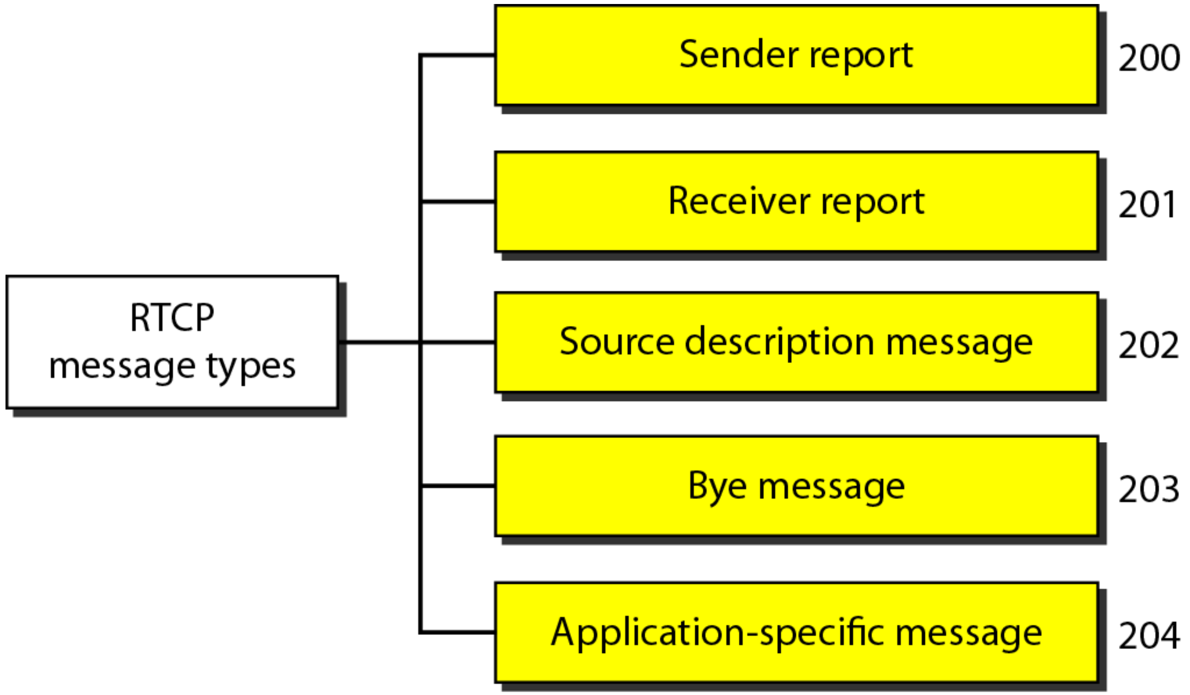
- **Source Identification**
  - In a multicast session, many users are participating and so there has to be a mechanism for a packet to say which participant actually sent it. A special identifier called a SSRC - Synchronisation Source is included in the protocol.
- The RTP protocol has a companion protocol called the RTCP.
  - RTCP = Real Time Control Protocol



RTCP - Real Time Control Protocol

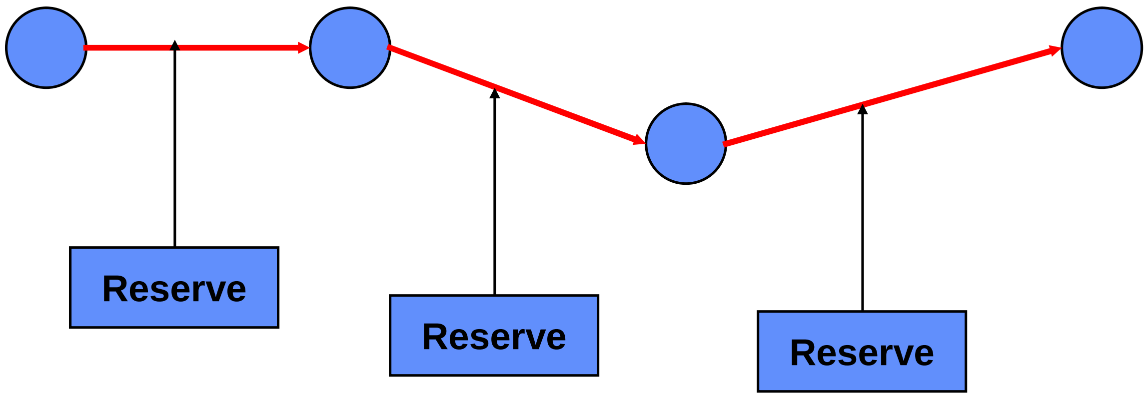
- **Facilities provided by the RTCP:**
  - **QoS feedback**
    - **Receivers in a session report back the quality of their reception from each sender.**
    - **Feedback includes**
      - **Lost packets, jitter, round trip delays**
  - **Intermedia Synchronisation**
    - **Audio and video are often carried as separate streams but need to be synchronised at the receiver (eg “lip sync”)**
  - **Identification**
    - **RTCP packets contain full details of email, phone number and name of participant - this is available to other participants.**
  - **Session Control**
    - **Allows you to send small notes or say “goodbye”!!**

RTCP Message Types



RSVP Protocol

- Reference: RFC 2205 - 2209
- General purpose signalling protocol that allows network resources to be reserved for a connectionless data stream, based on receiver controlled requests.



Authentication, Authorisation and Accounting

- **RADIUS: "Remote Authentication Dial In User Service" RFC2865**
- **DIAMETER: "Diameter Base Protocol" (successor of RADIUS)**