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Multimedia in IP

General Concepts

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Learning outcomes

- Understand the scope of VoIP models
- Describe RTP operation
- Understand the SIP and H.323 protocols
- Describe architectures for interconnecting POTS and the Internet.

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Concepts and Protocols
Voice over IP

More than multimedia streaming…

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IP dominance in communications

Circuit switched systems

- Products based on TDM still give the major profits in industry
- They are getting close to the cost and efficiency limits
- Obsolets…
- Conversation (voice services) is critical

Packet switched systems

- Services based on IP will be dominant (SIP, VoiceXML)
- New distributed characteristics between gateways and media servers
- Conversation (voice services) is still a critical aspect

Migration is evolutionary: there is ***interoperability***

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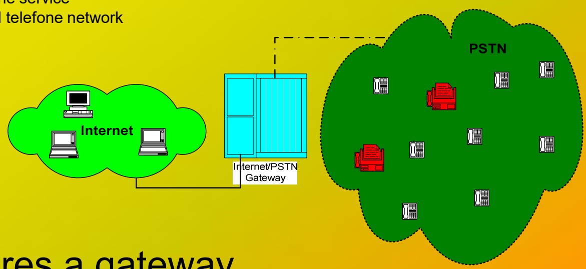
Multimedia in IP: a substitute for POTS? Interoperability!

POTS – Plain old telefone service
PSTN – Public switched telefone network



- Requires a gateway
  - What type of service translation?
  - What facilities are being provided?
  - What facilities to provide?
  - What can be provided (and how), taking into account the telecommunications environment?
- Currently VoIP is the default in many enterprises
  - And still connect to the phone network

POTS interoperation

- Interoperation of PSTN services with data networks (e.g., the establishment of a voice call between a phone based on internet and a traditional phone)
- Interoperation of data services with PSTN networks (e.g., paging/calling a user after an email reception)
- New services simultaneously based in PSTN and Internet facilities (eg, WEB-based helpdesk, capable of sending documentation through fax)

Although IP networks dominates now, there were many years of joint co-existence of both networks, which becomes a legacy

Multimedia in IP

- Many algorithms/applications supporting voice/video above IP
  - Vivo, ShockWave, AAC, MPEG-4, H.323, H264, RealAudio, etc…
- As long as some QoS exists in the network, an explosion of these applications is ever expected
  - Even without explicit QoS, multimedia took over the Internet
  - End points coding became much more adaptive
- IETF centered transport standardization:
  - Specially focused in the control of teleconference sessions and network protocols
    - Cooperates now with ITU-T
  - Has complete proposals to all audio/video communication aspects

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Data plane and control plane

- Data plane: determines data packet behavior
  - Packet forwarding (e.g. inside a router)
  - Packet differentiation (e.g., ACLs)
  - Link scheduling
  - Multimedia **transport** (e.g. the codec)
- Control plane: controls the state of network elements
  - Route selection (e.g. routing protocols)
  - RSVP, capability signaling, etc.
  - Multimedia **signaling** (e.g. the ringing tone)

In advanced architectures, these two planes often impact different functional units (boxes)

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Data+control

- Multimedia is associated to the notion of "session"
  - Requires both data (multimedia) and control information
    - E.g. voice is data, and #busy signal" is control
- In-band signaling
  - Sending of metadata and/or control information in the same "channel" than the data
- Out-band signaling
  - There is a dedicated "channel" created for the transmission of metadata and/or control

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What is signaling?

- Signalling is the process of interaction between network nodes to process calls
  - Signalling is for call **control**
  - Origin and destination nodes have to agree on the call establishment and its parameters
  - Network nodes have to prepare their resources/links for the calls – have to obtain information of the call initiation and its parameters
  - Servers for charging
- SS7 is the signalling system used in PSTN
  - There are others, and are being used… (ISDN)
- For PSTN, ISDN and SS7 are the more advanced systems
- Signalling also has to exist in the data world….
  - SIP, Megaco, H.323, ATM UNI, etc.

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SS7: System Signalling #7

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What is VoIP?

- VoIP is not a protocol!
  - VoIP is a set of protocols and equipments that allow coding, transport and routing of audio calls (multimedia) through IP networks
    - Both data (media) and signaling have to be tackled
    - Audio streams are coded in digital environment and encapsulated in IP for transport in the network.
- Examples of VoIP inclusion (required interoperation)
  - PSTN → VoIP → PSTN
  - VoIP Native → PSTN
  - VoIP Native → VoIP Native

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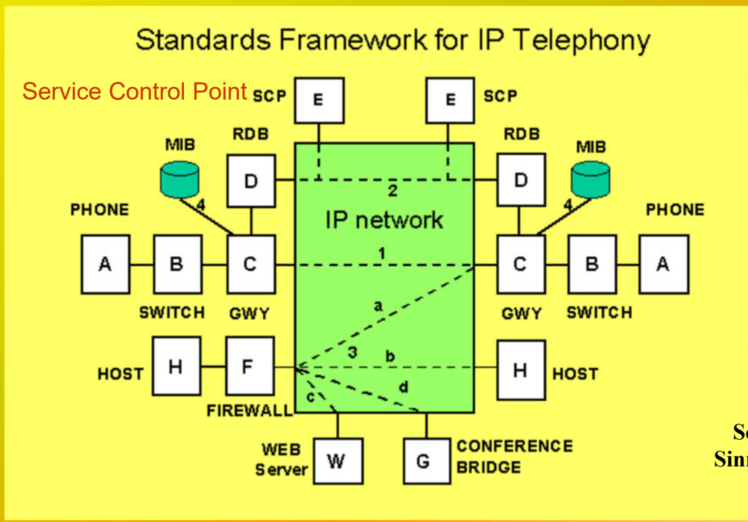
VoIP advantages

- Cost reduction
  - Do not need to pay for PSTN circuits for call transport (user side) / consolidate infrastructure (provider side)
  - Bandwidth reduction
    - Distributed nature of VoIP
    - Operation costs reduction – voice and data traffic both in the same network
- 'Open' standards and interoperability between operators
  - Does not depend on proprietary solutions
- Integration of voice and data networks
  - Considered as 'just another IP application'
  - Two major approaches: ITU-T (early on) and IETF (current)
  - As long as the quality is similar to the PSTN network, companies can easily invest in new services and applications

Voice over IP Framework



Source: Sinnreich et al

Most challenges are associated with control plane.

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Different levels of VoIP problem

1. The transport level
   - How to transport multimedia information. Covers also content, but we mostly talk about RTP (and associated protocols)
2. The session control
   - How to signal a VoIP session. Covers also application protocols, but we talk mostly about SIP and H.323
3. The gateway control
   - How to signal interface entities between Internet and POTS. We address mostly Megaco

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Some Standards and protocols

- Signalling *(mostly inside IETF)*
  - SS7 to IP (SIGTRAN)
    - Transport of voice signalling over Internet
  - SIP, Megaco, MGCP, H.323, etc.
  - PINT (PSTN and Internet Interworking)
    - Mechanisms for the Internet to use POTS services (e.g click-to-dial, click-to-fax-back)
- Media *(some standards outside IETF)*
  - Real Time Protocol (RTP)
  - Echo cancelation
  - Voice coding (G.7xx)
- Major developments are in the call control field (or signalling)
  - **Web streaming has taken over these standards, embedding all complexity in a "transparent service"**

Multimedia in IP

The Web view

Multimedia Networking Applications

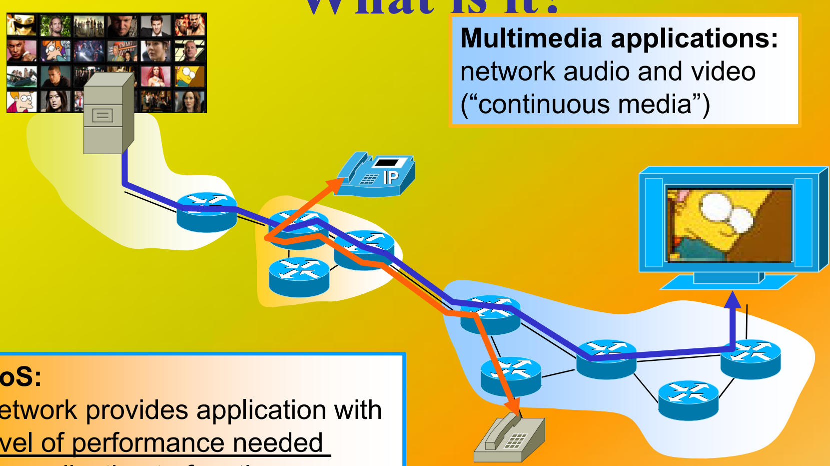
- Fundamental characteristics:
  - Typically delay sensitive
    - end-to-end delay
    - delay jitter
  - But loss tolerant: ... s glitches
  - Antithesis of data, which are loss

Jitter is the variability of packet delays within the same packet stream

- Classes of multimedia applications:
  - Streaming stored audio and video
  - Streaming live audio and video
  - Real-time interactive audio and video

Paulo Salvador (salvador@ua.pt)

Multimedia, Quality of Service: What is it?



Multimedia applications: network audio and video ("continuous media")

QoS:
Network provides application with level of performance needed for application to function.

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Internet Multimedia Support

- Integrated services philosophy.
  - Requires dedicated links/channels with QoS requirements.
- Differentiated services philosophy.
  - Fewer changes to Internet infrastructure.
- Best effort.
  - No major changes.
  - More bandwidth when needed.
  - Application-level control and distribution.

Would require QoS
Only possible in private networks or operator infrastructure

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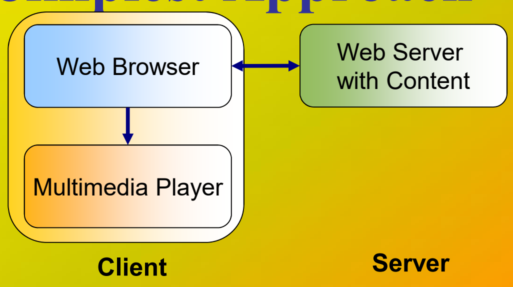
Internet Multimedia: Simplest Approach



- Audio or video stored in file.
- Files transferred as HTTP object (or using P2P).
  - Received in entirety at client as a file.
  - Then passed to default player in client.
- Audio&video is not streamed!
- No “pipelining”, long delays until playout!

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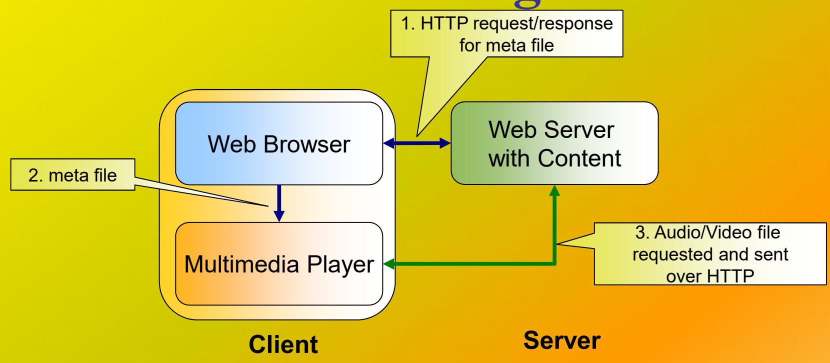
Internet Multimedia: Web Streaming



- Browser GETs metafile.
  - Content negotiation may happen.
- Browser launches player, passing metafile
- Player contacts server.
- Server streams audio/video to player.

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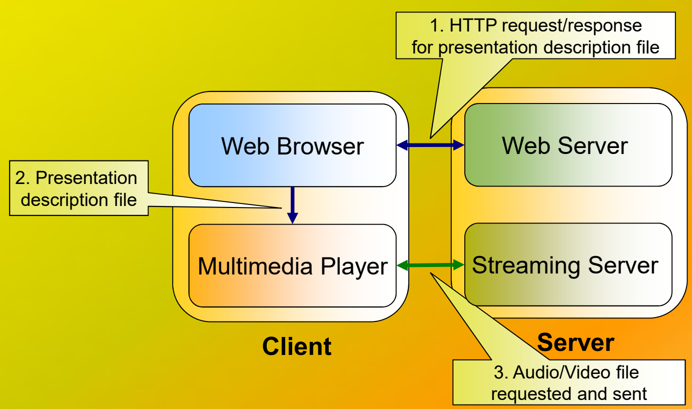
Streaming from a streaming server



- This architecture allows for non-HTTP protocol between server and media player.
- Can use UDP or TCP transport.

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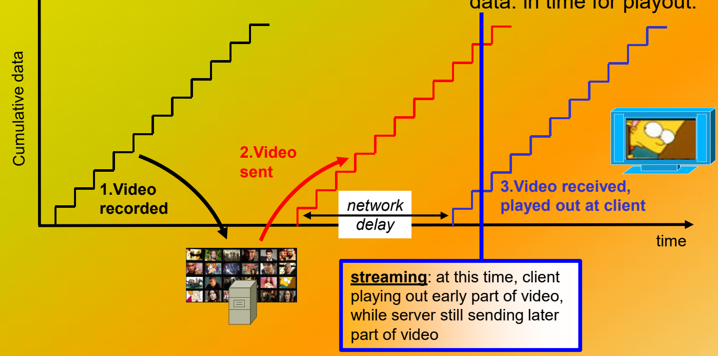
Web Streaming: Stored Multimedia

- Streaming:
  - Media stored at source is transmitted to client.
  - Client visualization begins before all data has arrived.
    - Timing constraint for transmitted data: in time for playout.



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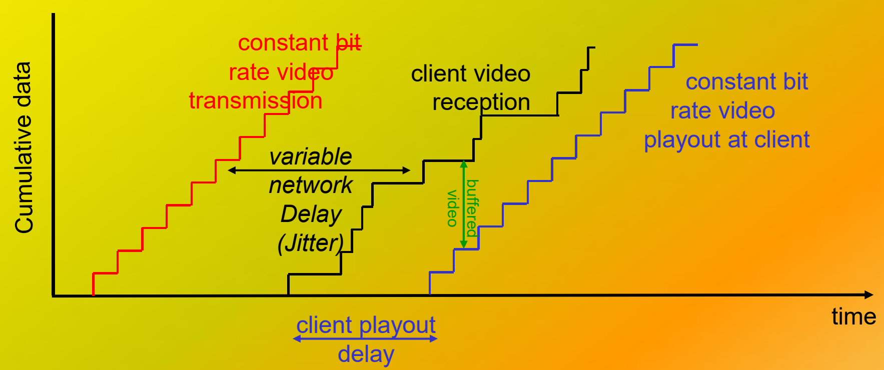
Streaming Multimedia: Client Buffering



- Client-side buffering, playout delay compensate for network-added delay, delay jitter.
  - Size of buffer/playout delay are parameters that can be adjusted dynamically
  - For VoIP, delay between 2 packets is about 20ms (plus *jitter*).

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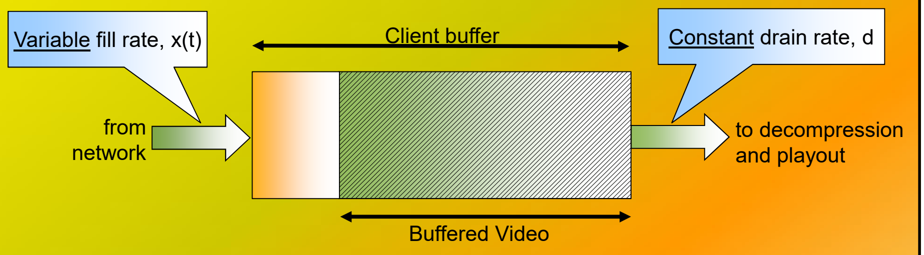
Streaming Multimedia: Client Buffering



- Client-side buffering, playout delay compensate for network-added delay, delay jitter.
  - Size of buffer/playout delay are parameters that can be adjusted dynamically

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Streaming Stored Multimedia

- Application-level streaming techniques for making the best out of best effort service:
  - Client side buffering.
  - Use of UDP versus TCP.
  - Multiple encodings of multimedia.
- Multimedia Player
  - Jitter removal,
  - Decompression,
  - Error concealment,
  - Graphical user interface with controls for interactivity.
- Network
  - Close to client content (multi-content) buffering for faster interactivity
  - Only viable in network operator proprietary services.

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Streaming Stored Multimedia: Interactivity

- VCR-like functionality: client can pause, rewind, fast-foward, push slider bar.
  - 10 sec initial delay OK.
  - 1-2 sec until command effect OK.
  - Timing constraint for still-to-be transmitted data: in time for playout.

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Streaming Live Multimedia

- Examples:
  - Internet TV/radio show.
  - Live sporting event.
- Streaming
  - Playback buffer.
  - Playback can lag tens of seconds after transmission.
  - Still have timing constraint.
- Interactivity
  - Fast forward impossible.
  - Rewind, pause possible!

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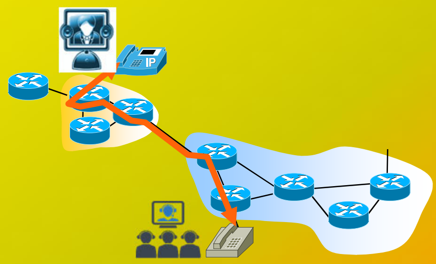
Interactive Real-Time Multimedia



- Applications:
  - IP telephony, video conference, online-game multimedia actions, distributed interactive worlds.
- End-end delay requirements:
  - Audio: < 150 msec good, < 400 msec OK
    - Includes application-level (packetization) and network delays.
    - Higher delays noticeable, impair interactivity.
- Requires session initialization
  - Advertise its IP address, port number, encoding algorithms, required contents, available contents

UDP Streaming vs. TCP Streaming

- UDP
  - Server sends at rate appropriate for client .
    - Often send rate = encoding rate = constant rate.
    - Then, fill rate = constant rate - packet loss.
  - Short playout delay (2-5 seconds) to compensate for network delay jitter.
  - Error recover: time permitting.
- TCP
  - Send at maximum possible rate under TCP.
  - Fill rate fluctuates due to TCP congestion control.
  - Larger playout delay: smooth TCP delivery rate.
  - HTTP/TCP passes more easily through firewalls.

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HTTP/TCP Streaming

- Multiple versions with distinct/complementary characteristics are generated for the same content
  - With different bitrates, resolutions, frame rates.
- Each version is divided into time segments.
  - e.g., two seconds.
- Each segment is provided on a web server and can be retrieved through standard HTTP GET requests.
- Examples of protocols:
  - MPEG's Dynamic Adaptive Streaming over HTTP (DASH).
    - Standard ISO/IEC 23009-1. YouTube's default.
  - Adobe HTTP Dynamic Streaming (HDS).
  - Apple HTTP Live Streaming (HLS).
  - Microsoft Smooth Streaming (MSS).

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User Control of Streaming Media: RTSP

- RTSP (Real Time Streaming Protocol): RFC 2326
  - Client-server application layer protocol.
  - For user to control display: rewind, fast forward, pause, resume, repositioning, etc…
- Does not define how audio/video is encapsulated for streaming over network.
- Does not restrict how streamed media is transported.
  - Can be transported over UDP or TCP.
- Does not specify how the media player buffers audio/video.
- RTSP messages are also sent out-of-band:
  - RTSP control messages use different port numbers than the media stream: out-of-band
    - Port 554
  - The media stream is considered “in-band”

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RTSP: out of band control

- FTP uses an "out-of-band" control channel:
  - A file is transferred over one TCP connection
  - Control information (directory changes, file deletion, file renaming, etc.) is sent over a separate TCP connection
  - The "out-of-band" and "in-band" channels use different port numbers
- RTSP messages are also sent out-of-band:
  - RTSP control messages use different port numbers than the media stream: out-of-band
  - Port 554
  - The media stream is considered "in-band"

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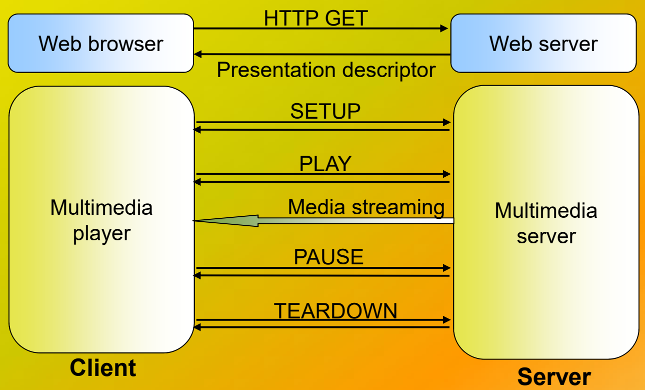
RTSP Operation



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RTSP Exchange Example

- C: SETUP rtsp://audio.example.com/twister/audio RTSP/1.0
- Transport: rtp/udp; compression; port=3056; mode=PLAY
- S: RTSP/1.0 200 1 OK
- Session 4231
- C: PLAY rtsp://audio.example.com/twister/audio.en/lofi RTSP/1.0
- Session: 4231
- Range: npt=0-
- C: PAUSE rtsp://audio.example.com/twister/audio.en/lofi RTSP/1.0
- Session: 4231
- Range: npt=37
- C: TEARDOWN rtsp://audio.example.com/twister/audio.en/lofi RTSP/1.0
- Session: 4231
- S: 200 3 OK

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Streaming Media: RTSP

➔RTSP: RFC 2326

- Application layer protocol (client-server)
- Presentation control of streaming
  - rewind, fast forward, pause, resume, reposition, etc…

Limitations:

- Does not define how audio/video is encapsulated for streaming (RTP)
- Does not impose transport mechanisms (UDP or TCP)
- Does not describe how the audio/video is played (nor the type of buffering)

Out of band control:

- RTSP messages use different ports from the “media stream” (which is “in-band”)
  - Port 554

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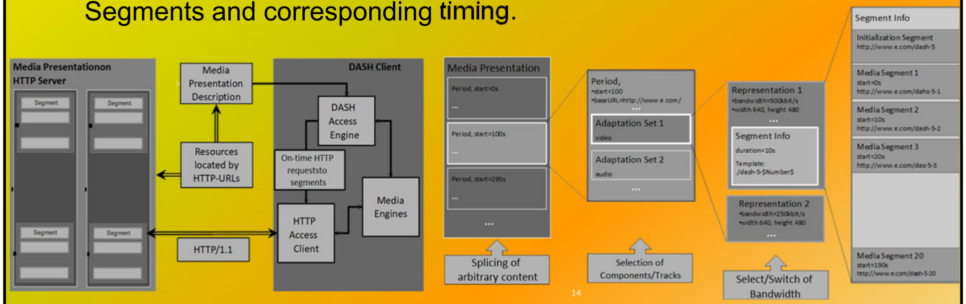
Dynamic Adaptive Streaming over HTTP (DASH)

- Developed to be an Open Standard Delivery Format.
  - MPEG DASH ISO/IEC 23009-1.
- Video streaming solution where pieces of video streams/files are requested with HTTP and spliced together by the client.
  - Client entirely controls delivery.
- Media Presentation Description (MPD) describes accessible Segments and corresponding timing.



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WebRTC

- Peer-to-peer connections.
  - An instance allows an application to establish peer-to-peer communications with another instance in another browser, or to another endpoint implementing the required protocols.
- RTP Media.
  - Allow a web application to send and receive media stream over a peer-to-peer connection (discussed in a minute)
- Peer-to-peer Data
  - Allows a web application to send and receive generic application data over a peer-to-peer connection.
- Peer-to-peer DTMF.

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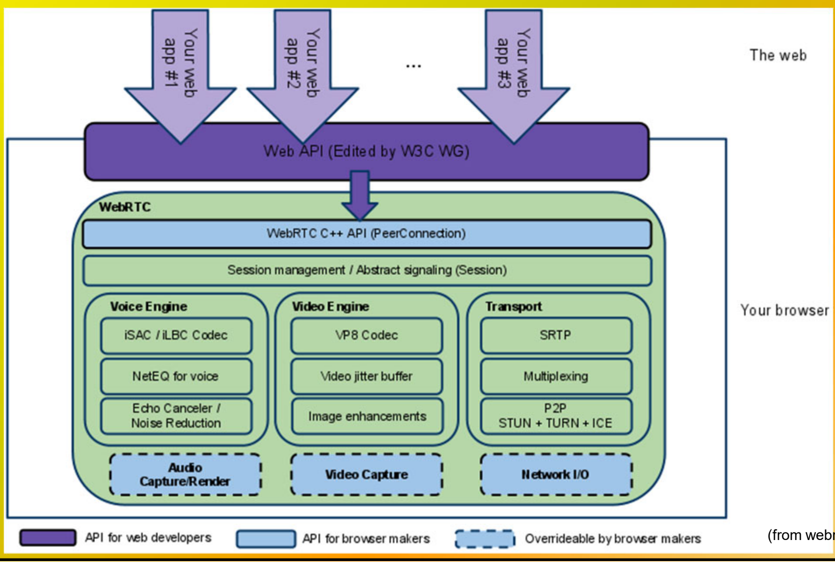
WebRTC Architecture



(from webrtc.org)

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Real Time Transport Protocol (RTP)

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Disadvantages of TCP

- Connection-oriented
  - Not appropriate to multicast
- Retains traffic (push)
- Retransmissions are not convenient to “soft” real time traffic (i.e. that accepts losses)
- Does not contain limitation on data length
- Does not provide timing information

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Disadvantages of UDP

- *Connectionless* packet service, without guaranties, without order
- Without error control
- Without flow control
- Without congestion control
- Using ports
- Should be used to real-time data:
  - Congestion and flow control mechanisms in TCP are not adequated

0 8 16 24 (31)

source port	destination port
length	checksum
data	

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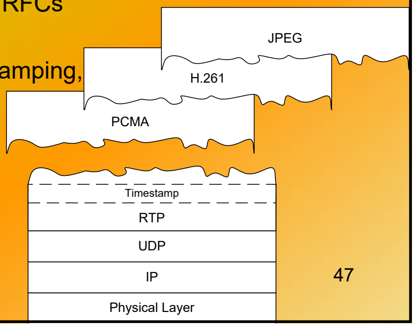
RTP

- RTP specifies a packet structure for packets carrying audio and video data
- Includes two distinct protocols over UDP
  - *RTP:* streaming transmission at application level (simple signaling)
    - » Data channel
  - *RTCP:* control messages
    - » Control and information channel
- RFC1889: general format of messages
  - Formats to specific media types in other RFCs
- Provides services of
  - Packet identification, sequencing, timestamping, feedback
  - But no delivery guaranties or QoS
- RTP runs in the end systems.
- Interoperability:
  - if two Internet applications run RTP, then they **may** be able to work together



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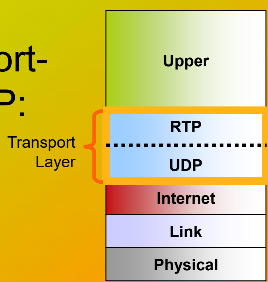
RTP runs on top of UDP

- RTP libraries provide a transport-layer interface that extend UDP:
  - Port numbers, IP addresses
  - Payload type identification
  - Packet sequence numbering
  - Time-stamping



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RTP and QoS

- RTP <u>does not</u> provide any mechanism to ensure timely delivery of data or provide other quality of service guarantees.
- RTP encapsulation is only seen at the end systems: it is not seen by intermediate routers.
  - Routers providing best-effort service do not make any special effort to ensure that RTP packets arrive at the destination in a timely matter.
  - Operators may create separate channels for specific services

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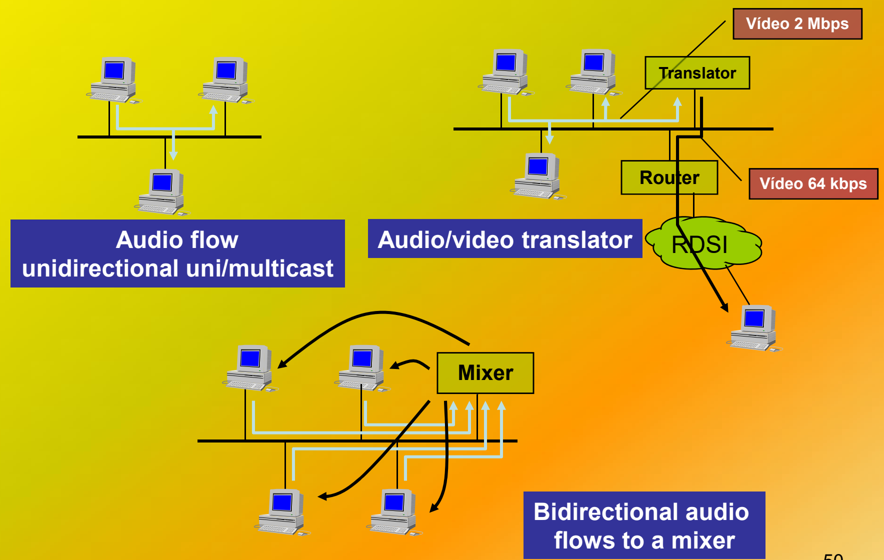
RTP scenarios



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Types of RTP entities (Relays)

- Translators
  - Modify data format
  - Do not modify SSRC nor timestamp
  - Examples:
    - Multicast to unicast, coding, flow quality reduction
- Mixers (SSRC field)
  - Generate an output through several inputs (CSRC field)
  - Examples:
    - Audio mixer, video PiP, ...

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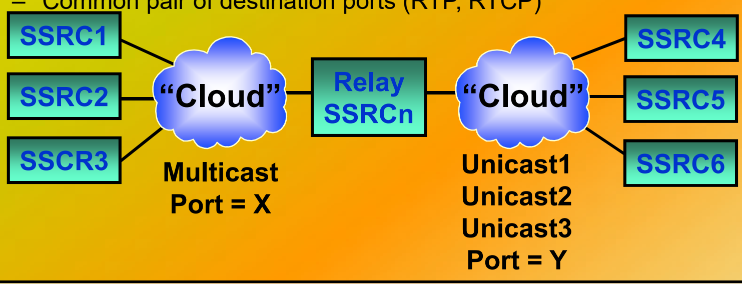
RTP session and cloud

- "session": "clouds" interconnected by relays, in which each participant has a unique identifier (SSRC)
  - Usually an unique address (multicast) and a pair of ports to RTP and RTCP
- "Transport cloud": sources, receivers and "relays" (translators/ mixers) of packet flows that share:
  - Direction of destination (1 multicast direction and 1 list of unicast directions)
  - Common pair of destination ports (RTP, RTCP)



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"Light" sessions

- There is no explicit control of group participation
  - Participants get together in groups
- There is no explicit conference control
- Members that have data to send, just send it
- Session packets quasi-periodical
  - Identity, reception reports, sincronization information
- Adequated to multicast models

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Concept: Application Level Framing (ALF)

- Application semantics should be reflected in the communication protocol
- Application to control the packing of information in packets
- It creates Application Data Units (ADUs)
  - Each ADU can be independently processed
- Associate ADUs in a single network packet (if possible)

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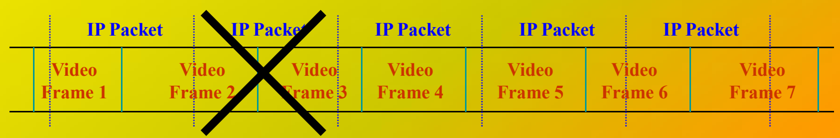
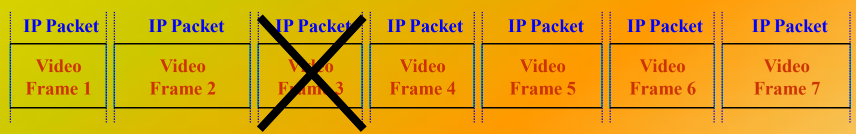
Example ALF:ADU

Application ignores the framing aspects (e.g. TCP)



Application frames are mapped in network layer packets (e.g. UDP without fragmentation)



Lost IP packet

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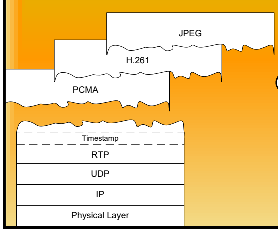
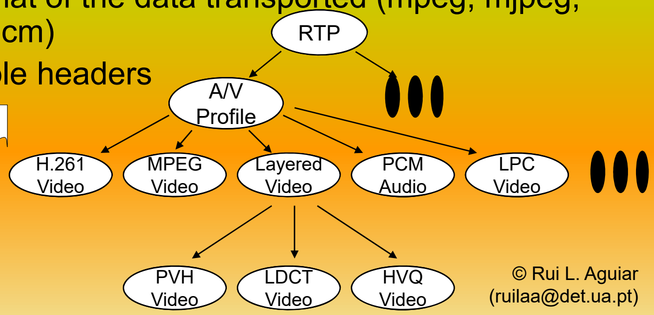
Back to RTP...

- Objective:
  - Global framework for packet delivery...
- A.L.F. principle appears in the RTP hierarchical structure (because of multimedia formats)
  - Requires profiles (e.g. Audio video)
  - And format of the data transported (mpeg, mjpeg, H.261, pcm)
  - Extensible headers





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Joint source and channel coding

- Principle
  - Corollary for A.L.F.
    - Need to consider the transmission channel when coding the data
- The source coding algorithm becomes sensitive to the network
  - Algorithms are modified for producing self-sufficient information.
  - Packet loss has low impact
- Example: H.261

  Next slides…

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Standard H.261 algorithm

- Video coding similar to MPEG.
- Predictive mechanism
  - "predicted image"
- Time compression
  - differential coding between frame N and frame N-1
- Assumes "lossless" channel
- If there are data loss...
  - Resynchronizes with the next Group of Blocks (GOB)
  - Reconstruction errors remain in the decoder.

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Intra H.261 Algorithm

- Reacts to IP network features
- Subset of H.261
- Conditional image reconstruction
- No differential reconstruction of frames
- Macroblocks become ADUs (application data units)

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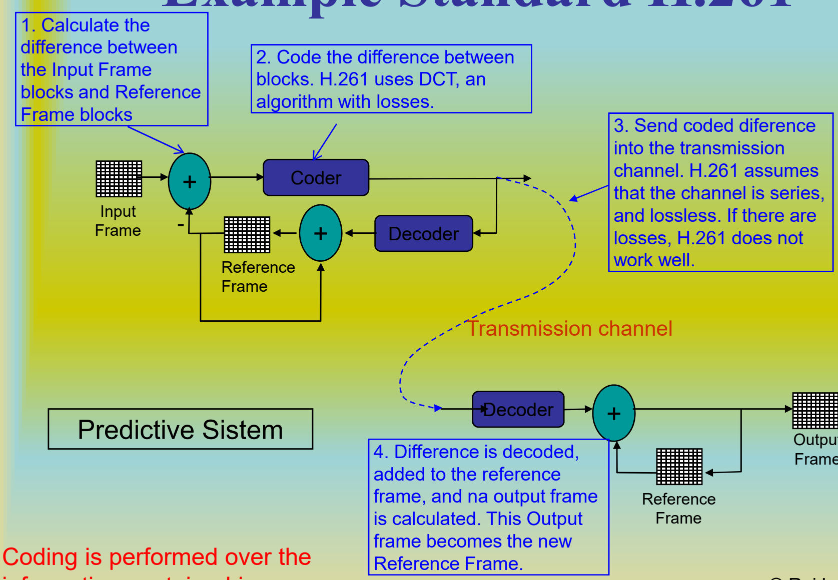
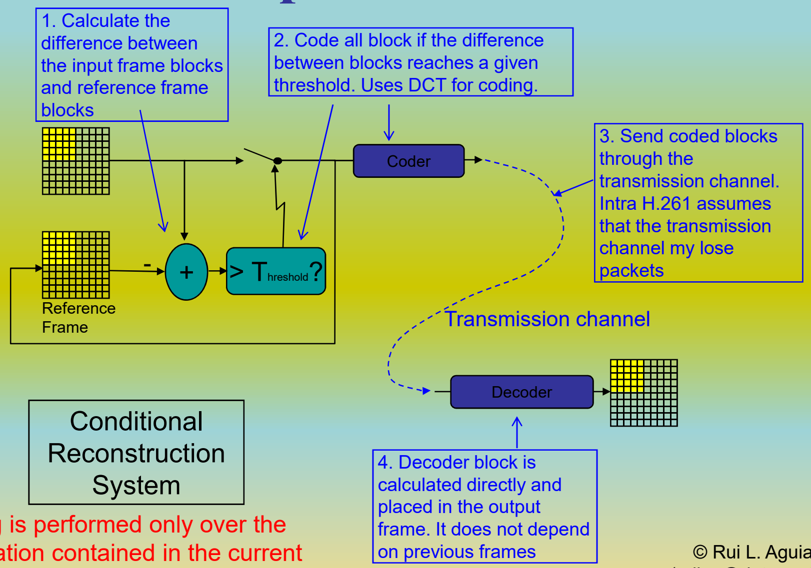
Example Standard H.261



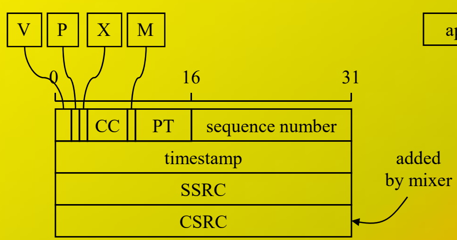
Coding is performed over the information contained in multiple adjacent frames

Example Intra H.261



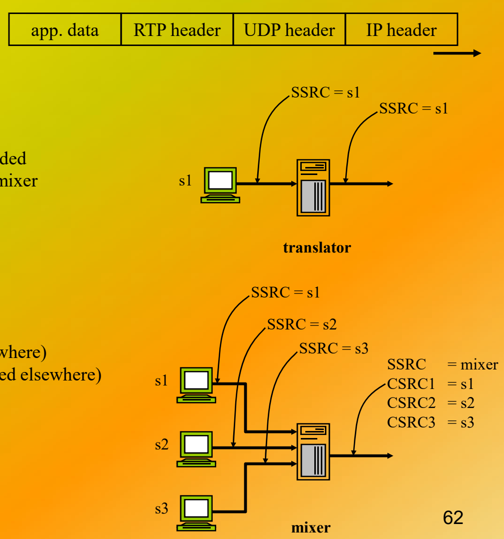
Coding is performed only over the information contained in the current frame

RTP: overview



V	2-bits, version number (=2)
P	1-bit, indicates padding
X	1-bit, indicates extension header present
CC	4-bits, number of CSRCs (CSRC count)
M	1-bit, profile specific marker (defined elsewhere)
PT	7-bits, payload type, profile specific (defined elsewhere)
SSRC	synchronisation source
CSRC	contributing source

timestamp has profile/flow-specific units



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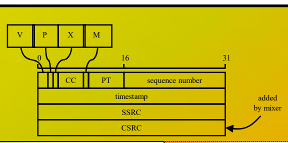
RTP Header



Flags	payload type	sequence number	timestamp	synchronization source identifier	misc. fields

- Flags:
  - **RC Count (CC)** – Number of CSRC identifiers (mixed)
  - **Marker (M)** – Announces frame limits: first packet of a burst in voice; end of packet sequence in video
- Payload Type (7 bits)
  - Indicates type of encoding currently being used. If sender changes encoding in middle of conference, sender informs the receiver through this payload type field
    - Payload type 0: PCM mu-law, 64 kbps
    - Payload type 3, GSM, 13 kbps
    - Payload type 7, LPC, 2.4 kbps
    - Payload type 26, Motion JPEG
    - Payload type 31. H.261
    - Payload type 33, MPEG2 video
- Sequence Number (16 bits)
  - Increments by one for each RTP packet sent, and may be used to detect packet loss and to restore packet sequence
- Timestamp field (32 bytes long)
  - Reflects the sampling instant of the first byte in the RTP data packet, in terms of sampling measurements
    - Audio: timestamp usually increases by 1 every sampling period (e.g., each 125 msecs to sampling at 8 KHz)
- SSRC field (32 bits long)
  - Identifies the source of the RTP stream. Each stream in a RTP session should have a distinct SSRC
- CSRC (*Contributing source identifiers*) – List of sources that contributed to the packet payload
  - Used when a mixer combines different packet sources
  - Receiver can then identify original sender

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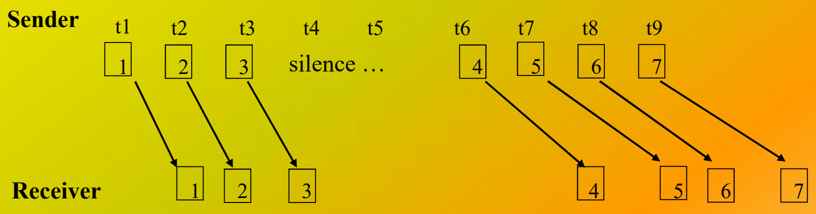
Timestamp vs sequence number

- Silence suppression and detection
- Variable length packets
- Jitter



"play" time vs packet loss detection

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RTP Example

- Consider sending 64 kbps PCM-encoded voice over RTP
- Application collects the encoded data in chunks, e.g., every 20 msec = 160 bytes in a chunk
- The audio chunk along with the RTP header form the RTP packet, which is encapsulated into a UDP segment
- RTP header indicates type of audio encoding in each packet
  - Sender can change encoding during a conference
- RTP header also contains sequence numbers and timestamps

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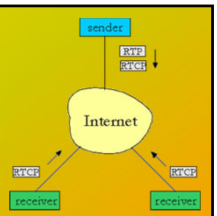
Real-Time Control Protocol (RTCP)



- Works associated to RTP, to obtain feedback information that can lead to behavior change
- Each participant in RTP session periodically transmits RTCP control packets to all other participants
- Sends all session in multicast: a multicast address per session, shared by RTP and RTCP packets
  - In different ports
  - RTCP traffic per participant is variable with time
- Each RTCP packet contains sender and/or receiver reports
  - Report statistics useful to application including number of packets sent, number of packets lost, interarrival jitter, etc...
- Essential to multicast
  - Diagnosis tool
  - Feedback control can lead to change in the sender transmission rate

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RTCP Protocol

- Provides information about reception quality
  - To senders and receivers
  - QoS information to the flow
    - packet info: loss, delay, jitter
    - end-system info: user info
    - application-specific or flow-specific info
- Identifies each participant
- Calculates the number of sources
- Minimum session control
  - Information about participants
  - Session leave, ...
  - Minimum synchronization
- Protocol “Announce-Listen”, *soft-state*
  - Good for scalability

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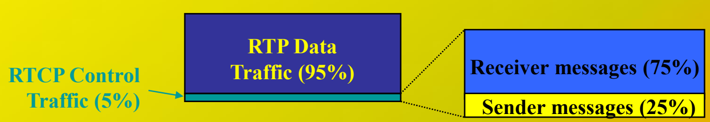
RTCP Scalability



- Uses 5% of the session bandwidth
  - Shared by all participants
  - 25% for senders
  - 75% for receivers
- Characteristics for scalability
  - Reports periodically sent with a variable delay
    - Information sent with different periodicity (e.g. name vs email)
  - Period based on the number of participants
    - Counting of the participants number, through the source identifier

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Types of RTCP packets

- Sender report (SR): sending of statistics by senders
  - SSRC of the RTP stream, the current time, the number of packets sent, and the number of bytes sent.
- Receiver Report (RR): sending of statistics by receivers
  - Fraction of packets lost, last sequence number, average interarrival jitter
- Source Description (SDES): CNAME, NAME, e-mail, ...
  - Sender e-mail address, sender's name, SSRC of associated RTP stream.
  - Provide mapping between the SSRC and the user/host name.
- BYE: Leaving the session
- APP: Specific for each application
- It is common the concatenation of PDUs: at least two should be sent in each UDP message
  - Mixers and translators also concatenate packets

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Sources (streams) synchronization

- RTP used to synchronize different streams
  - Consider videoconferencing application for which each sender generates one RTP stream for video and one for audio
    - Timestamps in RTP packets tied to the video and audio sampling clocks
    - Not tied to the wall-clock time
- RTCP reports have:
  - Timestamp in the last RTP packet
  - Time (wall clock) of the packet generation
- Receivers can use this information to synchronize different media sources (audio/video)
  - RTP timestamp value is centered in the sampling rates and not in the transmission time

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RTP limitations

- RTP standardizes and makes easier the transmission of continuous audio and video flows, but:
  - Does not reserve resources
  - Does not have QoS guaranties
  - Does not support congestion control
  - Does not support reliability
  - ...
- ...should work together with other protocols (RSVP) and networks (ATM) for QoS guaranties
  - An essential aspect for these flows
- Routers do not "see" RTP
  - They cannot provide priviledge services
- Scalability problems:
  - When many receivers get in the sessions simultaneously (many reports, not aggregated)

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VoIP
Voice (and Video and ...) over IP

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Overview recall: Voice over IP

- Network loss: IP datagram lost due to network congestion (router buffer overflow).
- Delay loss: IP datagram arrives too late for playout at receiver.
  - Delays: processing, queueing in network; end-system (sender, receiver) delays.
  - Typical maximum tolerable delay: 400 ms.
- Loss tolerance: depending on voice encoding, packet loss rates between 1% and 10% can be tolerated.
- Speaker's audio: alternating talk/speech with silent periods.
  - 64 kbps during talk/speech.
  - Packets generated only during talk/speech.
    - 20 msec chunks at 8 Kbytes/sec: 160 bytes data.
- Requires session establishment.
- VoIP protocols/frameworks:
  - Session Initiation Protocol (SIP)
    - Session Description Protocol (SDP)
  - H.323
- VoIP and PSTN interoperability in large/ISP scalable scenarios require complex control frameworks:
  - Media Gateway Controller Protocol (MGCP);
  - H.248/Megaco.

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SIP vs H.323

- SIP comes from IETF: Borrows much of its concepts from HTTP.
- H.323 is another signaling protocol for real-time, interactive.
  - Comes from the ITU (telephony).
- SIP has a Web flavor, whereas H.323 has a telephony flavor.
- SIP is a single component. Works with RTP, but it can be combined with other protocols and services.
- H.323 is a complete, vertically integrated suite of protocols for multimedia conferencing: signaling, registration, admission control, transport and codecs.

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H.323

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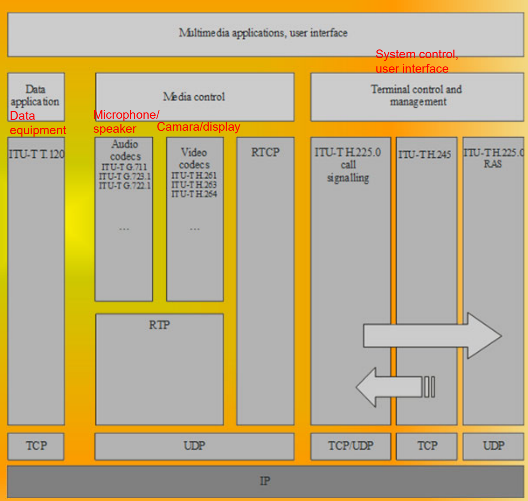
H.323

- H.323 is a set of recommendations from the International Telecommunication Union (ITU).
  - Contains several standards (signaling, control, transport, etc...).
- Consists of family of protocols that are
  - used for call set-up, call termination, registration, authentication and other functions.
  - transported over TCP or UDP protocols.



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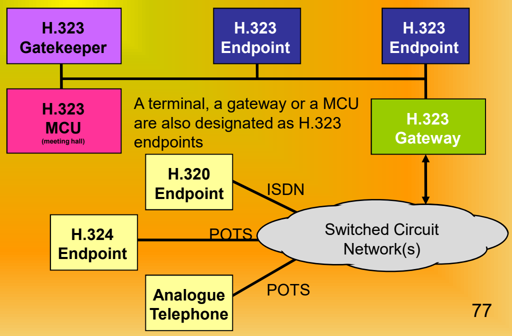
H.323

- Is a set of protocols for multimedia communications
- Can be used in any IP environment
- Contains several standards (signaling, control, transport, etc…)
- Defines everything, from codecs to the control of a remote camera



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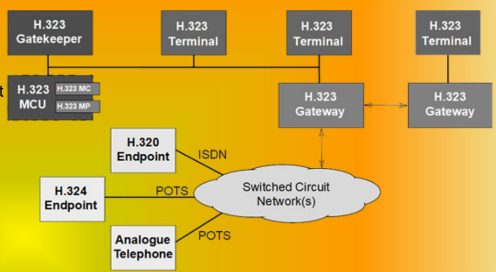
H.323 Elements

- Terminal
  - Endpoint on the network which provides for real-time, two-way communications with another H.323 terminal, Gateway, or Multipoint Control Unit.
  - This communication consists of control, indications, audio, video, and/or data between the two endpoints.
- Gateway (GW)
  - Endpoint on the network which provides for real-time, two-way communications between Terminals on the packet-based network and other Terminals on a switched circuit network or to another H.323 Gateway.
- Gatekeeper (GK)
  - Entity on the network that provides address translation and controls access to the network for H.323 terminals, Gateways and MCUs.
  - The Gatekeeper may also provide other services to the terminals, Gateways and MCUs such as bandwidth management and locating Gateways.



- Multipoint Control Unit (MCU)
  - Endpoint on the network which provides the capability for three or more terminals and Gateways to participate in a multipoint conference.
  - Consists of two parts: a mandatory Multipoint Controller (MC) and optional Multipoint Processors (MP).
  - Multipoint Controller (MC)
    - Entity on the network which provides for the control of three or more terminals participating in a multipoint conference.
  - Multipoint Processor (MP)
    - Entity on the network that provides centralized processing of audio, video and/or data streams in a multipoint conference.

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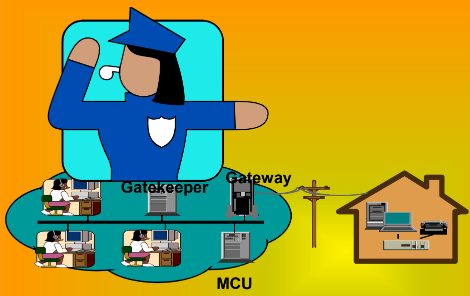
Gatekeeper



• Main point of network management (in a zone)
  Address translation (IP/telephone)
  Access control of network resources to terminals, gateways and MCU's
  Call management
  Bandwidth control
  Every endpoints of an H.323 network should be registered in the respective gatekeeper

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Gatekeeper in an H.323 system

- Gatekeeper is optional
  - When present, can provide a set of functionalities
    - Routing of call signalling (better control, intelligent routing decisions, load balacing of gateways)
    - However, these messages can be sent directly between users
- H.323 networks with IP/PSTN gateways should contain a gatekeeper to make address translation
- Mandatory functions
  - Address translation, admission and bandwidth control, zone management
- Optional functions
  - Call control signalling, call authorization and management

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Multipoint Control Unit

- Supports the required functionalities for three or more terminals and gateways to participate in a multi-point session
  - Multipoint Controller (MC)
    - Signalling and session control
  - Multipoint Processor (MP)
    - Processing (multiplexing and sending) of multimedia flows.
- MC e MP
  - Centralized multi-point session
    - Signalling, control, and multimedia data information traverse the MCU
- Only MC
  - Descentralized multi-point session
    - Only signalling and control information traverse the MCU

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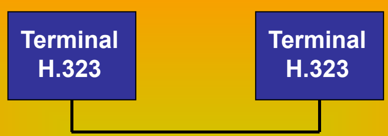
H.323 Terminals



•An H.323 terminal is a network endpoint that can have bidirectional real time communications with another H.323 terminal, gateway or MCU

•Signaling information, control, audio, video and/or data

•Hardware or software that implements H.323 terminal functionalities.

H.323 terminals support

- G.711 (PCM audio codec, 64Kbps)
- H.245 (media identification and control, such as voice and video)
- Q.931 (signaling for call establishment)
- Registration/Admission/Status (RAS) Channel (data channel to communicate with the Gatekeeper – it can or not exist)
- RTP/RTCP

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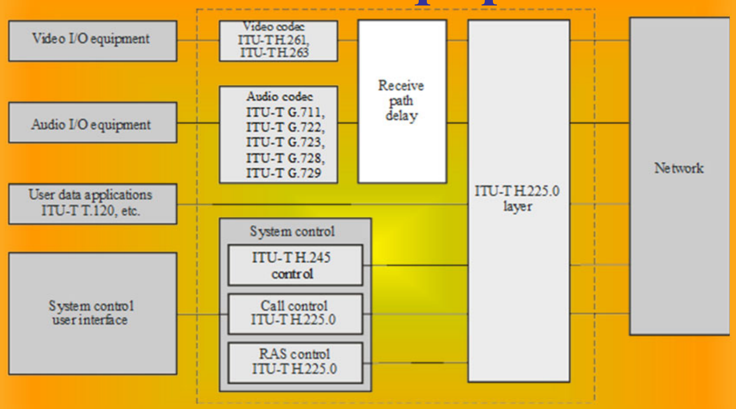
H.323 Terminal Equipment stack



- H.225
  - Registration, Admission and Status (RAS), which is used between an H.323 endpoint and a Gatekeeper to provide address resolution and admission control services.
  - Call Signaling, which is used between any two H.323 entities in order to establish communication (based on Q.931/Q.932).
- H.245
  - Control protocol for multimedia communication, which describes the messages and procedures used for capability exchange, opening and closing logical channels for audio, video and data, control and indications.

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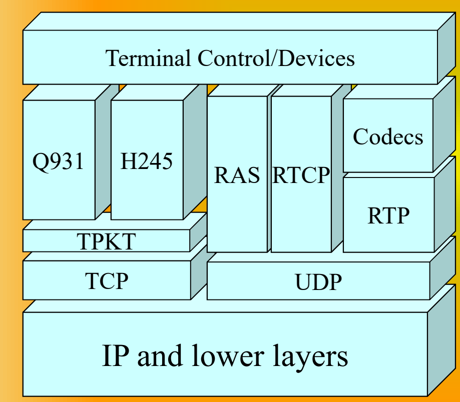
H.323 protocol stack



- Q931
  - Call signalling
- H245
  - Creation of channels and exchange of capacities
- RAS (registration, admission and status), H225.
  - Of terminals+gateways to gatekeepers
  - User registration
- Codecs:
  - Audio (G.711, G722,23.1,28,29)
  - Video (H.261, H.263)
- Data:
  - T.12x

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H.323 operation (protocols)

- Obtain gatekeeper permission ***(RAS Admission Request)***
- Find the address of the user to call ***(RAS Address resolution)***
- Press the number (call) ***(Q931 call setup)***
- Tell the partners what languages it understands/talks ***(H245 capability negotiation – Set, Ack, Reject)***
- Wait for the communication of its capabilities ***(H245 capability negotiation – Set, Ack, Reject)***
- Inform what languages will be used during the conversation ***(H245 Logical channel signaling; languages=codecs)***
- Start talking (and listening) ***(Data transfer with RTP/RTCP)***
- Upon termination, say Bye ***(H245 end session)***
- Disconnect ***(Q931 call termination, release complete)***
- Inform the gatekeeper that the call ended ***(RAS Disengage Request)***

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H.323 operation (more...)

- Multiples languages can be used during the communication
- The language can be changed during conversation, as long as the other partner understands it
  - An explicit announcement has to be done
- Say Bye before terminating is optional….
- ...

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H.225 RAS Messages

gatekeeper discovery and registration

- Gatekeeper discovery:
  - **Gatekeeper Request (GRQ), Gatekeeper Confirm (GCF)** and **Gatekeeper Reject (GRJ)**
  - If one gatekeeper answers positively, the endpoint should select which one to use.
- Endpoints registration:
  - **Registration Request (RRQ)** and **Unregistration Request (URQ)**
- Endpoints location:
  - **Location Request (LRQ), Location Confirm (LCF)** and **Location Reject (LRJ)**
  - Through the alias of another endpoint, it can obtain contact information of that endpoint.
- Admission to participate in a session:
  - **Admission Request (ARQ), Admission Confirmation (ACF) and Admission Reject (ARJ)**
- Change of bandwidth by an endpoint or gatekeeper
  - **Bandwidth Request (BRQ), Bandwidth Confirm (BCF)** and **Bandwidth Request (BRJ)**
- State information of an endpoint:
  - **Information Request (IRQ)** and **Information Request Response (IRR)**
- Session leave:
  - **Disengage Request (DRQ), Disengage Confirm (DCF**) and **Disengage Reject (DRJ)**
- Communication of available resources - gateways should inform gatekeepers about its capacities:
  - **Resource Available Indicate (RAI)** and **Resource Available Confirmation (RAC)**

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H.225 Call Signaling Q.931 Messages

- Call establishment messages:
  - **Setup, Setup Acknowledge, Alerting**, **Call Proceeding**, **Connect**, **Connect Acknowledge**, and **Progress**.
- Call Clearing messages:
  - **Disconnect**, **Release,** and **Release Complete**.
- Call Information Phase messages:
  - **Resume**, **Resume Acknowledge**, **Resume Reject**, **Suspend**, **Suspend Acknowledge**, **Suspend Reject**, and **User Information.**
- Miscellaneous messages:
  - **Congestion Control**, **Information**, **Notify**, **Status**, and **Status Inquiry**.
- Q.932/H.450 messages:
  - **Facility**, **Hold**, **Hold Acknowledge**, **Hold Reject**, **Retrieve**, **Retrieve Acknowledge**, and **Retrieve Reject**.

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H.225 Call Signaling (most common)

- **Setup** - Establish a session between endpoints.
- **Call Proceeding** (optional) - answer to a setup indicating that it received the establishment process of the running session.
- **Alerting** - message sent by a callee to indicate that the user was already notified (corresponds to the phone ringing).
- **Progress** - optional message sent by a gateway to indicate that the session is in progress.
- **Connect** - message sent by a callee that indicates session acceptation.
- **Release Complete** - message sent by an endpoint to terminate a session.
- **Facility** - message sent by an endpoint to another one to inform where to redirect the session (other information can be sent)
- **Notify** - optional message used by any H.323 entity to send information to another one.
- **Status Inquiry** - message used by an endpoint during a session lifetime to ask another one about its status.
- **Status** - message used to answer to a status inquiry message.

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Q.931 Call Signaling – establish, control and terminate connections

- *Setup* – Establish a session between endpoints
- *call proceeding* (optional) – answer to a setup indicating that it received the establishment process of the running session
- *Alerting* – message sent by a callee to indicate that the user was already notified (corresponds to the phone ringing)
- *progress* – optional message sent by a gateway to indicate that the session is in progress
- *connect* – message sent by a callee that indicates session acceptation
- *release complete* – message sent by an endpoint to terminate a session
- *facility* - message sent by an endpoint to another one to inform where to redirect the session (other information can be sent)
- *notify* - optional message used by any H.323 entity to send information to another one
- *status inquiry* – message used by an endpoint during a session lifetime to ask another one about its status
- *status* – message used to answer to a *status inquiry* message

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H.245 Control Messages

- Capacities and preferences negotiation of each participant entity
- Signalling of logical channels used for data communication
- Used after the exchange of **Setup** and **Connect** messages to open an H.245 control channel.
- Capacities negotiation (supported formats for sending and reception):
  - **terminalCapabilitySet, terminalCapabilitySetAck, terminalCapabilitySetReject**
- Master/slave determination to solve conflicts that may appear during a session lifetime:
  - **masterSlaveDetermination, masterSlaveDeterminationAck, masterSlaveDeterminationReject**
- Opening of logical channels for several flows:
  - **openLogicalChannel, openLogicalChannelAck, openLogicalChannelConfirm, openLogicalChannelReject**
- Closing of logical channels:
  - **closeLogicalChannel, closeLogicalChannelAck, requestChannelClose, requestLogicalChannelAck, requestLogicalChannelReject**
- When all logical channels are closed, the session can be terminated:
  - **endSession**

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ITU Recommendations

	H.320	H.321	H.322	H.323v1 H323v2	H.324
Network	Narrowband ISDN	Broadband ISDN/ATM	Guaranteed B/W	Non-Guaranteed B/W	PSTN/POTS
Approval	1990	1995	1995	1996/1998v2	1996
Audio	G.711 G.722 G.728	G.711 G.722 G.728	G.711 G.722 G.728	G.711 G.722 G.728 G.723.1 G.729, 729A	G.723
Video	H.261 H.263	H.261 H.263	H.261 H.263	H.261 H.263	H.261 H.263
Data	T.120	T.120	T.120	T.120	T.120
Control	H.230 H.243	H.242	H.230 H.242	H.245	H.245

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H.323 Call (with Gatekeepers)

- Multiple messages may be transported in the same IP packet.

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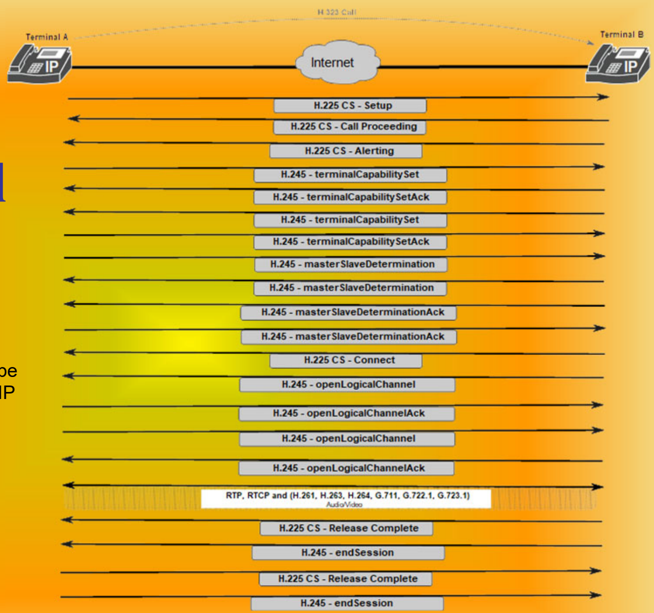
H.323 Direct Call



- Multiple messages may be transported in the same IP packet.

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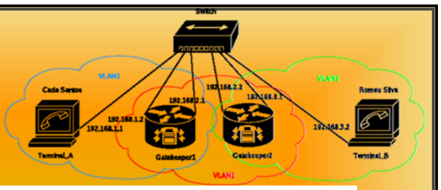
H.323 Session



No.	Time	Source	Destination	Protocol	Info
1	0.000000	192.168.1.1	255.255.255.2	H.225.	RAS: gatekeeperRequest
2	0.001943	192.168.1.2	192.168.1.1	H.225.	RAS: gatekeeperConfirm
3	0.011375	192.168.1.1	192.168.1.2	H.225.	RAS: registrationRequest
4	0.014833	192.168.1.2	192.168.1.1	H.225.	RAS: registrationConfirm
5	24.33812	192.168.1.1	192.168.1.2	H.225.	RAS: admissionRequest
6	24.34116	192.168.1.2	192.168.1.1	H.225.	RAS: admissionConfirm
7	24.34617	192.168.1.1	192.168.1.2	H.225.	CS: setup
8	24.36137	192.168.1.2	192.168.1.1	H.225.	CS: callProceeding
9	24.39984	192.168.1.1	192.168.1.2	H.225.	RAS: infoRequestResponse
10	24.90758	192.168.1.2	192.168.1.1	H.225.	CS: connect
11	24.92385	192.168.1.1	192.168.1.2	H.245	TerminalCapabilitySet
12	24.93220	192.168.1.1	192.168.1.2	H.245	MasterSlaveDetermination
13	25.00825	192.168.1.2	192.168.1.1	H.245	TerminalCapabilitySet
14	25.01673	192.168.1.2	192.168.1.1	H.245	MasterSlaveDetermination
15	25.02929	192.168.1.2	192.168.1.1	H.245	TerminalCapabilitySetAck
16	25.03250	192.168.1.2	192.168.1.1	H.245	MasterSlaveDeterminationAck
17	25.04127	192.168.1.1	192.168.1.2	H.245	TerminalCapabilitySetAck
18	25.04409	192.168.1.1	192.168.1.2	H.245	MasterSlaveDeterminationAck
19	25.06737	192.168.1.1	192.168.1.2	H.245	OpenLogicalChannel (gsmFullRate)
20	25.27152	192.168.1.2	192.168.1.1	H.245	[TCP ACKed lost segment] OpenLogicalChannel
21	25.29772	192.168.1.1	192.168.1.2	H.245	OpenLogicalChannelAck
22	25.30046	192.168.1.2	192.168.1.1	H.245	OpenLogicalChannelAck
23	25.40158	192.168.1.1	192.168.3.2	RTP	Payload type=GSM 06.10, SSRC=3780823517, Se
24	25.48155	192.168.1.1	192.168.3.2	RTP	Payload type=GSM 06.10, SSRC=3780823517, Se
25	25.60718	192.168.3.2	192.168.1.1	RTP	Payload type=ITU-T G.711 PCMU, SSRC=309704
26	25.60762	192.168.3.2	192.168.1.1	RTP	Payload type=ITU-T G.721, SSRC=3097049058,
27	31.57993	192.168.1.1	192.168.1.2	H.245	[TCP ACKed lost segment] EndSessionCommand
28	31.58027	192.168.1.1	192.168.1.2	H.225.	CS: releaseComplete
29	31.58559	192.168.1.2	192.168.1.1	H.245	EndSessionCommand
30	31.68600	192.168.1.1	192.168.1.2	H.225.	RAS: disengageRequest
31	31.68737	192.168.1.2	192.168.1.1	H.225.	RAS: disengageConfirm

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H.323 Direct Call

Source	Destination	Protocol	Length	Info
192.168.56.1	192.168.56.101	H.225.0/H	155	masterSlaveDetermination terminalCapabilitySet CS: setup OpenLogicalChannel
192.168.56.101	192.168.56.1	H.225.0	181	CS: callProceeding
192.168.56.101	192.168.56.1	H.225.0/H	460	masterSlaveDeterminationAck terminalCapabilitySetAck terminalCapabilitySet CS: empty
192.168.56.101	192.168.56.1	H.225.0	181	CS: alerting
192.168.56.1	192.168.56.101	H.225.0/H	109	masterSlaveDeterminationAck terminalCapabilitySetAck CS: empty
192.168.56.101	192.168.56.1	H.225.0/H	106	roundTripDelayRequest CS: empty
192.168.56.1	192.168.56.101	H.225.0/H	106	roundTripDelayResponse CS: empty
192.168.56.101	192.168.56.1	H.225.0	360	CS: connect OpenLogicalChannel
192.168.56.101	192.168.56.1	H.225.0/H	131	endSessionCommand CS: releaseComplete
192.168.56.1	192.168.56.101	H.225.0/H	131	endSessionCommand CS: releaseComplete

Source	Destination	Protocol	Length	Info
192.168.56.101	192.168.56.1	H.261	1023	H.261 message
192.168.56.101	192.168.56.1	H.261	1021	H.261 message
192.168.56.101	192.168.56.1	H.261	353	H.261 message
192.168.56.1	192.168.56.101	RTP	106	PT=DynamicRTP-Type-112, SSRC=0x8A1B4D04, Seq=48411, Time=0, Mark
192.168.56.1	192.168.56.101	RTP	106	PT=DynamicRTP-Type-112, SSRC=0x8A1B4D04, Seq=48412, Time=320
192.168.56.1	192.168.56.101	RTP	106	PT=DynamicRTP-Type-112, SSRC=0x8A1B4D04, Seq=48413, Time=640
192.168.56.101	192.168.56.1	H.261	336	H.261 message
192.168.56.101	192.168.56.1	RTP	214	PT=ITU-T G.722, SSRC=0x56D0B555, Seq=6053, Time=0, Mark
192.168.56.101	192.168.56.1	RTP	214	PT=ITU-T G.722, SSRC=0x56D0B555, Seq=6054, Time=160
192.168.56.101	192.168.56.1	RTP	214	PT=ITU-T G.722, SSRC=0x56D0B555, Seq=6055, Time=320
192.168.56.1	192.168.56.101	RTP	106	PT=DynamicRTP-Type-112, SSRC=0x8A1B4D04, Seq=48414, Time=960
192.168.56.101	192.168.56.1	RTP	214	PT=ITU-T G.722, SSRC=0x56D0B555, Seq=6056, Time=480
192.168.56.101	192.168.56.1	H.261	386	H.261 message
192.168.56.1	192.168.56.101	H.261	1023	H.261 message
192.168.56.1	192.168.56.101	RTP	106	PT=DynamicRTP-Type-112, SSRC=0x8A1B4D04, Seq=48415, Time=1280
192.168.56.101	192.168.56.1	RTP	214	PT=ITU-T G.722, SSRC=0x56D0B555, Seq=6057, Time=640
192.168.56.101	192.168.56.1	RTP	214	PT=ITU-T G.722, SSRC=0x56D0B555, Seq=6058, Time=800
192.168.56.1	192.168.56.101	RTP	106	PT=DynamicRTP-Type-112, SSRC=0x8A1B4D04, Seq=48416, Time=1600
192.168.56.101	192.168.56.1	H.261	346	H.261 message
192.168.56.101	192.168.56.1	RTP	214	PT=ITU-T G.722, SSRC=0x56D0B555, Seq=6059, Time=960
192.168.56.1	192.168.56.101	H.261	1021	H.261 message
192.168.56.1	192.168.56.101	RTP	106	PT=DynamicRTP-Type-112, SSRC=0x8A1B4D04, Seq=48417, Time=1920
192.168.56.101	192.168.56.1	RTP	214	PT=ITU-T G.722, SSRC=0x56D0B555, Seq=6060, Time=1120

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Advantages and disadvantages

+ Works!
+ There are many implementations... some are free…
+ Supports many languages (codecs).
+ Interoperates with other languages: H320 (ISDN), H324 (POTS)
+ Good role in a specific transition period of coexistence

- Very complicated…
- Many protocols can be combined in the H.323 environment... It implies technology redundancy….
- Problems when Gatekeeper is overloaded and MCU is full!
- Firewalls ? ... Difficult to develop (multiple ports have to be managed in a conversation).

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Session Initiation Protocol (SIP)

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SIP

- SIP = base protocol to establish sessions in the internet (peer-to-peer), low complexity and generic
  - Developed by IETF mmusic group, since 1995
  - Peer-to-peer signalling protocol (RFC 2543 in 1999, RFC 3261)
- Transports session description information of initiator (caller) to destination (callee)
  - Client-server approach (origination/answer)
  - Independent of protocol (UDP, TCP, AAL5, ...)
    - Supports multicast
    - But generally works through UDP…
    - Security at the transport and network layer provided with TLS (requires TCP) or IPSec
- Supports change of parameters in the middle of the session
- Signaling messages not frequent
  - Always with acknowledges
- Objective:
  - Allow maximum re-utilization of existent protocols
  - Use HTTP-alike coding (text-based)
  - Reuse already existent addresses (URLs, DNS, proxies…)
  - Be an alternative to H.323
  - Supporting new services
  - Being scalable, extensible

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SIP allows...

- Create, modify and terminate multimedia sessions with two or more participants
  - VoIP, distribution of multimedia data and multimedia conference
- Provides functionalities that can be used to implement the following services
  - Users location
  - Users availability
  - Determination of users capabilities
  - Negotiation (and re-negotiation) of the parameters of users participating in a session
  - Negotiation of session characteristics
    - Session Description Protocol
  - Users mobility
  - Security mechanisms
    - Prevention of denial of service attacks
    - Users authentication
    - Message integrity and confidentiality
- It does not distribute multimedia data
  - Part of IETF architecture of conference control (+SAP, + RTSP, + SDP, ...)
- It is not able to control media gateways

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SIP functionalities

- SIP supports five communication aspects:
  - User location - (given an e-mail type address) determination of the end system to be used for communication
    - Distributed directory lookups
  - User capabilities - determination of the media and media parameters to be used
  - User availability - determination of the willingness of the called party to engage in communications
  - Call (session) establishment - "ringing", establishment of session parameters at both called and calling party;
    - Including multi-party, using an MCU, or a fully-meshed strategy
  - Call (session) control - including transfer and termination of sessions, modifying session parameters, and invoking services
    - (Re)-negotiation of call parameters
    - Forwarding: manual and automatic
    - Personal mobility: different terminals with the same identifier
    - Call center: reach the first (load distribution) or all (conference)
    - Initiates, modifies and terminates sessions (conferences)
      - Including between gateways to the PSTN
- SIP has been heavily explored in current network concepts (IMS)

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SIP clients and servers

User Agent Server, UAS
User Agent Client, UAC

UserAgent

Servers

Gateways
Registrar
Redirect
Proxy

- UAC: user-agent client (application that starts the call)
- UAS: user-agent server that accepts, redirects or rejects calls
- redirect server: redirect requests
- proxy server: server + client; controls the call, gets the address of the proxy callee, can also redirect
- registrar: registers the location of the user
- user agent = UAC + UAS
  - Usually combine a registrar + (proxy or redirect server)

SIP Clients and Servers

- SIP Clients
  - Phones (software based or hardware).
  - Gateways
  - User Agents
  - A User Agent acts as a
    - Client when it initiates a request (UAC),
    - Server when it responds to a request (UAS).
- SIP Servers
  - Proxy server
    - Receives SIP requests from a client and forwards them on the client's behalf.
    - Receives SIP messages and forward them to the next SIP server in the network.
    - Provides functions such as authentication, authorization, network access control, routing, reliable request retransmission, and security.
  - Redirect server
    - Provides the client with information about the next hop or hops that a message should take and then the client contacts the next-hop server or UAS directly.
  - Registrar server
    - Processes requests from UACs for registration of their current location.
    - Registrar servers are often co-located with a redirect or proxy server.

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Proxy servers

- Intermediate entities that behave as servers and clients
  - Make requests in name of other clients
- Get location of other endpoints
- Route SIP messages
- Optional
  - Authentication and accounting

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Registration and redirect servers

- Registration
  - Entities where users register their UAs
  - Allow the mapping between users addresses and their UAs addresses
    - Database of location service
    - Request to database by proxy and redirect servers
      - Routing and redirect of messages
- Redirect
  - Returns alternative locations of UAs and servers
    - Receives requests
    - Requests the location service
    - Returns a list of alternative addresses to where the request should be redirected

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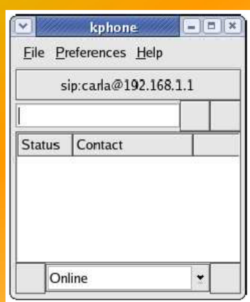
User Agent

- Endpoint of sessions
  - Initiates and terminates sessions
  - User Agent Server (UAS) and User Agent Client (UAC)
- UAC
  - Creates the requests (e.g. to initiate a session)
- UAS
  - Generates answers to requests (e.g. to answer a session request)
- Hardware or software equipment that implements UA functions



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SIP addresses

- URI (*Uniform Resource Identifier*)
  - Translated, by proxy server, to the UA address used by the user
  - A same user can have and use different UAs
  - sip:user@host:port;uri-parameters?headers
    - *uri-parameters* are parameters that affect the request for the resource identified by SIP URI
    - *headers* are fields to be included in the request
- sip:275313364@telecom.pt;user=phone
  - Identifies a user or a resource through the phone number 275313364 in the *telecom.pt* domain
  - To enforce that it is a phone number, the parameter *user* with the value *phone* is used

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SIP Messages

- SIP used for Peer-to-Peer Communication though it uses a Client-Server model.
- SIP is a text-based protocol and uses the UTF-8 charset.
- A SIP message is either a **request** from a client to a server, or a **response** from a server to a client.
  - A request message consists of a Request-Line, one or more header fields, an empty line indicating the end of the header fields, and an optional message-body;
  - A response message consists of a Status-Line, one or more header fields, an empty line indicating the end of the header fields, and an optional message-body.
  - All lines (including empty ones) must be terminated by a carriage-return line-feed sequence (CRLF).

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SIP messages

Method	Propósito
REGISTER	Registar um UA no serviço de localização.
INVITE	Estabelecer ou alterar os parâmetros de uma sessão.
ACK	Confirmar a recepção da resposta a um pedido de sessão.
CANCEL	Terminar um pedido de sessão pendente.
BYE	Terminar uma sessão.
OPTIONS	Interrogar uma entidade acerca das suas capacidades.

Some fields

- *To* – address of the destination entity
- *From* – address of the entity that sends the message
- *Call-ID* - identifies, together with the parameters *tag* of fields *To* and *From*, each session SIP and all registration requests of a UA
- *Via* – contains information about a path followed by the request from its origin, that should be used to route the answer
- *Proxy-Authenticate* – contains a challenge sent by a proxy server to be used in the authentication
- *Proxy-Authorization* – contains the answer to the challenge sent by a proxy server
- *Route* – used to indicate the route of a request through a set of proxy servers

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SIP Responses Codes and Purposes

- The first digit of the Status-Code defines the class of response.
  - 1xx: Provisional - request received, continuing to process the request;
  - 2xx: Success - the action was successfully received, understood, and accepted;
  - 3xx: Redirection - further action needs to be taken in order to complete the request;
  - 4xx: Client Error - the request contains bad syntax or cannot be fulfilled at this server;
  - 5xx: Server Error - the server failed to fulfill an apparently valid request;
  - 6xx: Global Failure - the request cannot be fulfilled at any server.

- Common Response codes:
  - 100 Trying
    - The request has been received and that some unspecified action is being taken.
  - 180 Ringing
    - Trying to alert the user.
  - 200 OK
  - 301 Moved Permanently and 302 Moved Temporarily
    - User can no longer be found at the address in the Request-URI.
  - 400 Bad Request
    - Request could not be understood.
  - 401 Unauthorized
    - Request requires user authentication.
  - 403 Forbidden
    - Server understood the request, but is refusing to fulfill it.
  - 404 Not Found
    - Server has definitive information that the user does not exist.

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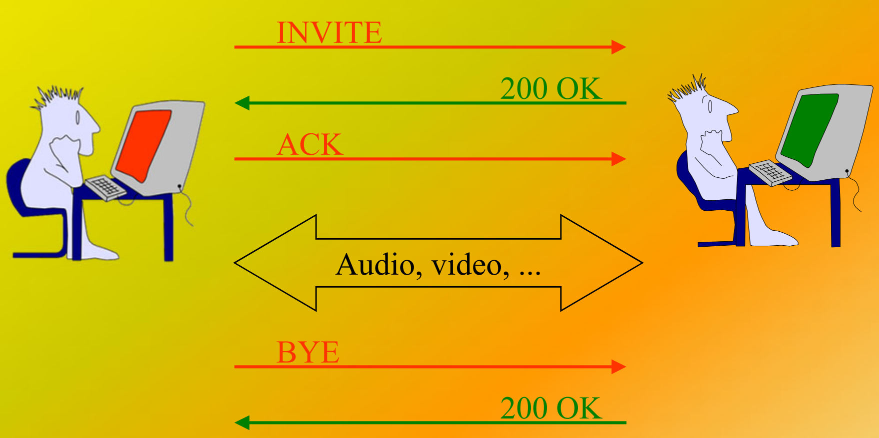
Signalling – basic example



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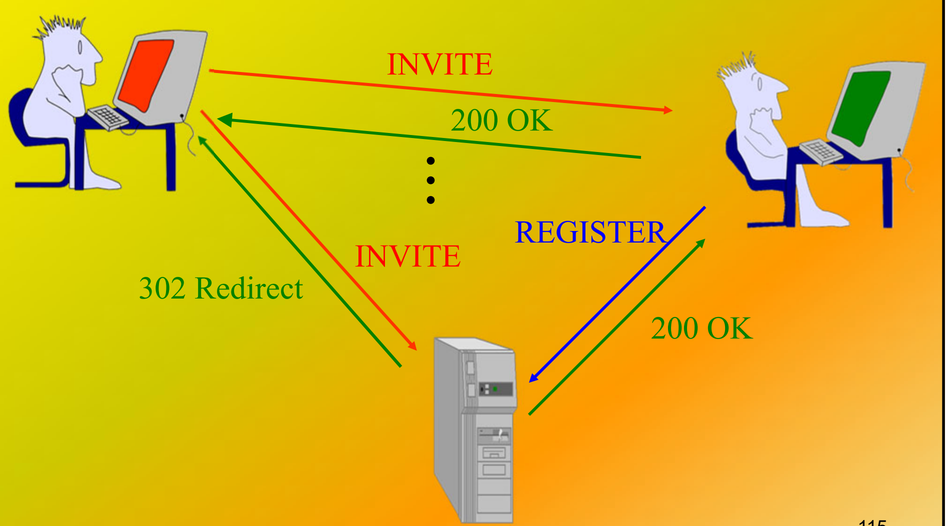
Redirect example



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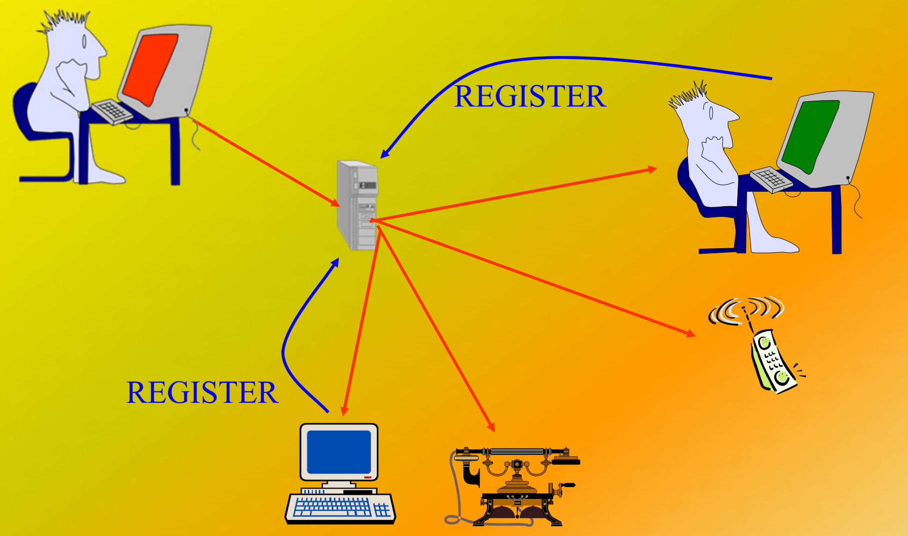
Multi-register example



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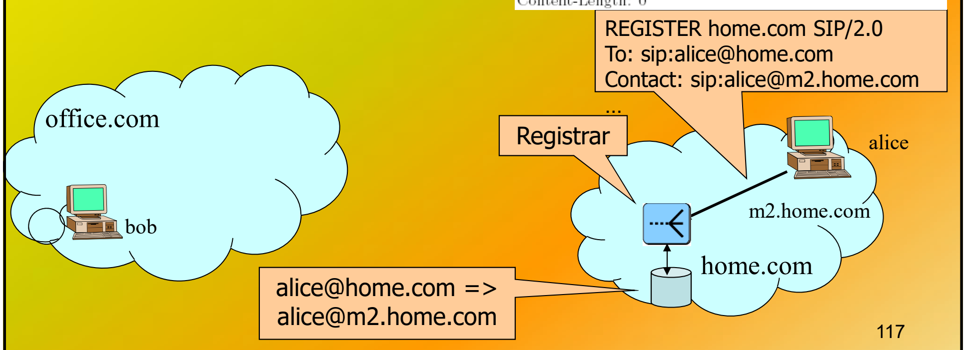
Setup of a basic call - registration

- Identifier of email type:

<sip:alice@home.com>

- Alice phone registers in home.com

```
REGISTER sip:A.com SIP/2.0
Via: SIP/2.0/UDP pc.A.com:5060;branch=z9hG4bKnashds7
To: Carla <sip:carla@A.com>
From: Carla <sip:carla@A.com>;tag=456248
Call-ID: 843817637684230@998sdasdh09
CSeq: 1826 REGISTER
Contact: <sip:carla@pc.A.com>
Expires: 900
Content-Length: 0
```



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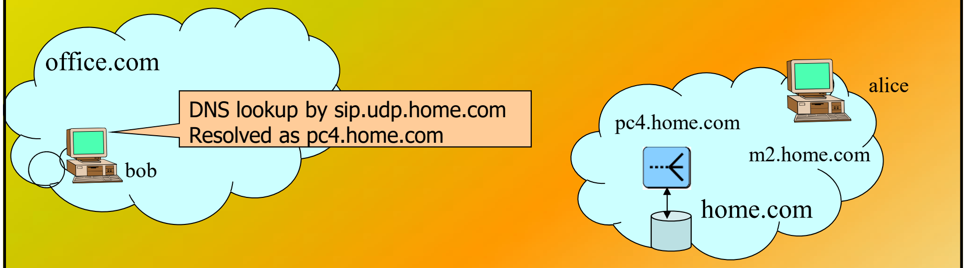
Setup of a basic call - DNS lookup

- Identifier of email type: <sip:alice@home.com>
- Alice phone registers in home.com
- Bob calls alice@home.com; phone makes DNS lookup



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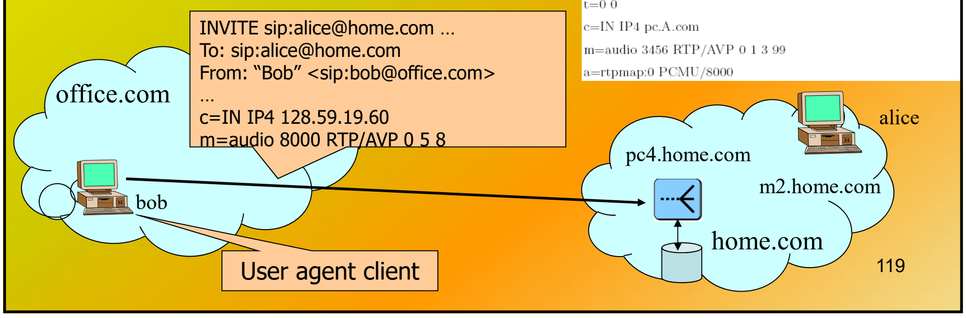
Setup of a basic call - INVITE

- Identifier of email type: <sip:alice@home.com>
- Alice phone registers in home.com
- Bob calls alice@home.com; phone makes DNS lookup
- **Phone makes an INVITE; it acts as a UAC**

```
INVITE sip:romeu@B.com SIP/2.0
Via: SIP/2.0/UDP pc.A.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Romeu <sip:romeu@B.com>
From: Carla <sip:carla@A.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Content-Type: application/sdp
Content-Length: 142
==========corpo da mensagem==============
v=0
o=carla 53655765 2353687637 IN IP4 pc.A.com
s=Session SDP
t=0 0
c=IN IP4 pc.A.com
m=audio 3456 RTP/AVP 0 1 3 99
a=rtpmap:0 PCMU/8000
```



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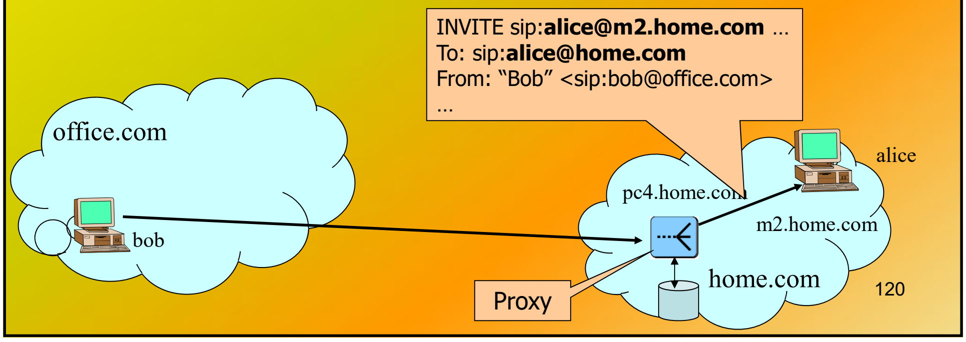
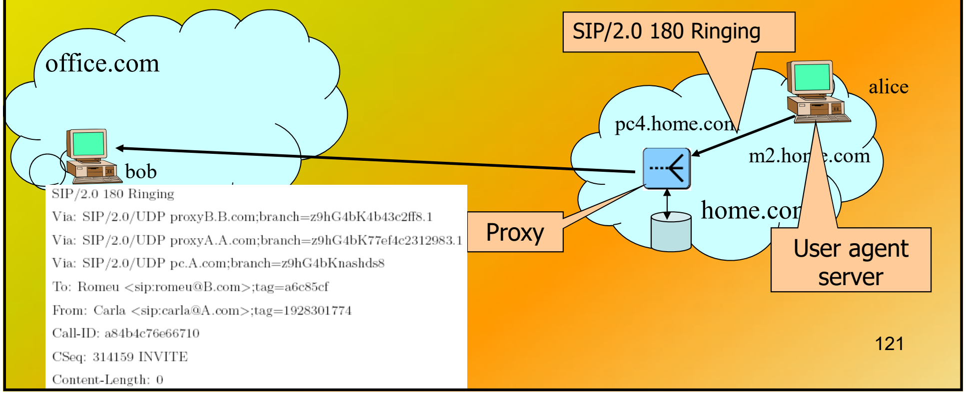
Setup of a basic call - INVITE

- Identifier of email type: <sip:alice@home.com>
- Alice phone registers in home.com
- Bob calls alice@home.com; phone makes DNS lookup
- Phone makes an INVITE; it acts as a UAC
- **Server is able to make the proxy of the call to the current location**



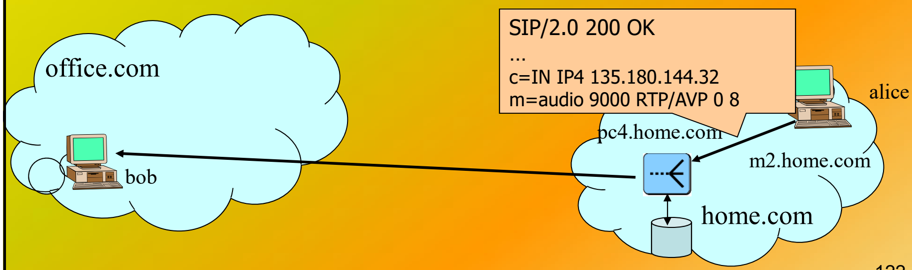
Setup of a basic call - Ringing

- Identifier of email type: <sip:alice@home.com>
- Alice phone registers in home.com
- Bob calls alice@home.com; phone makes DNS lookup
- Phone makes an INVITE; it acts as a UAC
- Server is able to make the proxy of the call to the current location
- **Alice phone rings; it acts as a UAS**



Setup of a basic call - 200 OK and ACK

- Phone makes an INVITE; it acts as a UAC
- Server is able to make the proxy of the call to the current location
- Alice phone rings; it acts as a UAS
- **When Alice hangs up the phone, the call is accepted, a 200 OK message is sent to Bob with the agreed configuration parameters, Bob phone sends a ACK to complete the setup.**



Setup of a basic call - 200 OK and ACK

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP proxyB.B.com;branch=z9hG4bK4b43c2ff8.1
Via: SIP/2.0/UDP proxyA.A.com;branch=z9hG4bK77ef4c2312983.1
Via: SIP/2.0/UDP pc.A.com;branch=z9hG4bKnashds8
To: Romeu <sip:romeu@B.com>;tag=a6c85cf
From: Carla <sip:carla@A.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Contact: <sip:romeu@pc.B.com>
Content-Type: application/sdp
Content-Length: 141
==========corpo da mensagem==============
v=0
o=carla 53655765 2353687637 IN IP4 pc.B.com
s=Session SDP
t=0 0
c=IN IP4 pc.B.com
m=audio 3456 RTP/AVP 0 1 3
a=rtpmap:0 PCMU/8000
```

```
ACK sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP pc.A.com;branch=z9hG4bKnashds9
Max-Forwards: 70
To: Romeu <sip:romeu@B.com>;tag=a6c85cf
From: Carla <sip:carla@A.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 ACK
Content-Length: 0
```

- ACK can be sent directly between UAs, since both addresses are now known

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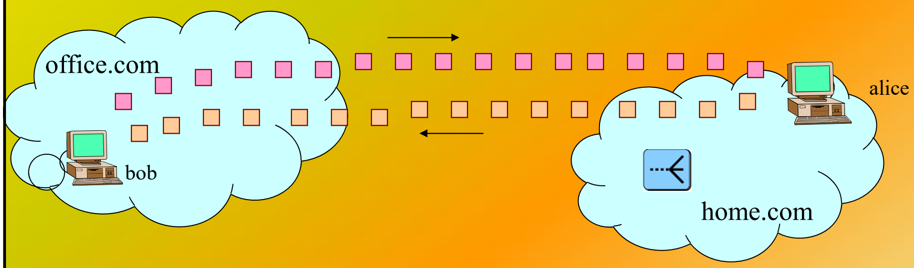
Setup of a basic call - data

- Audio packets are transferred through RTP



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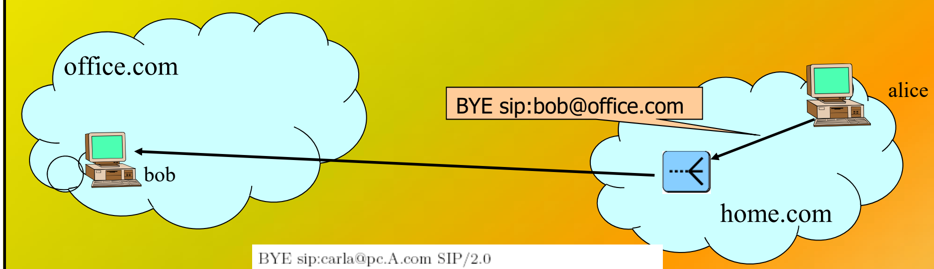
Setup of a basic call - BYE

- **When one of the endpoints hangs off, a BYE is sent**




```
BYE sip:carla@pc.A.com SIP/2.0
Via: SIP/2.0/UDP pc.A.com;branch=z9hG4bKnashds10
Max-Forwards: 70
From: Carla <sip:carla@A.com>;tag=a6c85cf
To: Romeu <sip:romeu@B.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 231 BYE
Content-Length: 0
```

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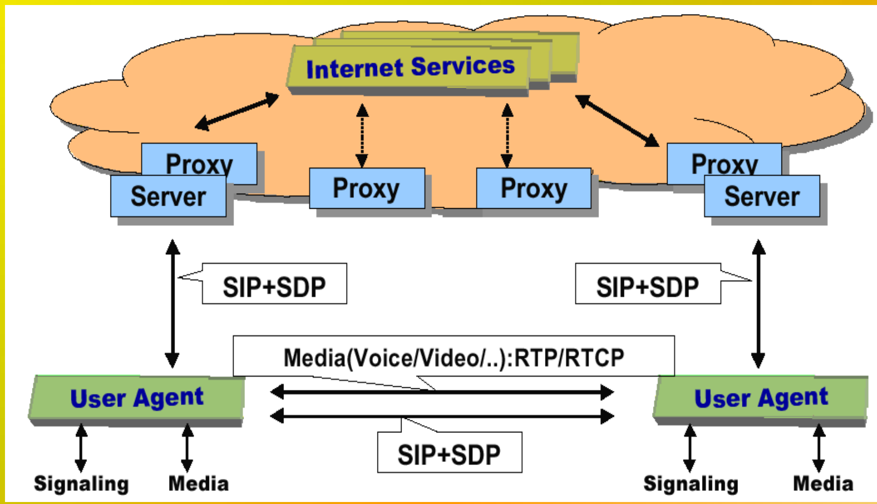
End-to-End SIP model



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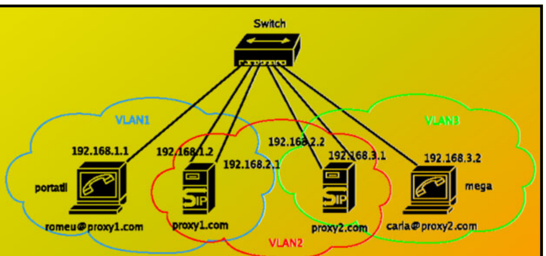
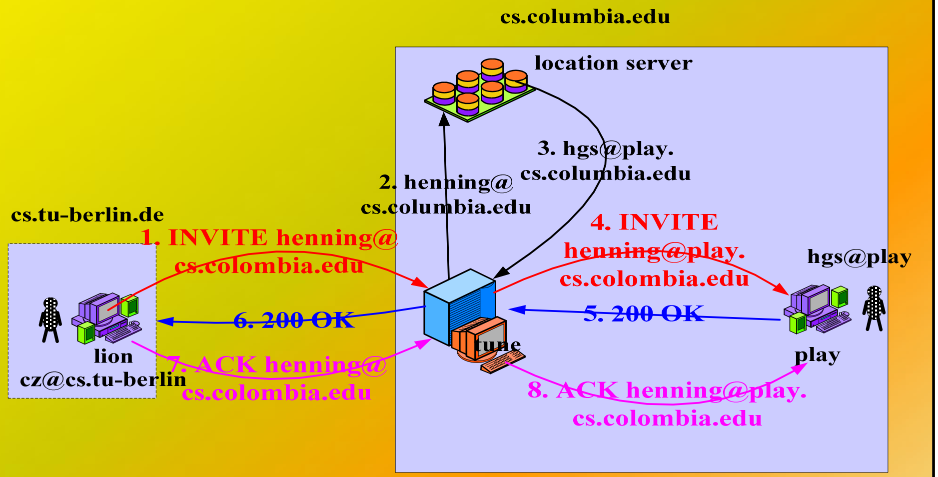
SIP Session



No.	Time	Source	Destination	Protocol	Info
1	0.000000	192.168.1.1	192.168.1.2	SIP	Request: REGISTER sip:proxy1.com
2	0.000530	192.168.1.2	192.168.1.1	SIP	Status: 401 Unauthorized (0 bindings)
3	5.121578	192.168.1.1	192.168.1.2	SIP	Request: REGISTER sip:proxy1.com
4	5.122829	192.168.1.2	192.168.1.1	SIP	Status: 200 OK (1 bindings)
5	21.65474	192.168.1.1	192.168.1.2	SIP/SDP	Request: INVITE sip:carla@proxy2.com, with session description
6	21.65531	192.168.1.2	192.168.1.1	SIP	Status: 100 trying -- your call is important to us
7	21.66426	192.168.1.2	192.168.1.1	SIP	Status: 180 Ringing
8	25.37596	192.168.1.2	192.168.1.1	SIP/SDP	Status: 200 OK, with session description
9	25.38055	192.168.1.1	192.168.1.2	SIP	Request: ACK sip:carla@192.168.3.2;transport=udp
10	25.50695	192.168.1.1	192.168.3.2	RTP	Payload type=ITU-T G.711 PCMU, SSRC=167772160, Seq=0, Time=20387
11	25.54920	192.168.1.1	192.168.3.2	RTP	Payload type=ITU-T G.711 PCMU, SSRC=167772160, Seq=1, Time=20387
12	25.54937	192.168.1.1	192.168.3.2	RTP	Payload type=ITU-T G.711 PCMU, SSRC=167772160, Seq=2, Time=20387
13	25.73593	192.168.3.2	192.168.1.1	RTP	Payload type=ITU-T G.711 PCMU, SSRC=167772160, Seq=0, Time=8174C
14	25.73595	192.168.3.2	192.168.1.1	RTP	Payload type=ITU-T G.711 PCMU, SSRC=167772160, Seq=1, Time=8174C
15	25.73601	192.168.3.2	192.168.1.1	RTP	Payload type=ITU-T G.711 PCMU, SSRC=167772160, Seq=2, Time=8174C
16	33.04344	192.168.1.2	192.168.1.1	SIP	Request: BYE sip:romeu@192.168.1.1;transport=udp
17	33.04890	192.168.1.1	192.168.1.2	SIP	Status: 200 OK

Call Proxy scenario



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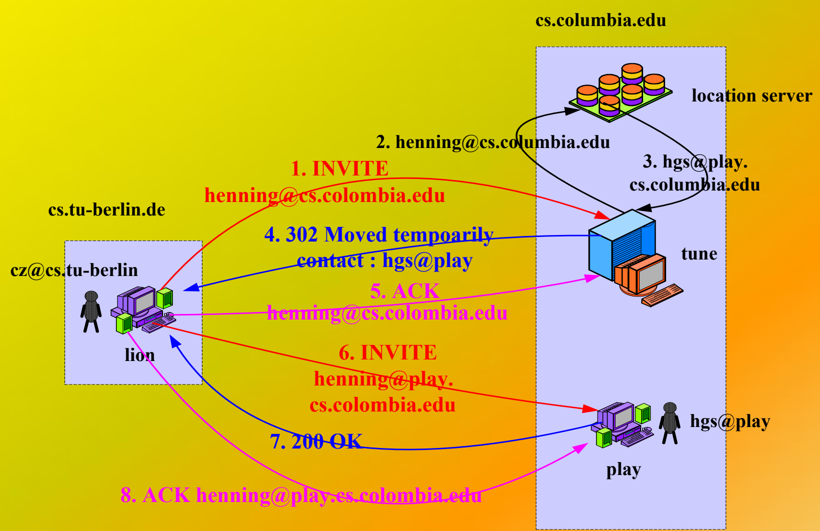
Redirect Server



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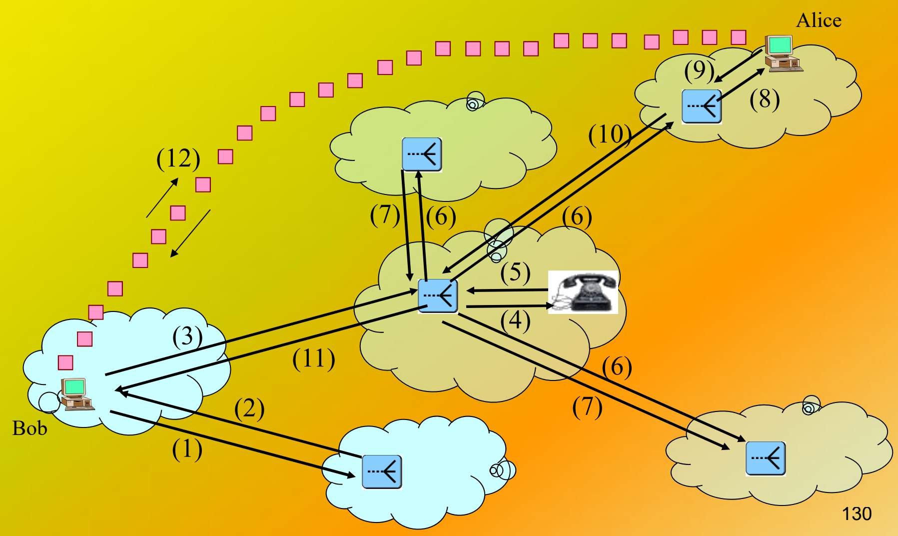
Advanced scenario

Can you build the logic?



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SIP potencialities

- Other defined facilities (generally in extensions)
  - *Presence and instant messaging* (methods of general notifications – IETF: SIMPLE WG), caller preference, callee capabilities, ...
    - Allow the unification of servers and common databases!
  - Integration of web, email, fax/video... In an unified way
  - Uses RTSP, similarly to HTTP (request-response)
- Programming of services
  - SIP-CGI, CPL, SIP-servlet
- SIP conference

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SIP Extensions

- SIP Specific Event Notification (RFC 3265)
  - SUBSCRIBE and NOTIFY messages.
  - Extensible framework by which SIP nodes can request notification from remote nodes indicating that certain events have occurred.
  - E.g. request notifications for voicemail messages waiting.
- SIP INFO Method (RFC 6086)
  - INFO message.
  - Allow for the carrying of session related control information that is generated during a session.
  - E.g DTMF tones emulation.
- SIP Extension for Event State Publication (RFC 3903)
  - PUBLISH message.
  - Allows to publish event state used within the SIP Events framework.
  - E.g. User/terminal status change (Away, Busy, etc...)

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SIP Presence and Instant Messaging

- SIP for Instant Messaging and Presence Leveraging Extensions (SIMPLE)
  - Provides for presence and buddy lists,
  - Instant Messaging in the enterprise,
  - Telephony enabled user lists.
- Presence
  - SIP-Specific Event Notification (RFC 6665).
    - SUBSCRIBE and NOTIFY methods.
  - Session Initiation Protocol (SIP) Extension for Event State Publication (RFC 3903)
    - PUBLISH mechanism.
- Instant Messaging
  - Page Mode
    - Doesn't require a session. Uses MESSAGE method (RFC 3428).
  - Session Mode
    - Message Session Relay Protocol (RFC 4975, RFC 4976).
    - Text-based protocol for exchanging content between users
    - Requires the establishment of an MSRP session.
      - Set-up using MSRP URI, within SIP and SDP signaling.

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SIP for Presence

- The SUBSCRIBE method is used to request current state and state updates/notifications from a remote node for a specific event.
  - Must contain an "Event" header field with information to identify the resource for which event notification is desired.
    - e.g., Voicemail (`Event: message-summary`).
  - Should contain an "Expires" header field indicating the duration of the subscription.
    - Unsubscribing is handled as refreshing a subscription, with the "Expires" header field set to "0".
  - May contain an "Accept" header field indicating the body formats allowed in notifications.
- The NOTIFY requests are sent to inform subscribers of changes in state (events) to which the subscriber has a subscription.
  - Does not terminate its corresponding subscription.
- 200 OK responses are used to acknowledge SUBSCRIBE and NOTIFY requests.
- The PUBLISH method is used to create, modify, and remove an event state.
  - e.g., Presence (away, busy, available, etc...) - `Event: presence`

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SIP for Instant Message (IM)

- The MESSAGE method (an extension to SIP) allows the transfer of Instant Messages (IM).
- MESSAGE requests carry the content in the form of MIME body parts.
  - Content-Type header defines content format.
- MESSAGE requests do not themselves initiate a SIP dialog.
  - May be sent in the context of a dialog initiated by some other SIP request.

```
Session Initiation Protocol (MESSAGE)
 Request-Line: MESSAGE sip:2001@192.168.56.102 SIP/2.0
 Message Header
  CSeq: 29 MESSAGE
  Via: SIP/2.0/UDP 192.168.56.1:5060;branch=z9hG4bK6abbfdfc-2361-e511-8e33-7824afcb1a1a;rport
  User-Agent: Ekiga/4.0.1
  From: <sip:Vieira@192.168.56.102>
  Call-ID: d0affdfc-2361-e511-8e33-7824afcb1a1a@SalAsus
  To: <sip:2001@192.168.56.102>
  Expires: 5000
  Content-Length: 5
  Content-Type: text/plain;charset=UTF-8
  Max-Forwards: 70
 Message Body
  Line-based text data: text/plain
   teste
```

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Session Description Protocol (SDP)

- Protocol used to describe multimedia sessions announcements, requests to join or other ways of starting a multimedia session
  - When initiating multimedia teleconferences, VoIP calls, streaming video, or other sessions, is required to transmit to participants media details, transport addresses, and other session description metadata.
- A multimedia session is a set of streams that is active for a period of time
- Not "exactly a protocol", but describes data used in other protocols
  - SDP is purely a format for session description.
  - SDP (RFC 2327, RFC 4566) provides a standard representation for such information, irrespective of how that information is transported.
  - SDP is intended to be general purpose so that it can be used in a wide range of network environments and applications: SIP, RTSP, H.332, PINT.
  - SIP carries (encapsulates) SDP messages.

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SDP Session Description

- An SDP session description is entirely textual.
- Consists of a number of lines of text of the form *<type>*=*<value>*
  - *<type> is* one case-significant character.
  - *<value>* is structured text whose format depends on *<type>*.
- Consists of a session-level section followed by zero or more media-level sections.
  - The session-level part starts with a "v=" line and continues to the first media-level section.
  - Each media-level section starts with an "m=" line.

- Types

Session description
v= (protocol version)
o= (originator and session identifier)
s= (session name)
i=* (session information)
u=* (URI of description)
e=* (email address)
p=* (phone number)
c=* (connection information -- not required if included in all media)
b=* (zero or more bandwidth information lines)
One or more time descriptions ("t=" and "r=" lines; see below)
z=* (time zone adjustments)
k=* (encryption key)
a=* (zero or more session attribute lines)
Zero or more media descriptions

Time description
t= (time the session is active)
r=* (zero or more repeat times)

Media description, if present
m= (media name and transport address)
i=* (media title)
c=* (connection information -- optional if included at session level)
b=* (zero or more bandwidth information lines)
k=* (encryption key)
a=* (zero or more media attribute lines)

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SDP: Session Description Protocol

- E.g:
  - v=0
  - o=g.bell 877283459 877283519 IN IP4 132.151.1.19
  - s=Come here, Watson!
  - u=http://www.ietf.org
  - e=g.bell@bell-telephone.com
  - c=IN IP4 132.151.1.19
  - b=CT:64
  - t=3086272736 0
  - k=clear:manhole cover
  - m=audio 3456 RTP/AVP 96 - media
  - a=rtpmap:96 VDVI/8000/1 - attributes
  - m=video 3458 RTP/AVP 31
  - m=application 32416 udp wb

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SIP vs H.323

- SIP comes from IETF: Borrows much of its concepts from HTTP.
- H.323 is another signaling protocol for real-time, interactive.
  - Comes from the ITU (telephony).
- SIP has a Web flavor, whereas H.323 has a telephony flavor.
- SIP is a single component. Works with RTP, but it can be combined with other protocols and services.
- H.323 is a complete, vertically integrated suite of protocols for multimedia conferencing: signaling, registration, admission control, transport and codecs.

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SIP vs H.323

- Request response based in text (HTTP-alike)
- SDP (types of media and transport addresses)
- Types of server: registrar, proxy, redirect
- Defines a minimum set and uses profiles and extensions (KISS)

- ASN.1 coding with specific coding rules
- Sub-protocols: H.245, H.225 (Q.931, RAS, RTP/RTCP), H.450.x...
- H.323 Gatekeeper
- Defines extensively the functions

-Both use RTP/RTCP through UDP/IP
  -H.323 only through UDP
- H.323 is considered “heavy-weight”, ITU-T biased

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Learning outcomes

- Understand the scope of VoIP models
- Describe RTP operation
- Understand the SIP and H.323 protocols
- Describe architectures for interconnecting POTS and the Internet.

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Interconnecting (two) Large Networks

How to interconnect PSTN and ISPs

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PSTN interoperation

- Translate audio (PCMU/PCMA)
- Translate signalling (PRI/T1,ISUP)
  - Different signalling
  - Advanced capabilities in SIP are not mapped in PSTN
- Translate identifiers (phone numbers)
- Determine transition points

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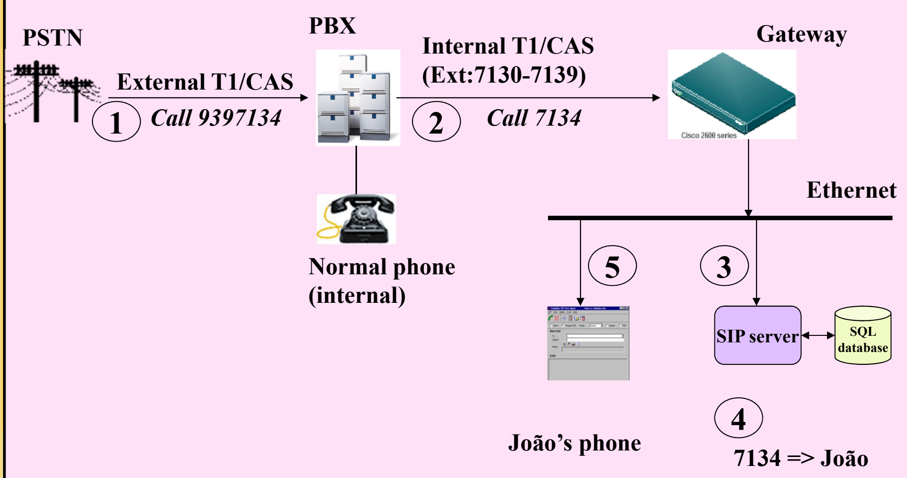
PSTN calls to IP



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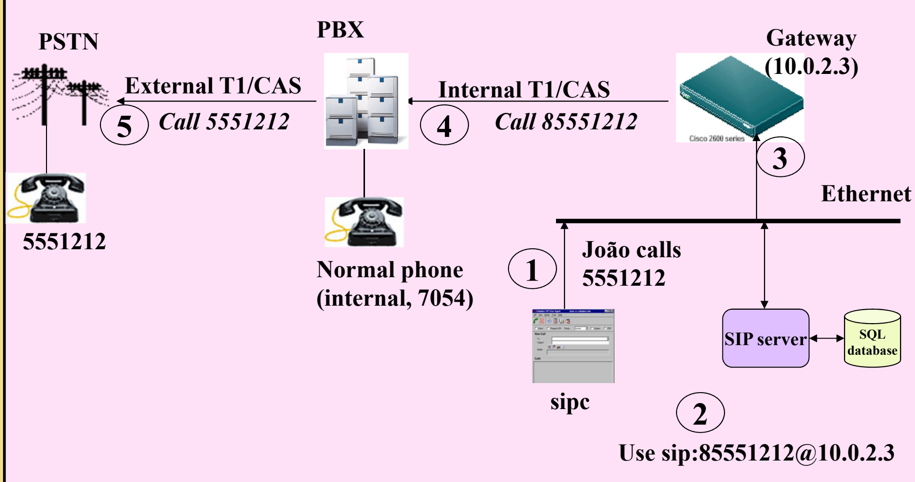
IP calls to PSTN



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ISPs and PSTN

- Having VoIP (specially voice) sessions connecting to old-style phone networks implies:
  1. Interconnecting voice signalling
  2. Interconnecting data (voice)
     - Typically this is set by routing tables in both sides
  3. Linking both inteconnection actions
  4. Selecting where to do each one of these

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What about REAL interoperation?

- Signaling boxes between the data and circuit systems must be interconnected
  - Multiple interconnection points may exist
- Systems must select best interconnection points
  - This implies best routing solution
    - And this is mixed routing – both in data and circuit systems
  - Interoperation points may be different for the data and control planes
- Different types of boxes may exist (interoperation of data/control/both)

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VoIP and PSTN Interoperability in Large Scalable Scenarios

- Requires an application programming interface and a corresponding protocol for controlling VoIP Gateways from external call control elements.
- Signaling must be inter-operable between PSTN and VoIP.
- Protocols:
  - Media Gateway Controller Protocol (MGCP) - RFC 2705
  - MGCP evolution/successor → H.248/Megaco (RFC 3015) → H.248.1/Gateway Control Protocol (RFC 3525)
    - These are control plane signaling only.
  - SIGTRAN (Signaling Transport) is the standard telephony protocol used to transport Signaling System 7 (SS7) signals over the Internet.
    - Stream Control Transmission Protocol (SCTP) – RFC 3286
      - Is an IP transport designed for transporting signaling information over an IP network.
      - Reliable transport protocol with support for framing of individual message boundaries.

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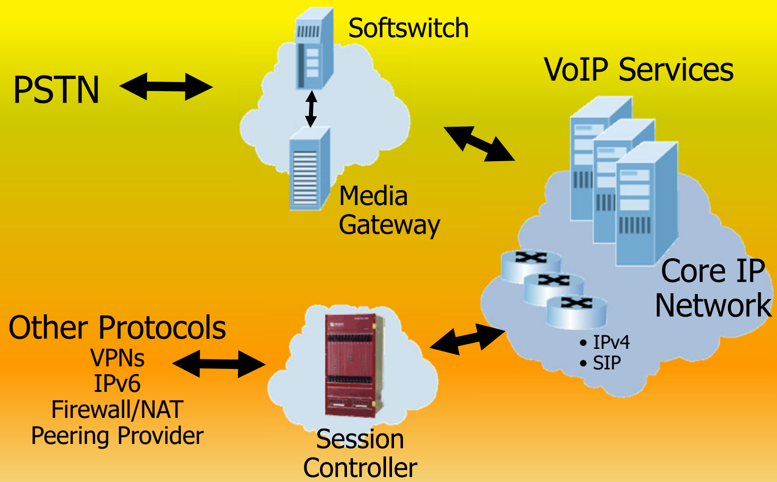
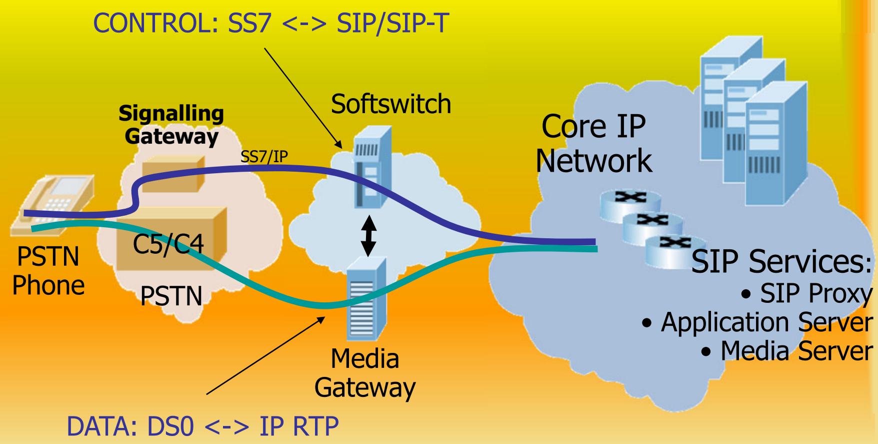
Network interoperation



Technologically addressed by media gateways and softswitches (PSTN ⇔ IP) and session controllers (IP ⇔ IP), and associated VoIP data plane protocols

PSTN/IP Interworking



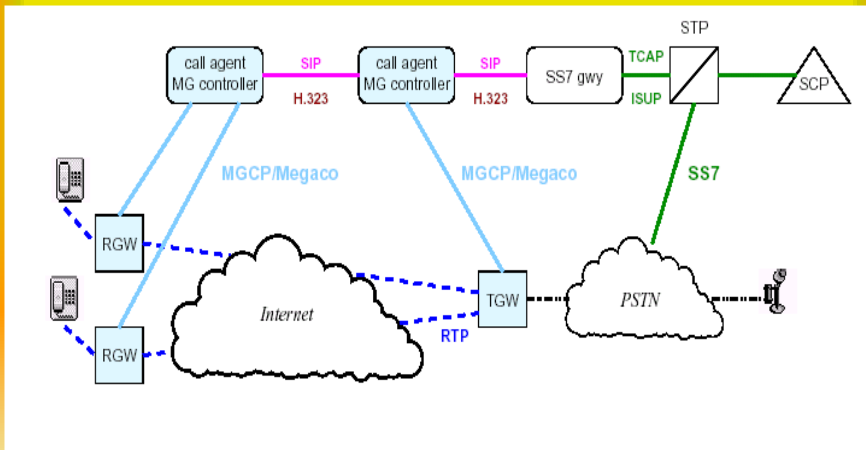
Softswitches are associated with Media Gateways (often colocated in the same box), to bridge both signalling and data.

MGCP e Megaco

- Media Gateway Controller Protocol (RFC 2705)
- Controls phone Gateways resorting to *external* control elements, the media gateway controllers (MGC) a.k.a. call agents
  - **Gateways: Eg: RGW (residential gateway): physical interconnection between VoIP networks and phone interfaces at homes**
  - **The call control "intelligence" is outside the gateways, and is controlled by external elements**
  - **master-slave philosophy**
- Objective: scalable gateway infrastructure between PSTN and IP networks
- MGCP Successor: H.248/Megaco
- These are control plane signaling ONLY.

MGCP Architecture

Objective: VoIP large scalable implementations



RGW: Residential Gateway
TGW: Trunk Gateway

MGCP/H.248 Elements

- Media Gateway Controller (MGC)
  - Controls the parts of the call state that pertain to connection control for media channels in a MG.
- Media Gateway (MG: RGW/TGW)
  - Converts media provided in one type of network to the format required in another type of network.
  - MG could terminate bearer channels from a switched circuit network (e.g., DS0s) and media streams from a packet network (e.g., RTP streams in an IP network).
- Signaling Gateway (SG)
  - Responsible for transferring signaling messages (e.g., SS7 messages) to different protocols and transports.
    - Signaling Transport (SIGTRAN)
    - e.g., SS7 to SIGTRAN (SCTP/IP).

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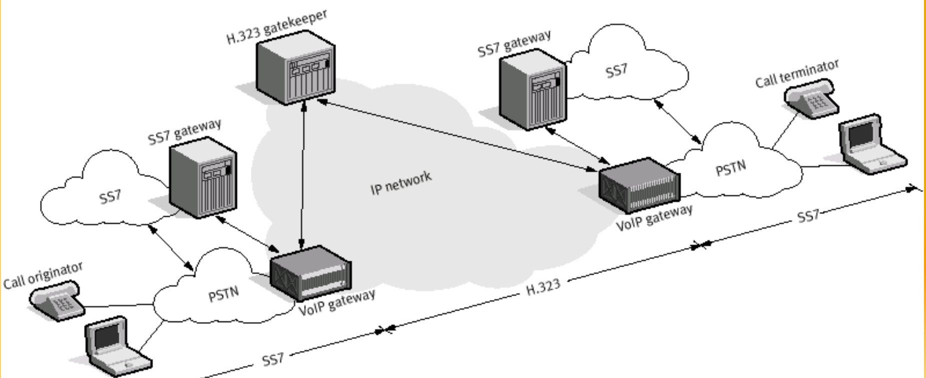
VoIP and SS7



SS7-Based VoIP Network