Media over Internet Protocol and Internet Telephony

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Contents

- Voice, Video over IP
  - RTP/RTCP
  - Packetizing voice and video, codecs
  - QoS
- Internet Telephony
  - H.323
  - SIP/SDP
  - ENUM, TRIP, CPL, etc.
  - Media Streaming, RTSP

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Two Kinds of Real-Time Multimedia

- Conversational
  - Telephony
  - Delay is critical
  - HiFi quality is not important
    (3.4 kHz speech, 300 b/w pixels in video)

- Unidirectional
  - Radio, television
  - Delay is not important
    (~5 seconds, even more in multicast is enough)
  - Emphasis on media quality (CD, HDTV)

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Why to Use IP?

- Reducing overhead
  - Speech coding, silence compression
- Utilizing commodity network technology
  - No more special network/connection for voice or multimedia
- New kind of services
  - Multimedia, multicast
  - High-quality interactive media (e.g., DAT-quality music)
- Optimal routing
  - Media is not tied to signaling path

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RTP – Real-Time Transport Protocol

- RTP/RTCP specified in RFC 1889 (Proposed Standard)
  - Draft standard is in WG last call
- Provides means for application to restore packet sequence and timing:
  - sequence number, timestamp, source identifier
- Runs on top of any transport protocol
  - UDP, TCP, AAL5
  - "Real-time" does not mean "fast"
- Lightweight, designed according ALF, ILP
  - Application integrates RTP implementation
- Designed for multicast
  - "unicast is special case of multicast"

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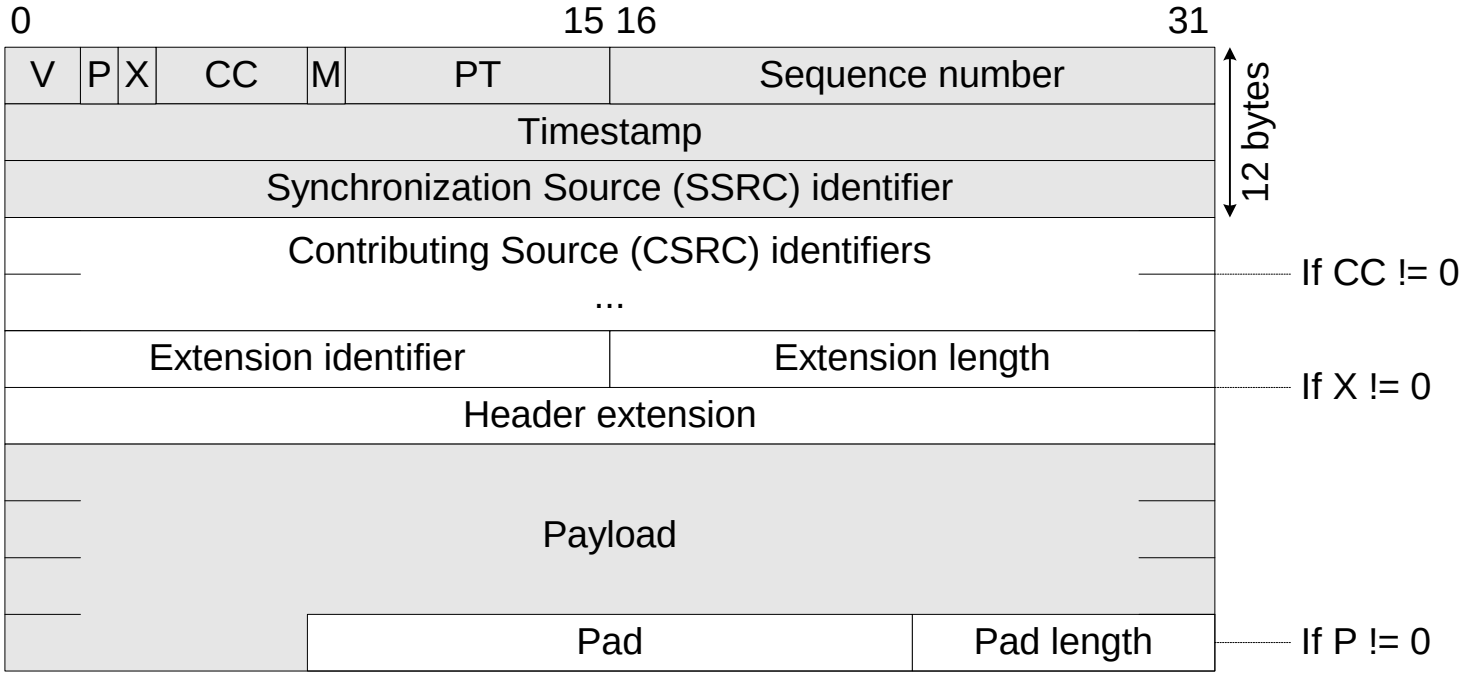
RTP Packet



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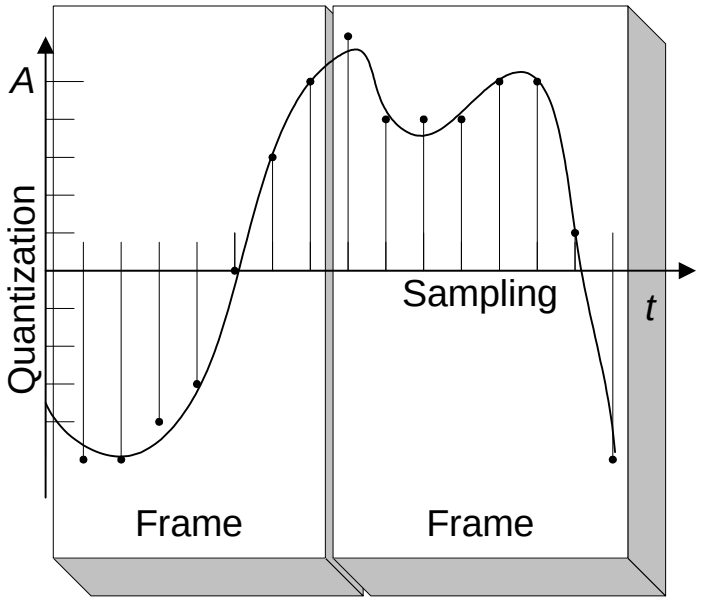
Putting Speech into Packets

Sender:

- A/D conversion
- Collecting samples into a frame
- Coding or compressing frame
- Sending frame(s) within a packet

Receiver:

- Reordering and synchronizing packets
- Decoding or decompressing frames
- D/A conversion



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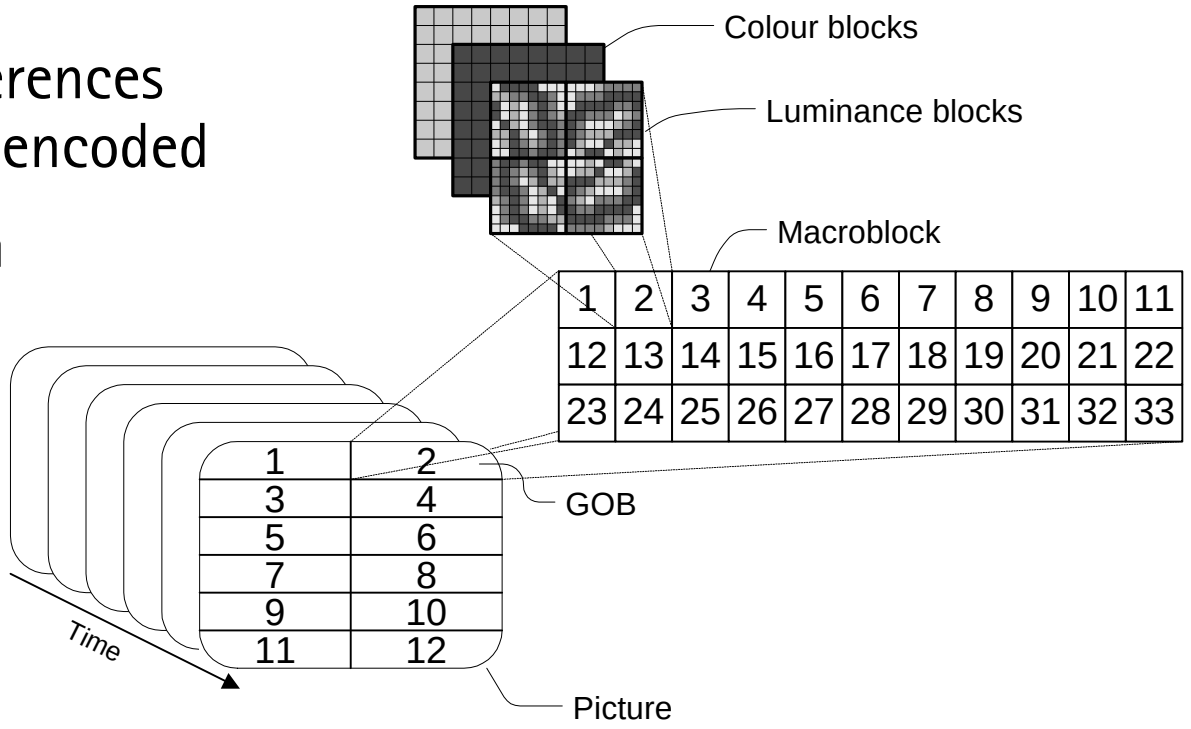
Video in Packets

- Video is stream of pictures
- Picture divided into blocks
- Blocks encoded using discrete cosine transform (like JPEG)
- INTRA/INTER mode:
  - INTER mode: differences between pictures encoded
  - Motion prediction



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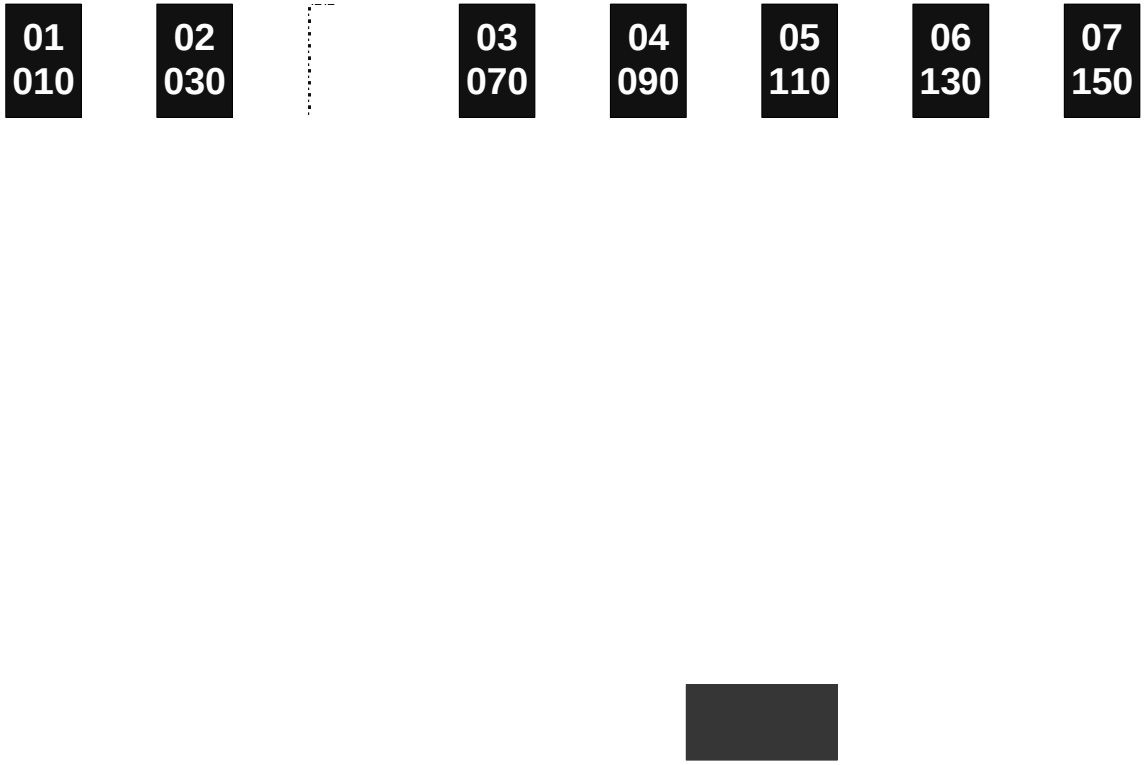
RTP in Action



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RTCP - Real-Time Control Protocol

- Designed for loosely coupled multimedia multicast session
  - Little use for point-to-point monomedia
- Packet types: SR, RR, SDES, BYE, APP
- SR: Sender report, RR: receiver report
  - Means for congestion control, calculating packet loss, jitter,
  - Synchronization between streams
- SDES: Source description
- BYE: Participant is departing conference
- APP: Application-specific packets
  - e.g., NACK

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Audio-Video Profile

- RFC 1890 - RTP Profile for Audio and Video Conferences with Minimal Control
- Profile defines guidelines for
  - timestamp rate
  - packetization
  - marker bit usage
- Packetization for different codecs:
  - G.711A, G.711μ, L8, L16, DVI4, GSM FR, 1016, G.722, G.728, H.261, nv
  - More to come:
    GSM-EFR, AMR, G.723.1, G.729, QCELP, H.263, JPEG, MPEG4, etc.

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End-to-End Quality of VoIP

- Audio codecs
  - Different algorithms:
    - PCM, ADPCM, RP-LPT, LELP, CELP, jne.
  - Lower bitrate
    ⇒lower speech quality
    ⇒more MIPS, longer delay
- Delay (conversational: ...140 ms ... 400 ms)
  - algorithmic, coding, packetization
  - network
  - synchronization, decoding
- Packet erasure (...2% ... 20 %)
  - lost in network
  - dropped due to excessive delay

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Delay Budget

- Example:
  AMR 12.2 codec, 2 frames in packet, 64 kb/s ISDN connections and inter-continental link

Algorithmic	25 ms
Encoding	5 ms
Packetization	20 ms
Network (ISDN)	14 ms
Network (Core)	80 ms - 100 ms
Network (ISDN)	14 ms
Synchronization	20 ms - 40 ms
Decoding	+ 2 ms
Total	200 ms

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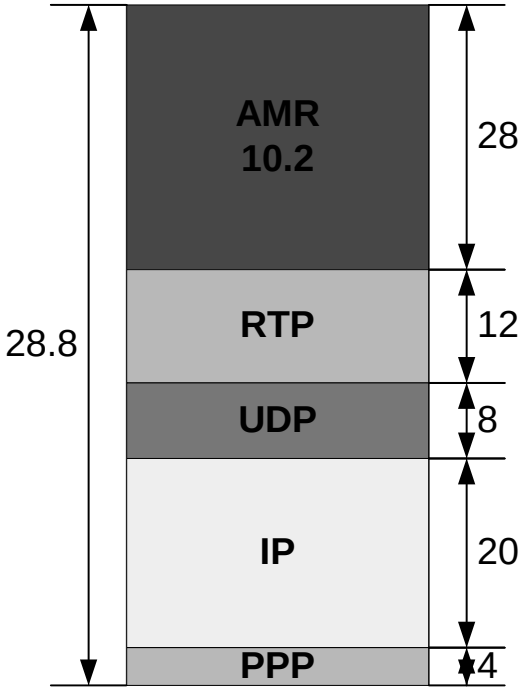
Header Overhead



- Packetization,
  RTP/UDP/IP/LL headers
- Example:
  - AMR 10.2 kb/s mode,
    50 frames/s,
    25.5 octets/frame
  - Packetization adds 2.5 octets,
    RTP 12, UDP 8, IP 20, PPP 4-5
  - Total bandwidth: 28.8 kb/s
- Putting more frames in one packet
  - multiplies packetization delay
  - even worse packet loss degradation

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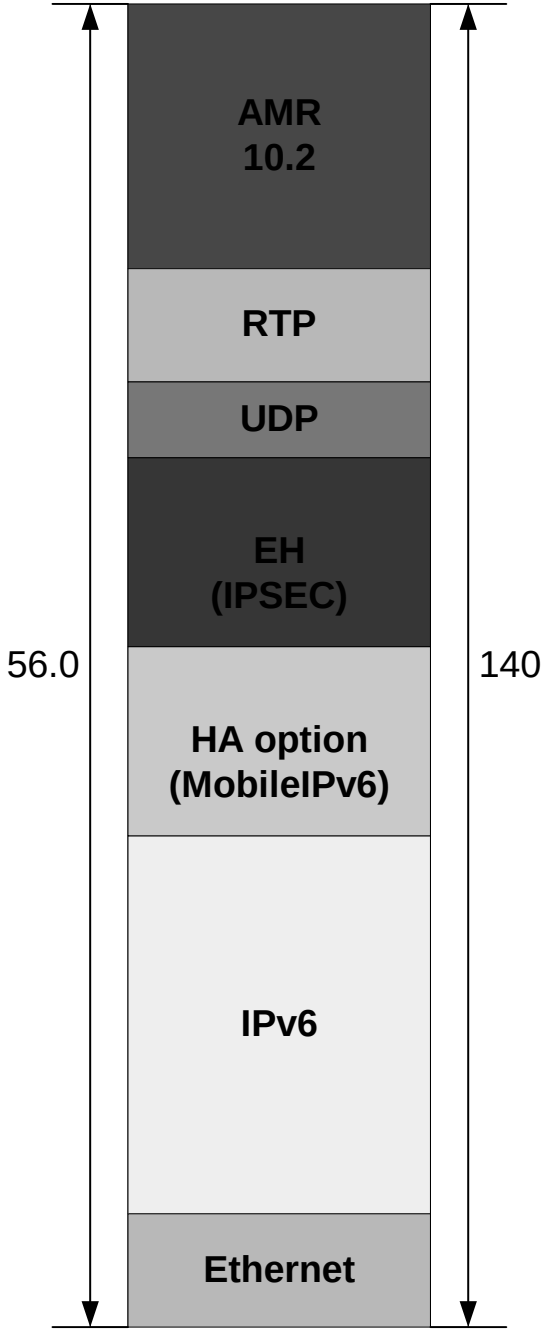

... and more

- RTP +
  UDP +
  IPSEC +
  Mobile IP +
  IPv6 +
  IEEE802.3 + ...

- More than 80% of packet is headers
- CRTP reduces header overhead to 2..4 octets in the best case
- (...but not with IPSEC)

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VoIP and Network Performance

- Processing, queueing, propagation delay
- VoIP has large packet overhead
  - "Lot of little packets full of headers"
  - header overhead causes relatively long propagation delay
  - high number of packets increase load on routers, longer processing delay
- Real-time requirements
  - by-pass long queue, use short queue
  - packets may be lost already on low load
- IntServ, DiffServ, MPLS

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VoIP on Wireless Networks

- Wireless links differ from wireline
- Lot of bit errors (1E-3 — 1E-4)
  - Error correction codes
- Bandwidth varies
  - Bad reception requires more error correction, leaving less bandwidth for data
  - Codecs with variable bit rates (AMR)
- Really narrowband links
  - Header overhead is too much, but error rate causes problem for normal header compression
  - Robust header compression, ROHC

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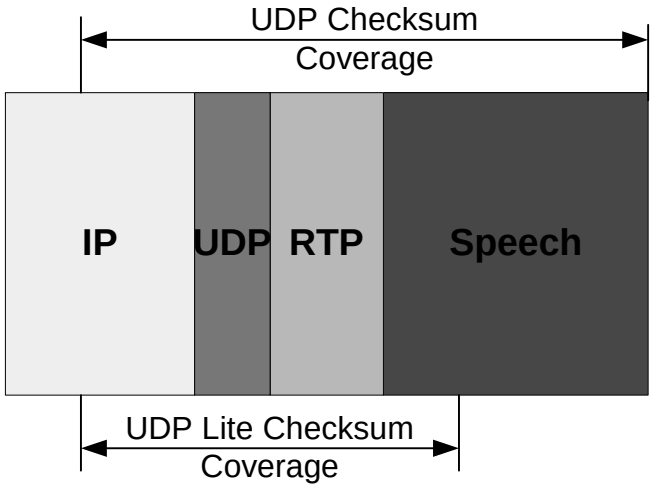
UDP Lite

- One bit error causes packet loss
- Media codecs tolerate bit errors better than packet loss

$\Rightarrow$

- Use unequal error protection
- Ignore some bit errors, detect some



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Internet Telephony

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H.323 Suite from ITU-T

- Based on Q.931, H.320, H.324
- "Videophone over TCP/IP LAN"
- Follows classic telephony model
  - Binary encoding (Q.931, ASN.1/PER)
  - Predefined services
  - Lot of baggage
- Entities: terminal, gateway, gatekeeper
- Designed by a committee
- Too complex but still missing features

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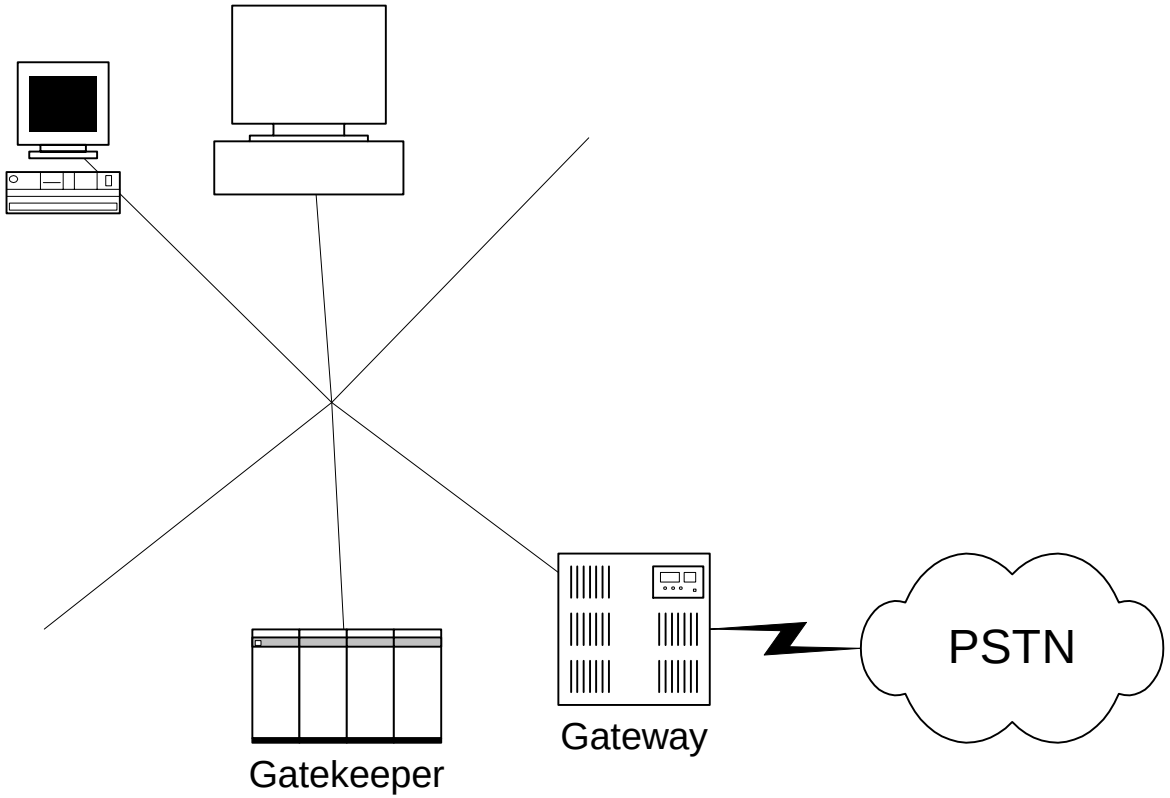
H.323 Components



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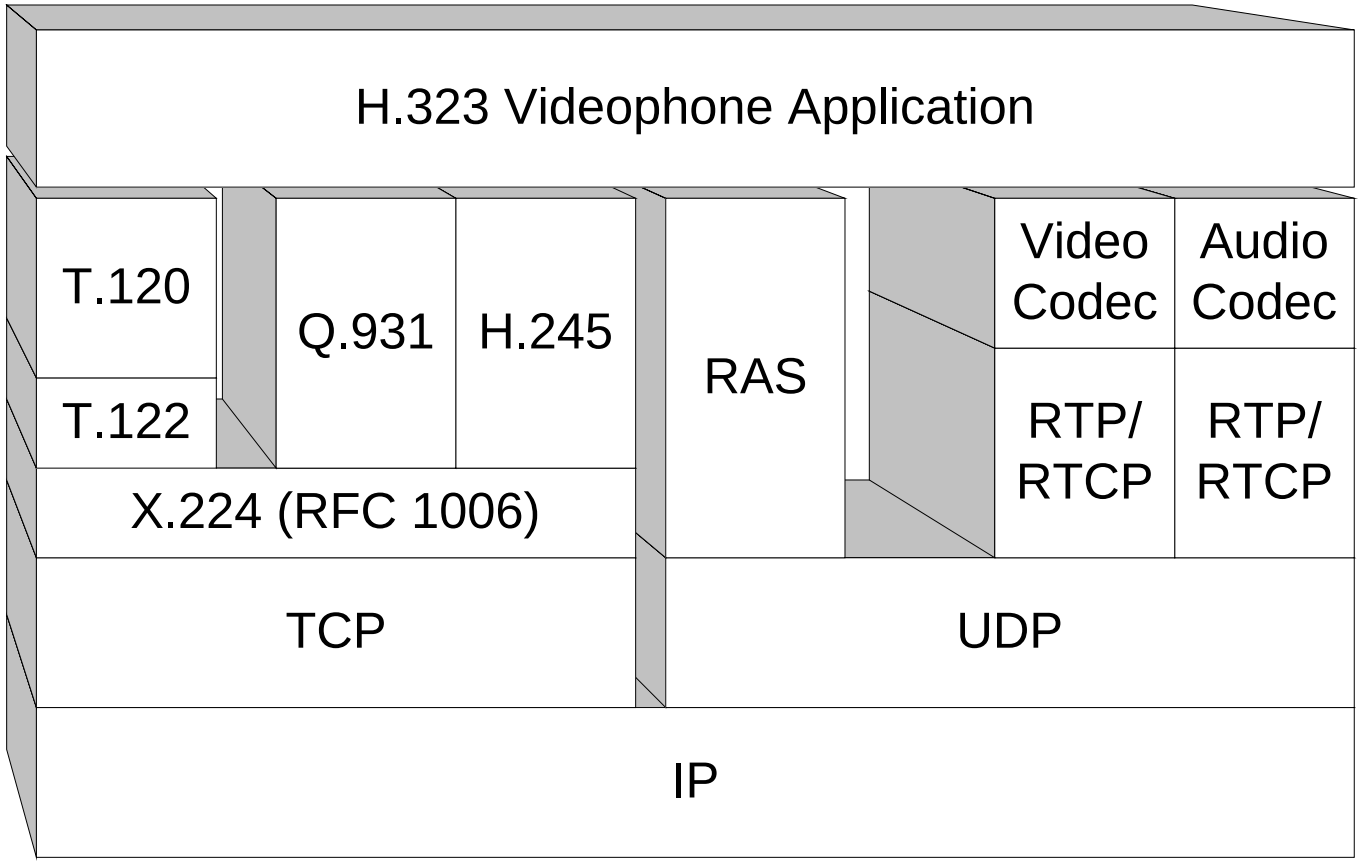
Basic H.323 Protocol Stack



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T.120

- Protocol suite for multiparty conferencing
  - Application sharing (T.128)
  - Whiteboarding (T.126)
  - Text chat (T.134, T.140)
  - File sharing (T.127)
- Transport-independent
  - Modem, ISDN, Internet (T.123)
  - Uses its own multipoint servers (T.125)
- E.g., Netmeeting

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SIP from IETF

- Based on SAP, RTSP, HTTP
  - Multicast multimedia conferencing
  - Original scope was inviting individuals to multicast conferences (Session Invitation Protocol)
  - Evolved to set up unicast and person-to-person calls (Session Initiation Protocol)
- "HTTP over UDP"
- Simple, lightweight, flexible
- RFC2543, draft standard in progress

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SIP in Nutshell

- Modelled after HTTP/1.1:
  - Plain text headers
  - Transaction-based - request-response
  - Redirections
  - Addressing using URLs
  - Authentication model
  - Allows very simple "miniservers"
- Built-in mobility
- Transport-independent: UDP, TCP, SCTP
- Entities: user agents, proxies, registrars, redirect servers

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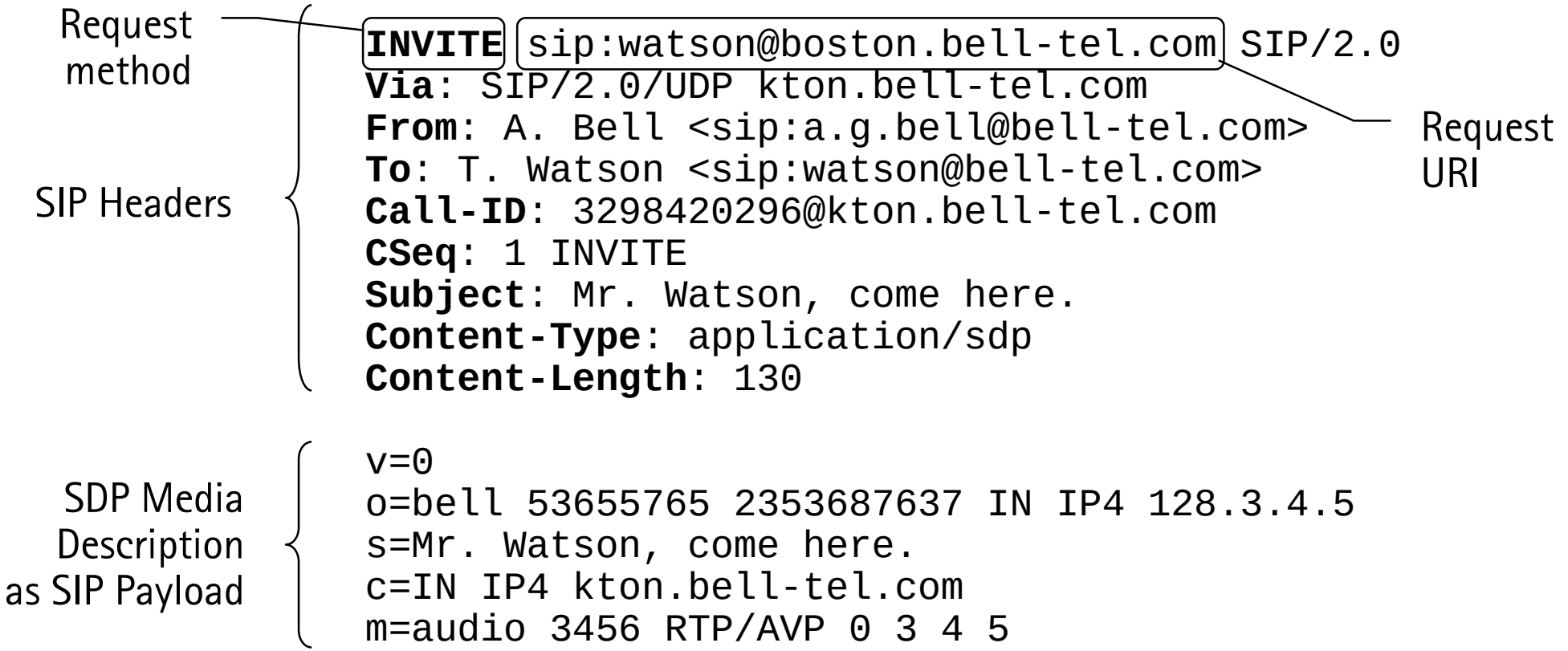
SIP Message



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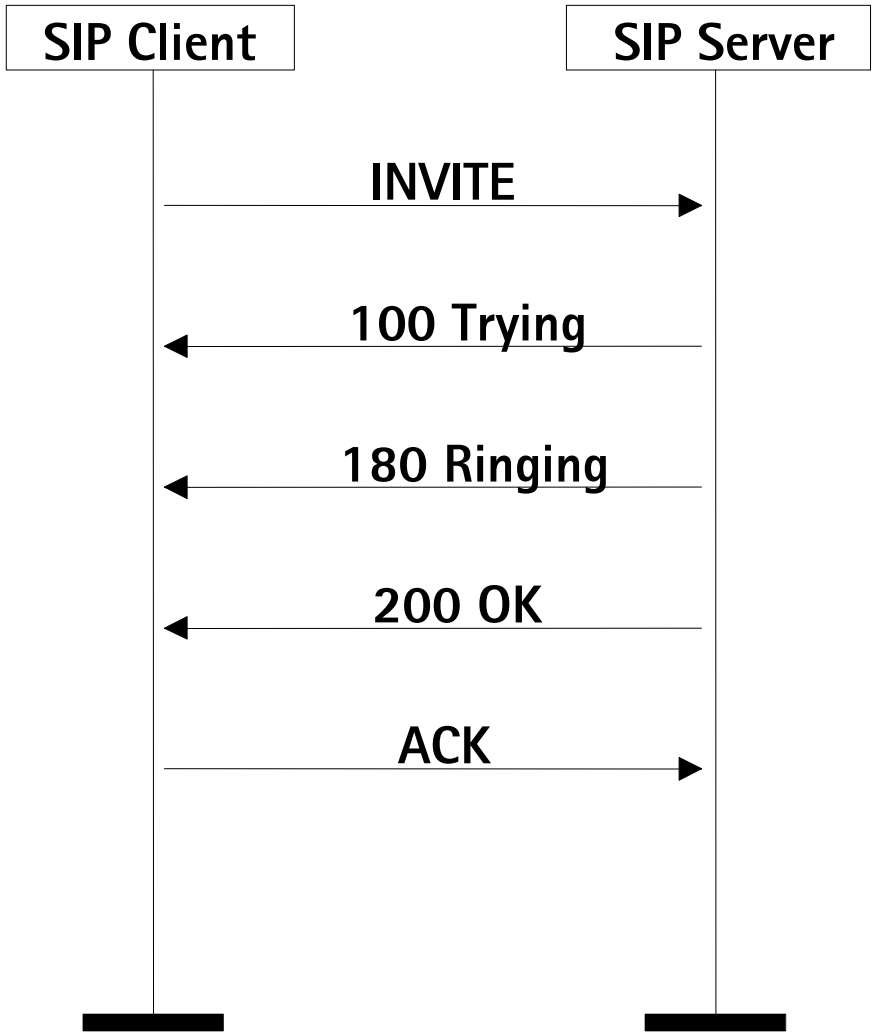
SIP Transactions

- SIP transaction consist of
  - a request sent by client to server,
  - optionally followed by provisional responses from server to client, and
  - a final response from server to client;
  - an ACK message from client to server in case of final response to INVITE



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SIP Requests

- Transaction requests
  - INVITE - call setup (also hold, resume, transfer, ...)
  - BYE - call release
  - INFO - transmit call-related information
  - OPTIONS - query UA capabilities and/or status
  - REGISTER - user contact information update
- Other requests
  - CANCEL - aborting transaction
  - ACK - acknowledge final response to INVITE

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SIP and SDP

- SIP uses SDP (RFC2327) to describe sessions
  - SDP is transmitted as payload of SIP call setup messages
  - IP addresses, port numbers, codec types, packetization, etc., for each media
- SDP is very simple
  - full implementation < 2 kLoC (H.245 is ~ 100 kLoC)
  - allows selecting audio and/or video codecs
  - but selecting individual codec modes is hard
  - may be replaced in future with, e.g., XML-based solution

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SIP User Agents

- User Agent (UA) consists of a client and a server
  - Terminal, MCU, Gateway are User Agents
- User Agent Client (UAC):
  - Initiates SIP transactions
- User Agent Server (UAS):
  - Accepts and executes SIP transactions
- Both server and client required during a call!

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SIP Proxies

Stateless proxy

- forwards SIP messages
- does not maintain transaction nor call state $\Rightarrow$ scalable

Stateful proxy

- forwards SIP transaction - keeps transaction state
- acts as server to one direction and as client to another
- can fork requests: incoming SIP request may create several virtual clients, each executing a transaction, the combined result is returned back to the original client

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SIP Redirect Servers and Registrars

Redirect Server

- accepts request and returns a list of addresses to the client
- very simple implementation: needs to understand two headers
- scalable, never maintains call state

Registrar

- server accepting REGISTER transactions
- updates location database
- usually co-located with proxy/redirect server

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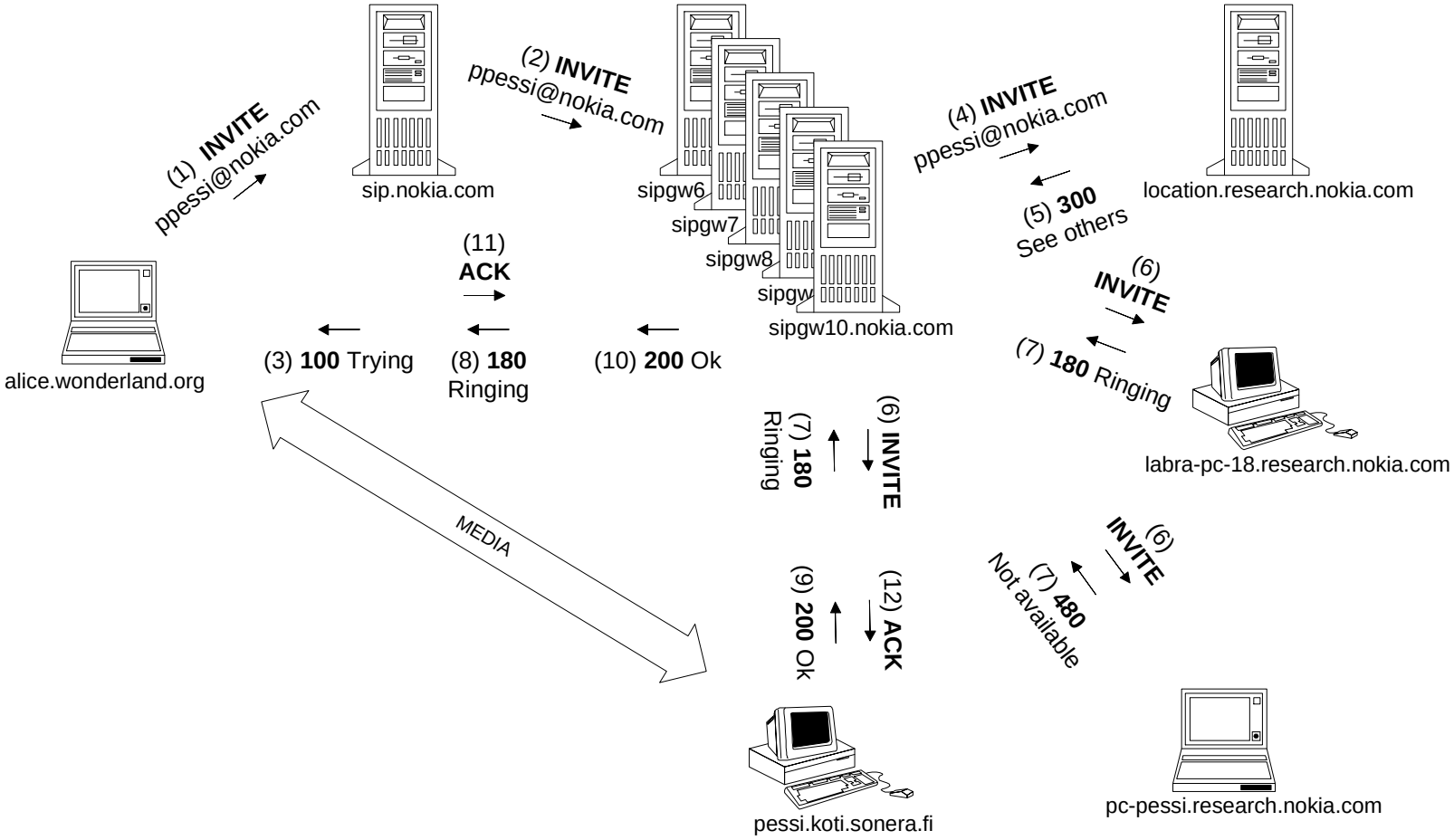
SIP Call Example



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Creating SIP Services

- SIP services can be built like WWW services
  - CGI scripts, servlets, miniservers
  - Services are created within SIP proxy network
  - No special IN servers
- Location-based services
- Active services
  - Caller and callee servlets negotiate connection
- CPL
  - XML-based rules for processing and routing calls
  - Robust, suitable for large-scale servers

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CPL – Call Processing Language

- XML-based rules for routing calls

```
<cpl>
  <subaction id="voicemail">
    <location url="sip:jones@voicemail.example.com">
      <redirect />
    </location>
  </subaction>
  <incoming>
    <location url="sip:jones@phone.example.com">
      <proxy timeout="8">
        <busy> <sub ref="voicemail" /> </busy>
        <noanswer>
          <address-switch field="origin">
            <address is="sip:boss@example.com">
              <location url="tel:+19175551212"> <proxy /> </location>
            </address>
            <otherwise><sub ref="voicemail" /></otherwise>
          </address-switch>
        </noanswer>
      </proxy>
    </location>
  </incoming>
</cpl>
```


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ENUM

- Protocol for mapping telephone numbers into URLs
  - Mapping based on service required
  - Using DNS NAPTRs
- E.g., calling to +358 9 4376 6852:
  - Converted to domain name
    `2.5.8.6.6.7.3.4.9.8.5.3.e164.arpa.`
  - `6.7.3.4.9.8.5.3.e164.arpa.` resolves to NAPTR
    `10 10 "U" "SIP+E2U" "!+(.*)!sip:+\1@research.nokia.com"`
  - Telephone number is matched to regexp, and resulting URL is used to place the call

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TRIP

- Inter-domain call routing protocol
  - Based on BGP
- Protocol-independent
  - H.323, SIP
- Routing based on policy
  - cost, quality, reliability, features

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SIMPLE

- Instant messaging and presence for SIP
  - Using SIP proxy network
  - Instant messages are delivered to user agents registered with recipient address
  - Notification events sent to presence subscribers whenever registration status (how to reach the user) changes
  - New elements: presence servers, presentity
- CPIM-compatible
- Work in progress

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Media Streaming

- SAP (RFC 2974)
  - Session source multicasts session description
  - No negotiation - take it or leave it
- RTSP (RFC 2326)
  - Client asks server for session description
  - HTTP-like transaction model
  - Media and codec negotiation
  - Multiple streams, codec conversion
  - Firewall/NAT-friendly
  - Distributed content servers

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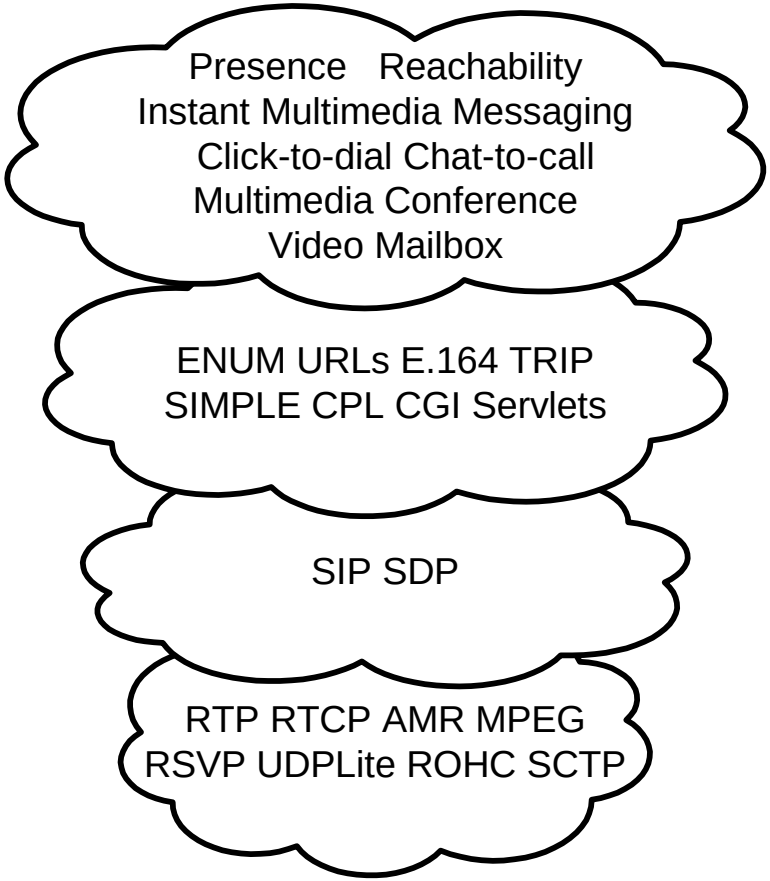
Internet Telephony



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