VoIP &SIP

Summary

- What is VoIP
- Real Time Protocol
- SIP
  - Servers, proxies, registrars
  - NAT traversal
  - mobility
  - Android support

Voice over IP (VoIP)

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Ch 18

- **Did you know that :**
  - **Most telephony today is transported with VoIP?**
  - **Most PBX-es installed use VoIP?**
  - **In Ethiopia, Oman VoIP use is a criminal offence?**
  - **Hangouts, Whatsapp, Skype, TeamSpeak, TeamViewer, Viber, Yahoo Mesg all use VoIP?**

Seven Myths About VoIP

1. VoIP is free
2. The only difference between VoIP and regular telephony is the price
3. Quality of service isn’t an issue nonadways, because there’s plenty of bandwidth in the network
4. VoIP can’t replace regular telephony, because it still can’t guarantee quality of service
5. VoIP is just another data application
6. VoIP isn’t secure
7. A Phone is a Phone is a Phone

What is VoIP?

- VoIP is an end-to-end architecture
  - Voice transported in IP packets
- Comparison with PSTN
  - Circuit switch vs. Packet switch
  - Latency
  - Dataplane, control plane
  - Mobility
- VoIP headsets
  - Physical, Software
  - Built into Android

Packet Encapsulation

- Encapsulate 10-20ms of speech in a packet

IP	UDP	RTP	Voice

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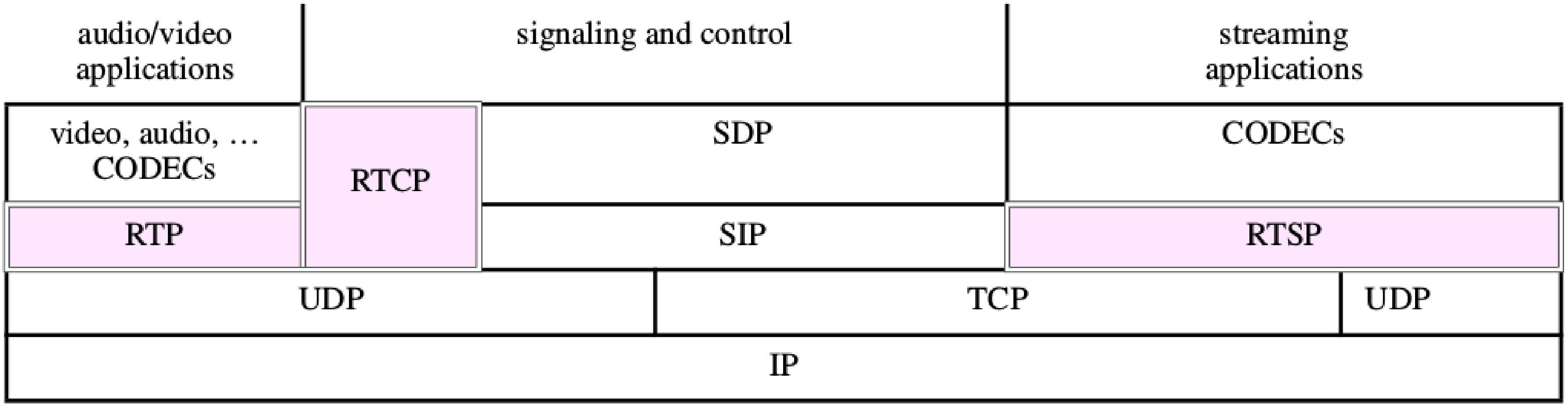
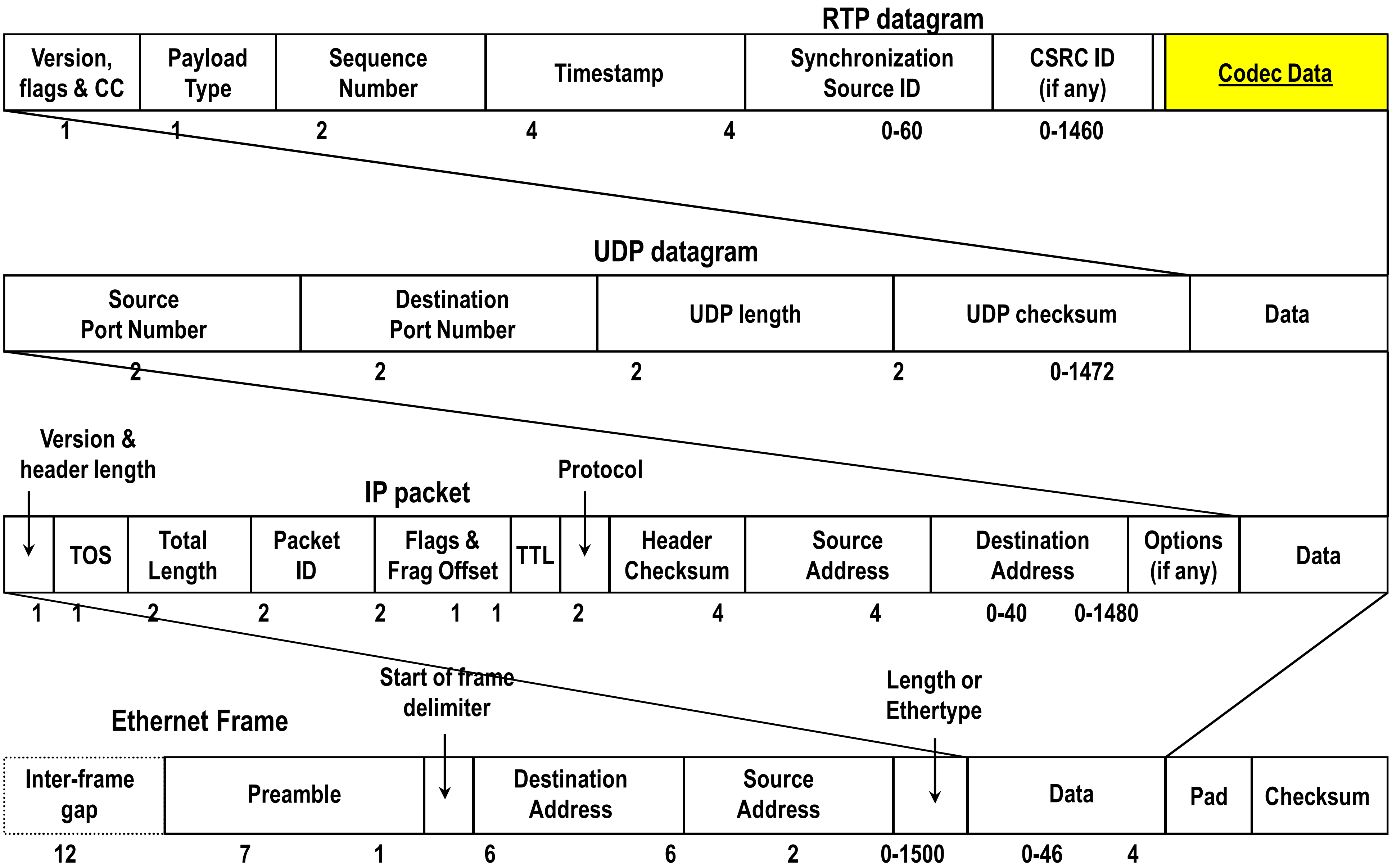
- RTP, RTCP, RTSP

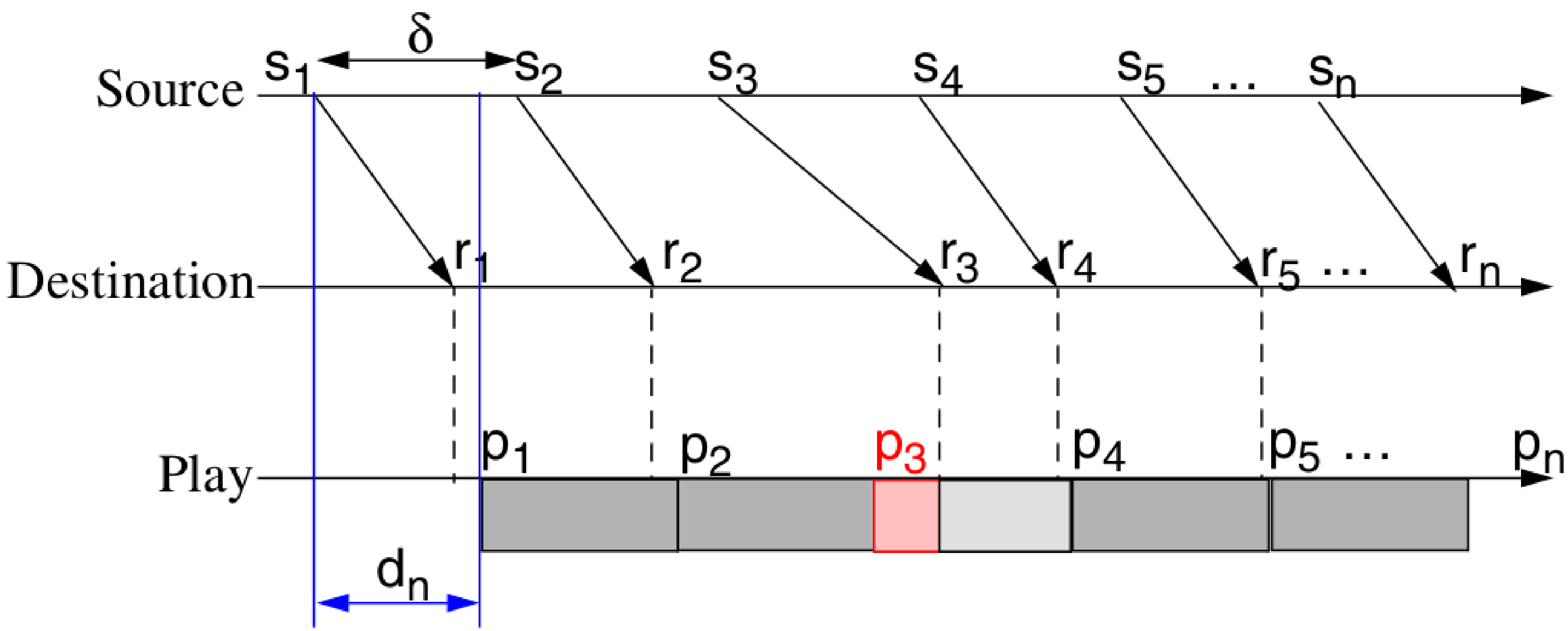


Packet Encapsulation



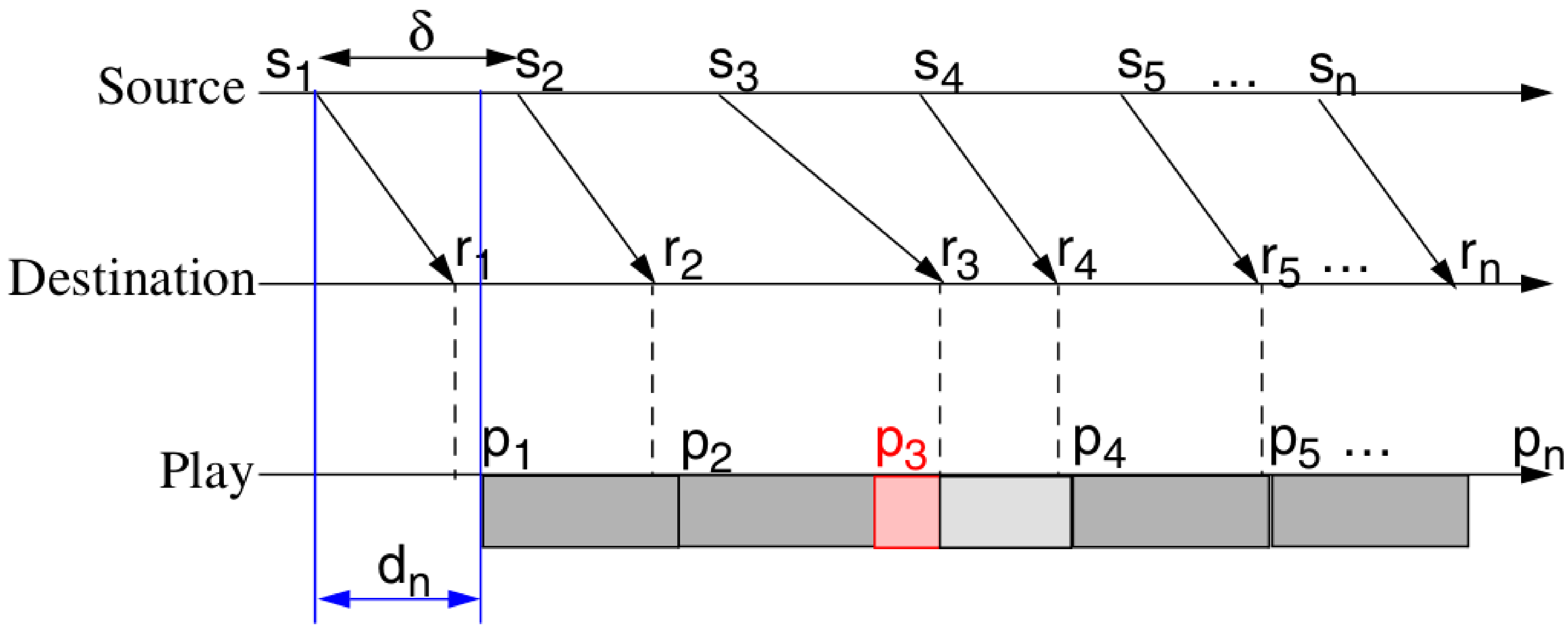
Realtime delivery

- Realtime app = maintain time relationship at receiver
  - Play in same order as original (sequence number)
  - Play time to reproduce original (time stamp)
  - Once decided p1, <u>all packets have deadlines</u>!



Jitter = $(r_i - r_{i-1}) - (s_i - s_{i-1})$

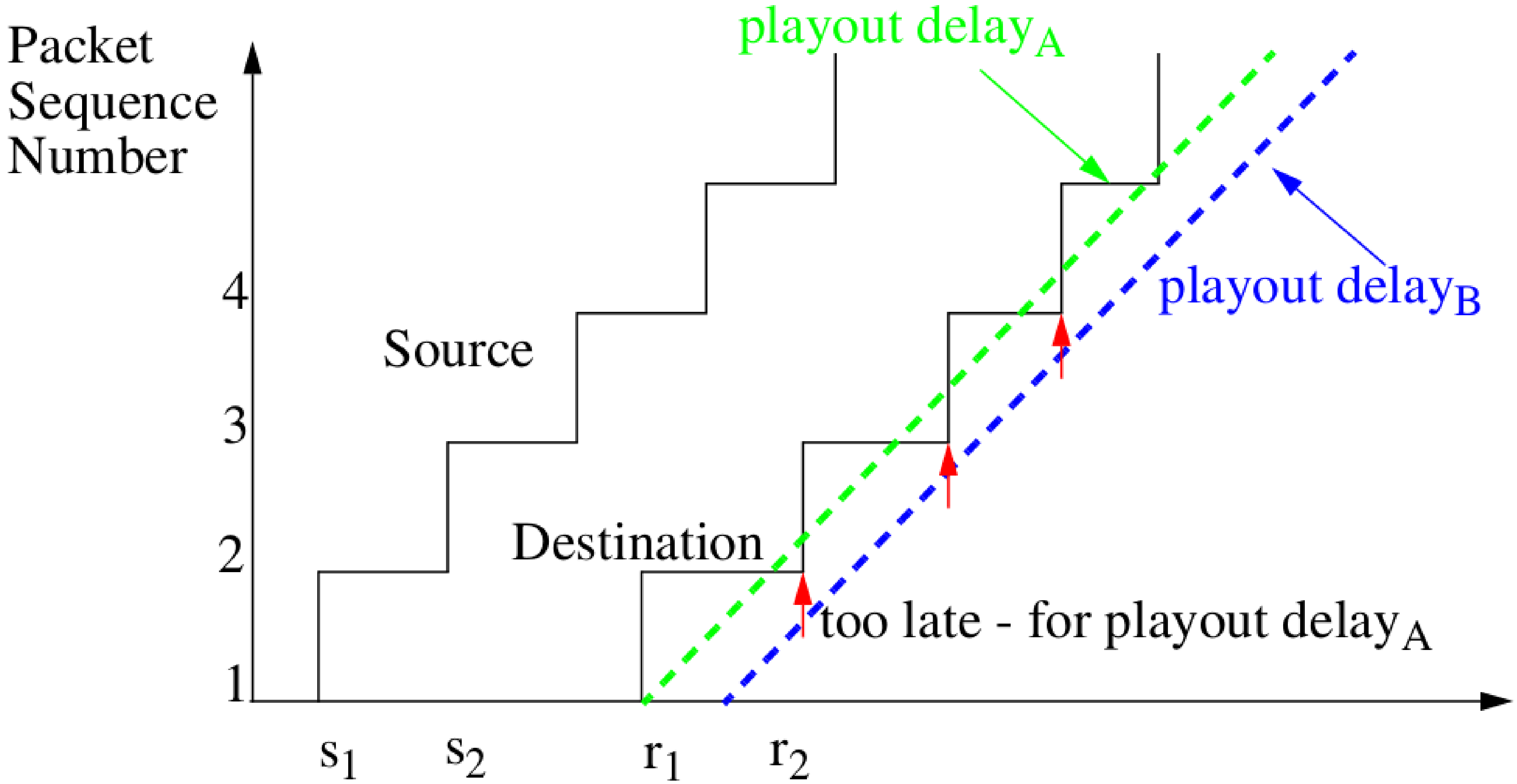
Realtime delivery



- What is jitter?
  - Packet delay variance $= (r_i - r_{i-1}) - (s_i - s_{i-1})$
  - Negative jitter: late packet
  - Positive jitter: early packet
- How to shield listener from jitter?
  - Playout buffer (extra delay)

Dealing with jitter

- **Playout buffer = delay at receiver to smooth jitter**



Delay and jitter

- Audio end-to-end delay components
  - encoding
  - packetization
  - transmission
  - propagation, switching/routing
  - receiving
  - decoding
  - dejittering
  - playing

sender

network

receiver

playout buffer ADDS delay

Voice quality metrics

- Listening quality
- Conversational quality
- Network quality
  - Delay, loss, jitter
  - Delay limits
    - < 150ms acceptable
    - < 400ms tolerable
    - > 400ms unacceptable
- Mean Opinion Score (MOS)
  - Excellent = 5, Good = 4, Fair = 3, Poor = 2
  - Functions derived using human listeners to assign MOS to a given (loss,delay) conversation[1]

[1] Cole , Rosenbluth "Voice over IP Performance Monitoring", http://ccr.sigcomm.org/archive/2001/apr01/ccr-200104-cole.pdf

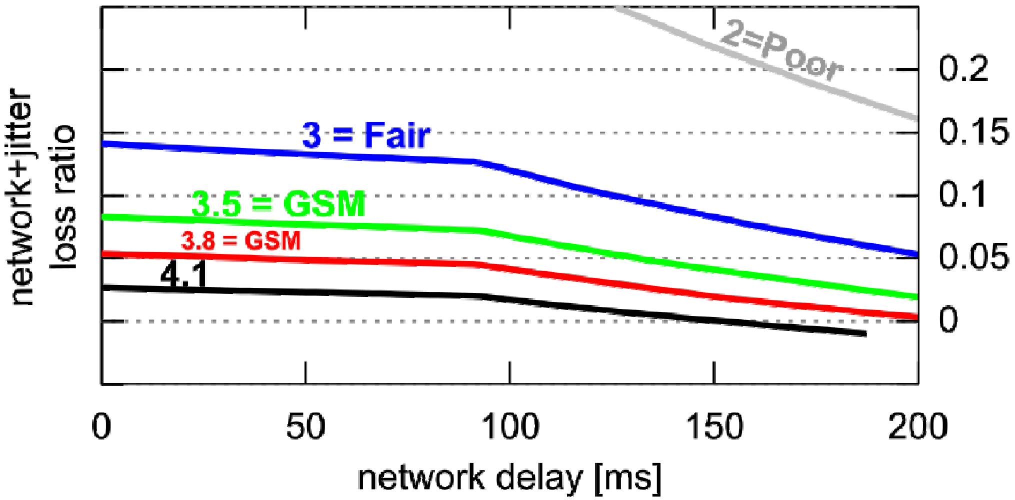
voice codecs

- **Voice <u>co</u>der-<u>dec</u>oder**
- **Needs time &CPU to signal process/compress**
- **Open or closed source**

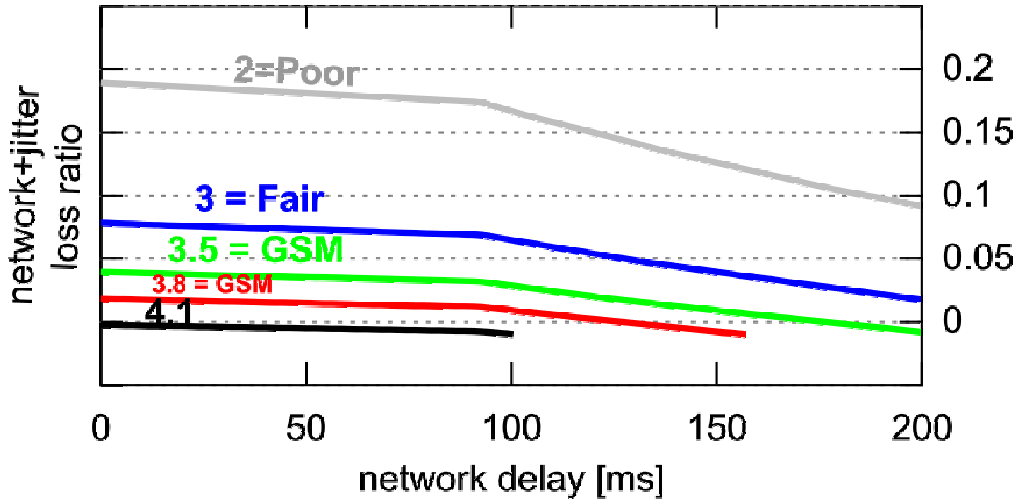
Codec	bitrate	Framesize [ms]	codec Delay[ms]	MOS ideal cond.
G.711	64kbps	10, 20,30	25	4.1
G.729	8kbps	10, 20,30	15, 25,35	3.92
GSM-FR	14kbps	22.5	20	3.5
SILK (skype)	6-40Kbps	20	?	5

MOS(delay, loss)

G.711 MOS



G.729 MOS



- Conditions: 25ms vocoder delay, 60ms playout buffer
- Used known MOS(delay,loss) functions to generate curves
- G.729 = high compression, less resilient to loss
- G.711 = needs more bandwidth, more loss resilient

Session Initiation Protocol (SIP)

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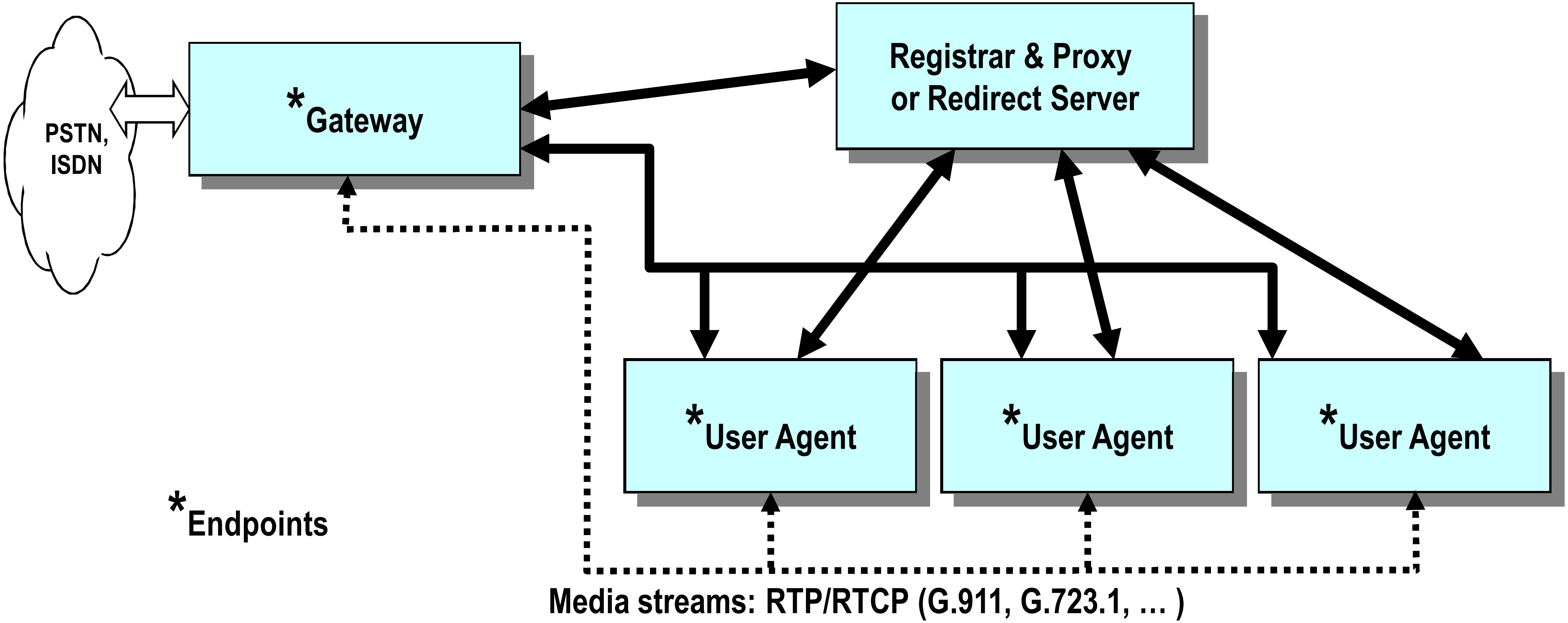
- RFC 3261
- Text based protocol, similar to HTTP, SMTP
- Locate user given email-style address
- Setup session
- negotiate call parameters
- Personal mobility:different terminal, same identifier
- SIP does not use RTP, a session does
- SDP (Session Description Protocol) – info about call, encoding, ports
- RFC 5411 - “A Hitchhiker's Guide to the Session Initiation Protocol (SIP)” – more than 100 RFCs

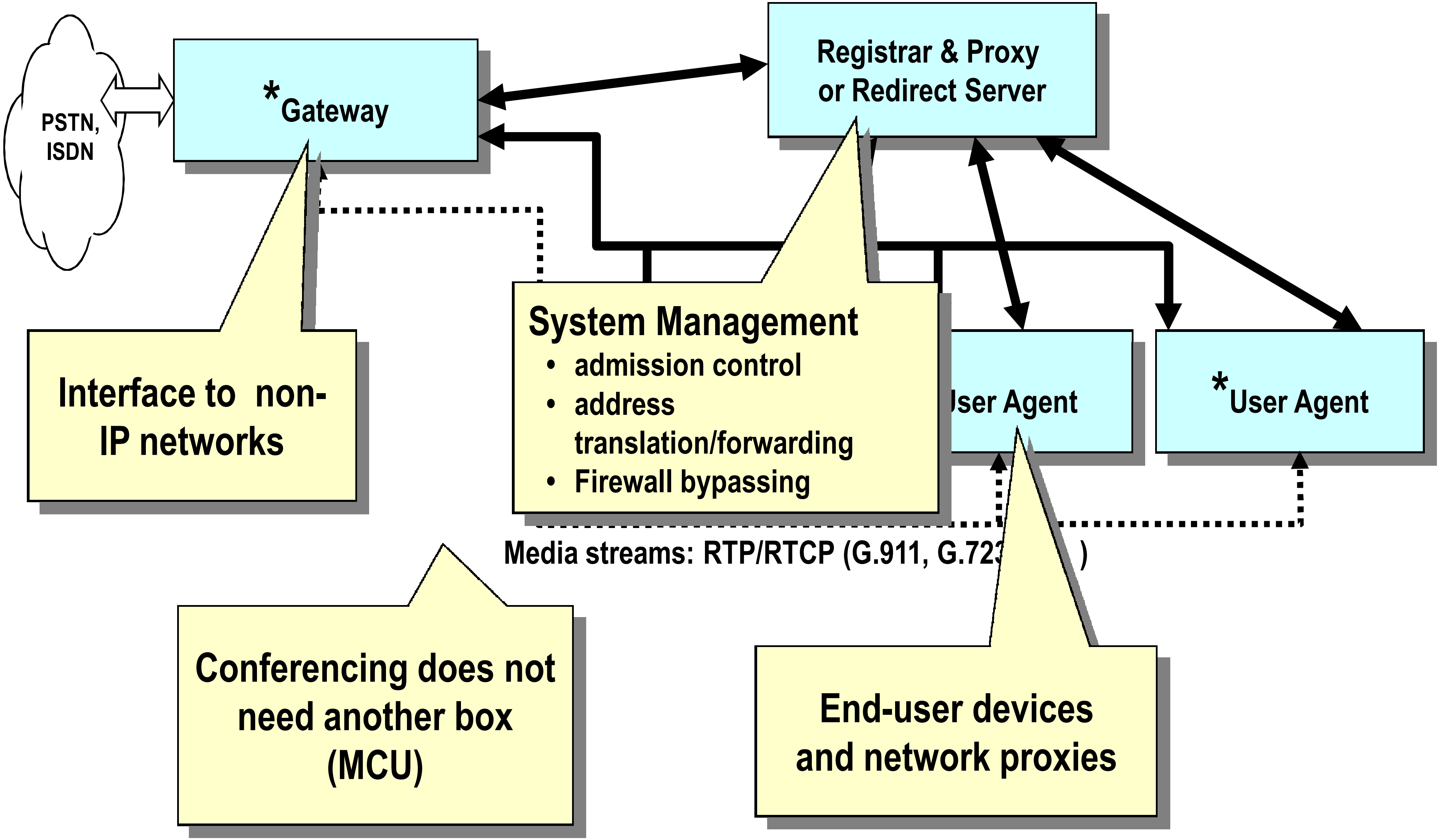
SIP components

- user-agent
  - UAC: client initiate requests
  - UAS: server generates response: accept, redirect, refuse call
- redirect server: redirect requests
- proxy server: server + client
- registrar server: track user locations
- often combined: registrar + (proxy or redirect server)
  - admission control
  - address translation/forwarding
  - NAT traversal

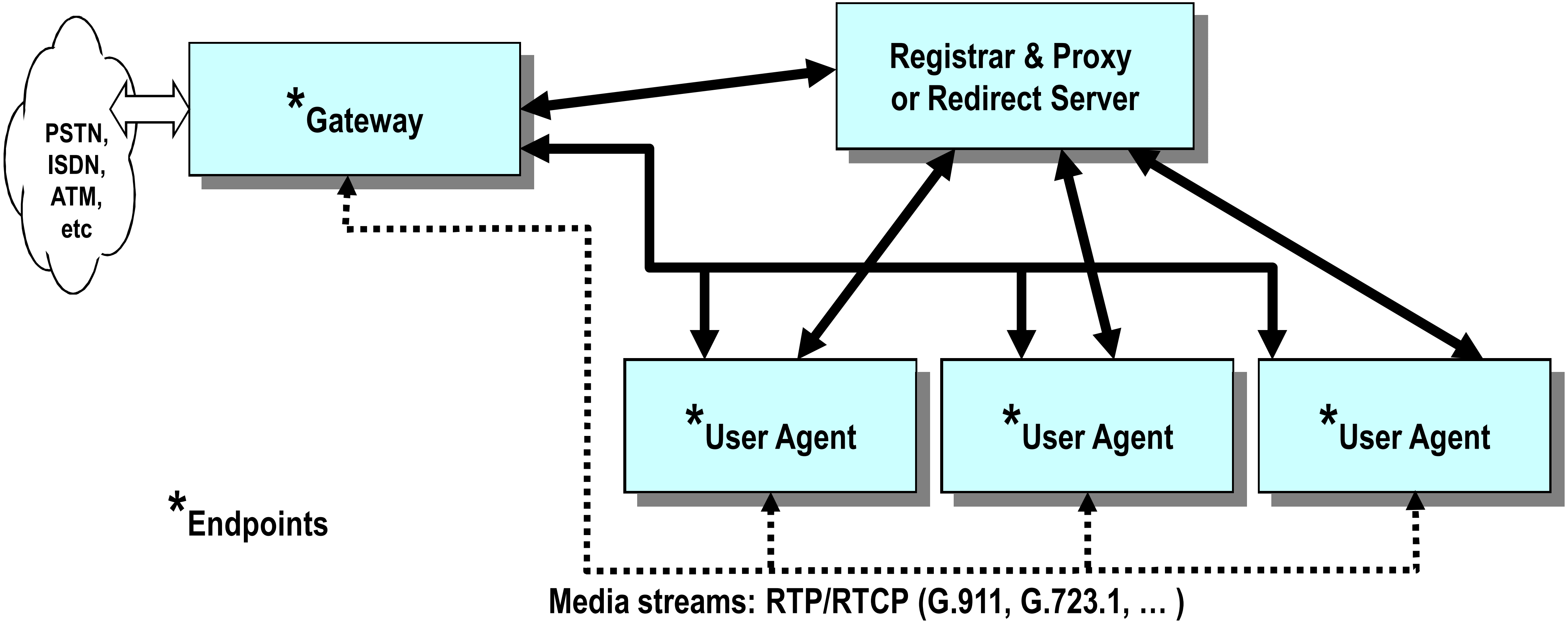
IETF SIP Architecture



IETF SIP Architecture Tour



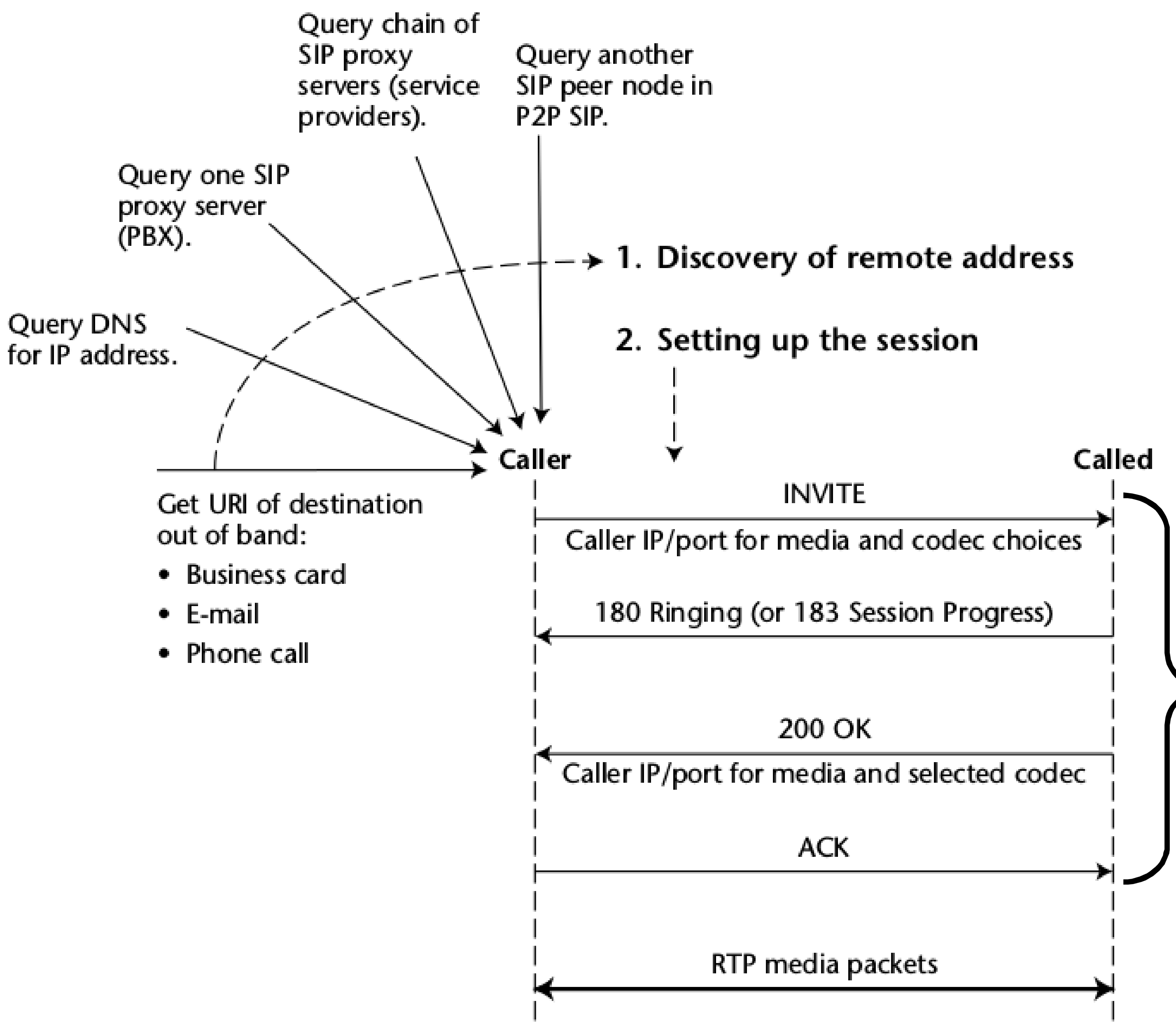
IETF SIP Architecture Tour



Components of the SIP protocol suite:

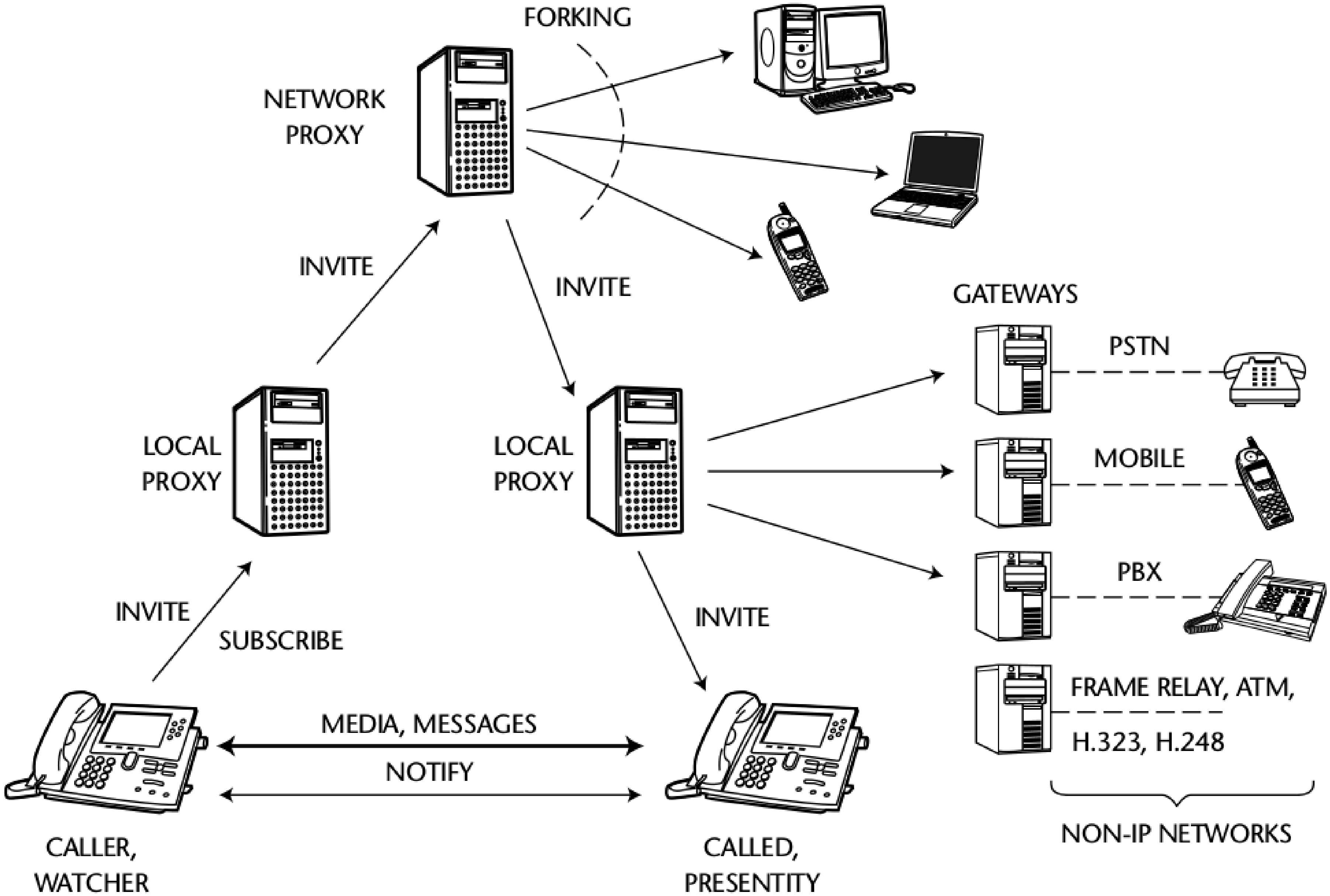
- SIP = almost all signaling, optional services, etc.
- SDP = negotiation/capabilities
- DNS = address translation
- RSVP = QoS bandwidth guarantee

SIP in a nutshell

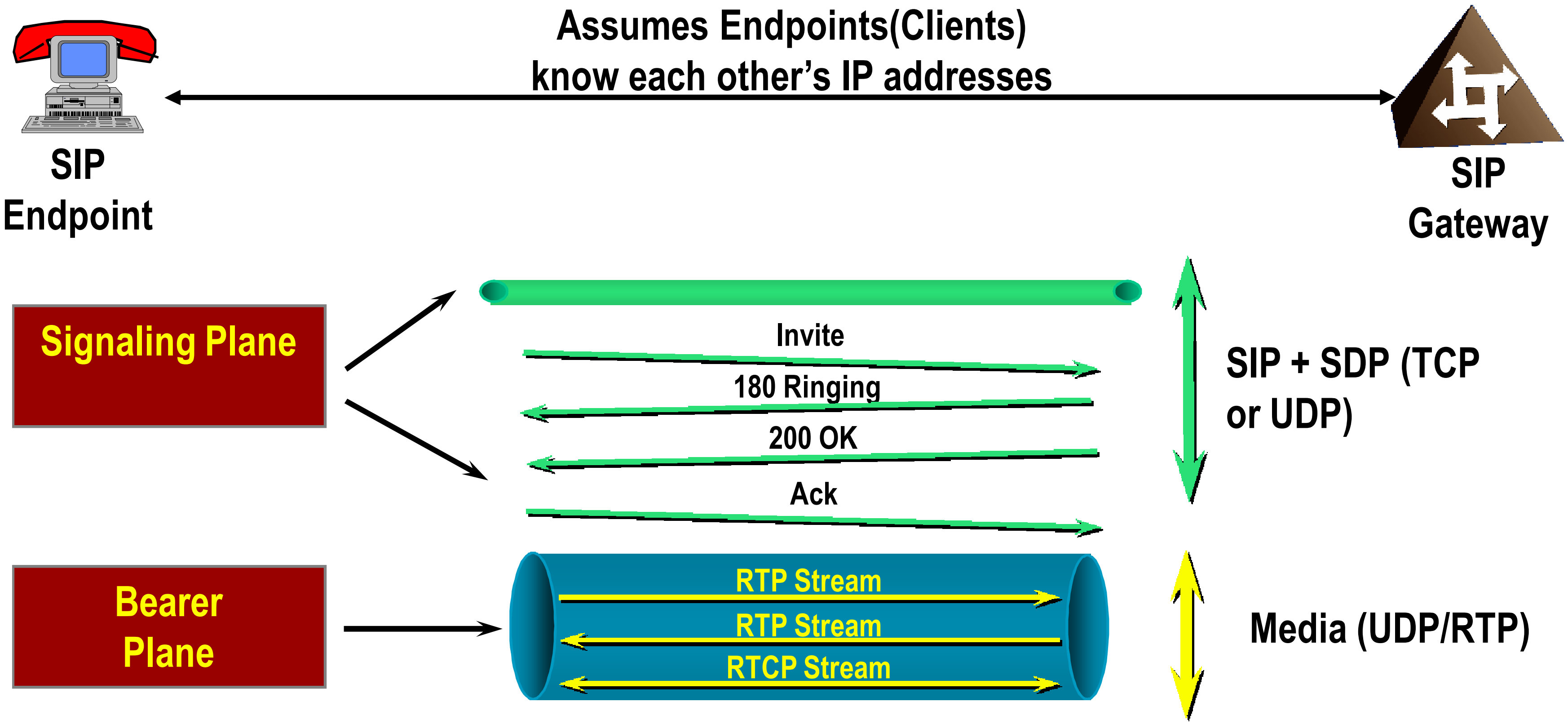


Three way handshake

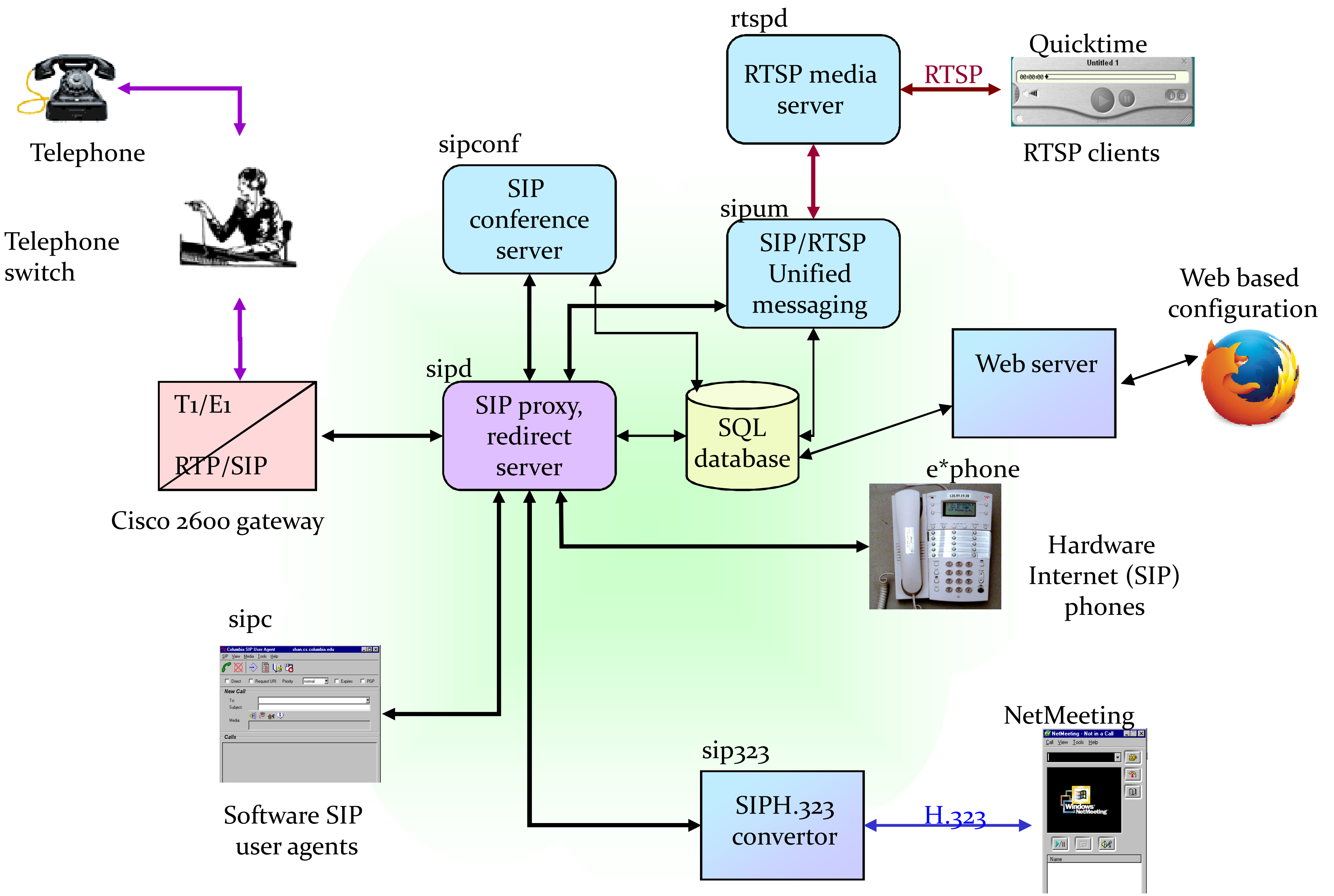
SIP enabled IP network



SIP Call Signaling



Example SIP-based Architecture

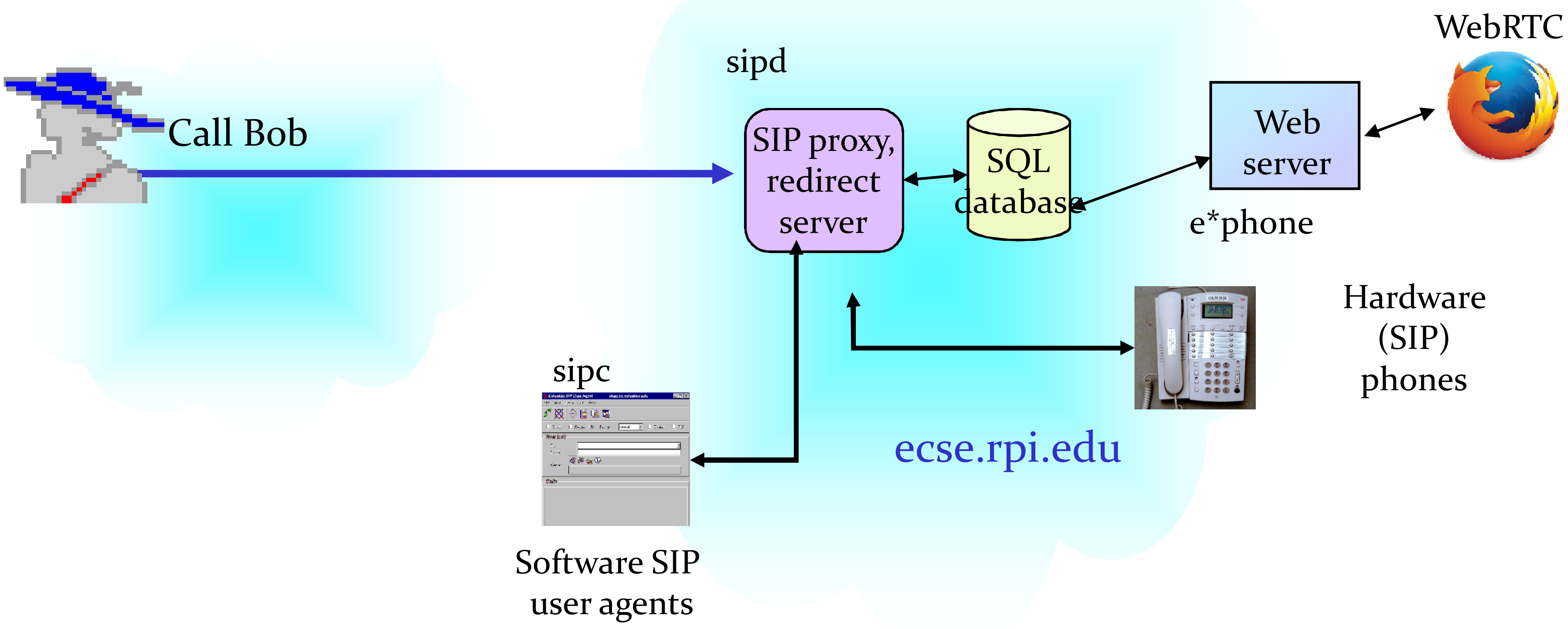


Example Call

- Bob signs up for the service from the web as "bob@ecse.rpi.edu"
- He registers from multiple phones
- Alice tries to reach Bob

`INVITEsip:Bob.Wilson@`ecse.rpi.edu

- sipd canonicalizes the destination to `sip:bob@ecse.rpi.edu`
- sipd rings both e*phone and sipc
- Bob accepts the call from sipc and starts talking



SIP Sessions

- Users may move between endpoints
- Users may be addressable by multiple names
- Users may communicate in several different media
- SIP: enables internet endpoints to
  - Discover each other
  - Characterize the session
- Location infrastructure: proxy servers, invite/register…
  - Name mapping and redirection services
- Add/remove participants from session
- Add/remove media from session

SIP Capabilities

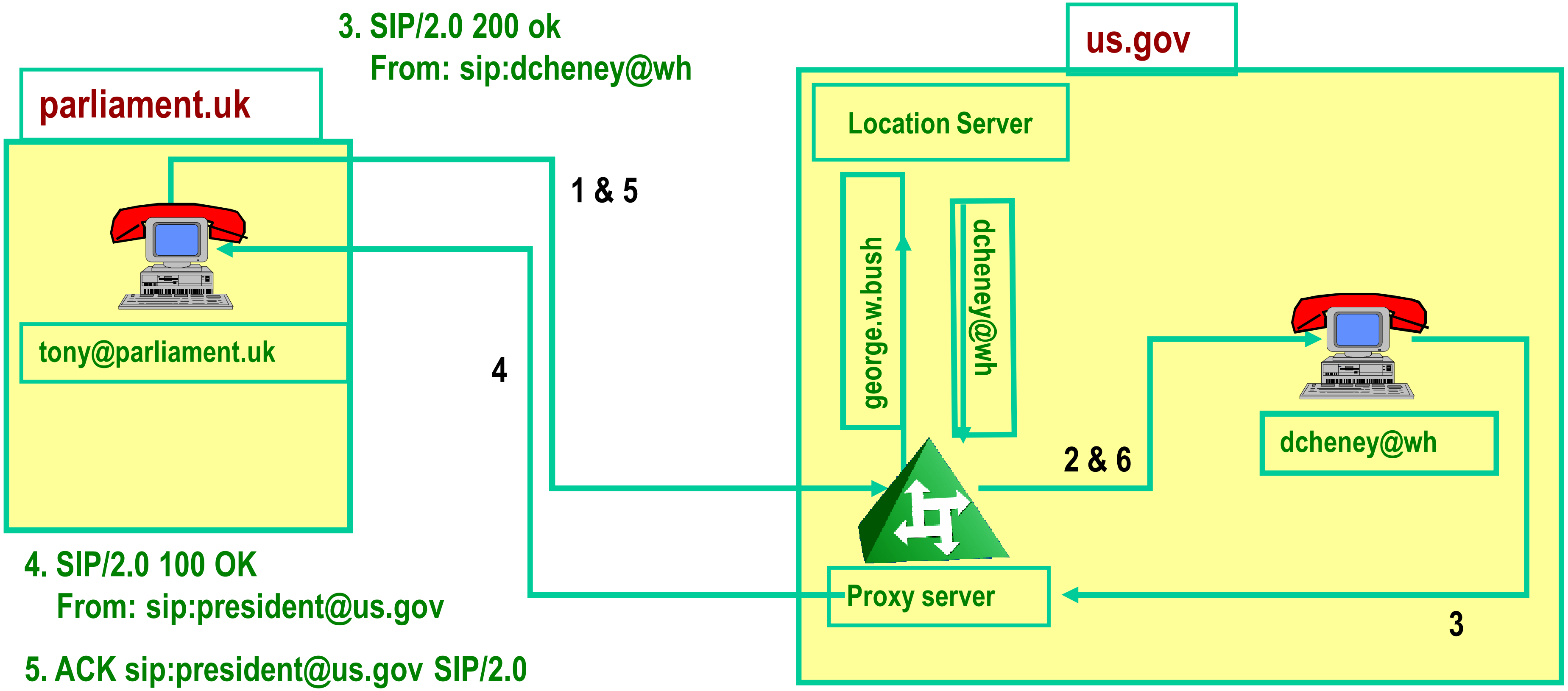
- User location: determination of the end system to be used for communication;
- User availability: determination of the willingness of the called party to engage in communications;
- User capabilities: determination of the media and media parameters to be used;
- Session setup: "ringing", establishment of session parameters at both called and calling party;
- Session management: including transfer and termination of sessions, modifying session parameters, and invoking services.

What SIP is not…

- SIP is not a vertically integrated communications system.
  - It is a component in a multimedia architecture.
- SIP does not provide services.
  - Rather, SIP provides primitives that can be used to implement different services.
  - For example, SIP can locate a user and deliver an opaque object to his current location.
- SIP does not offer conference control services
  - … such as floor control or voting
  - SIP does not prescribe how a conference is to be managed.

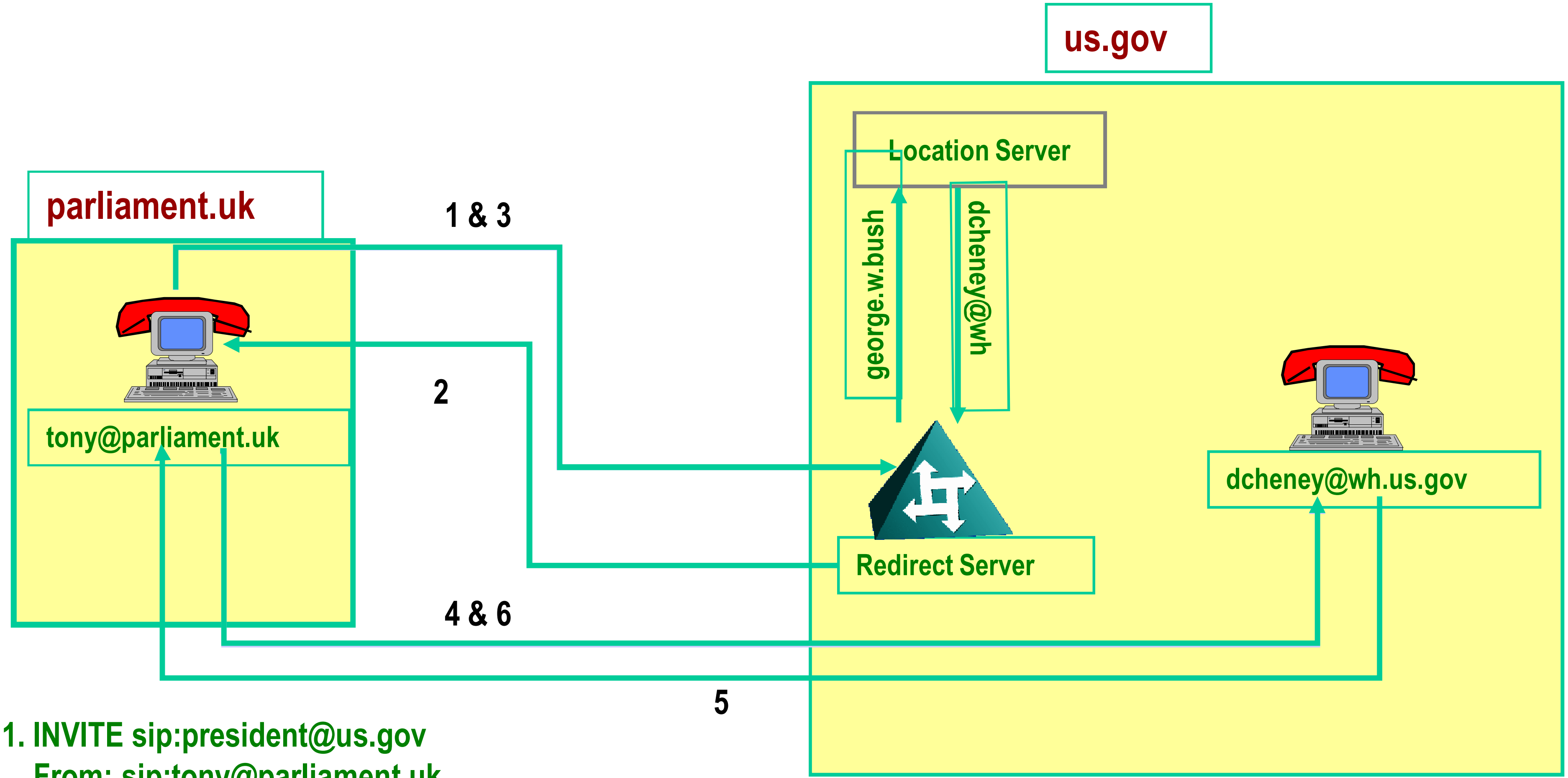
Example with Proxy Server

1. INVITEsip:president@us.gov SIP/2.0
   From: sip:tony@parliament.uk
2. INVITEsip:dcheney@wh SIP/2.0
   From: sip:tony@parliament.uk
3. SIP/2.0 200 ok
   From: sip:dcheney@wh



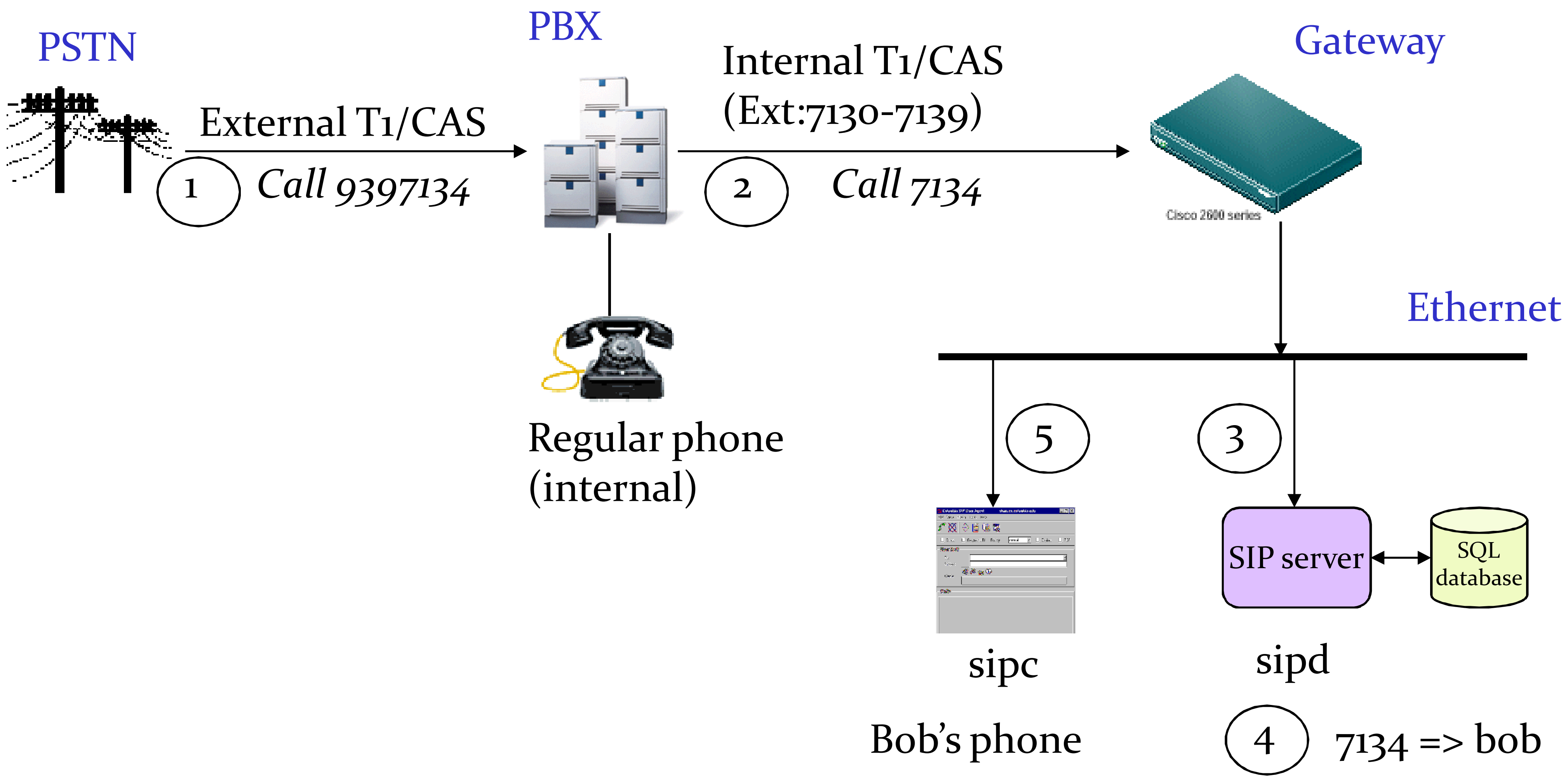
4. SIP/2.0 100 OK
   From: sip:president@us.gov
5. ACK sip:president@us.gov SIP/2.0
   From: sip:tony@parliament.uk
6. ACK sip:dcheney@wh SIP/2.0
   From: sip:tony@parliament.uk

Example with Redirect Server



1. INVITE sip:president@us.gov
   From: sip:tony@parliament.uk
2. SIP/2.0 320 Moved temporarily
   Contact: sip:dcheney@wh.us.gov
3. ACK sip:president@us.gov
   From: sip:tony@parliament.uk
4. INVITE sip:dcheney@wh.us.gov
   From: tony@parliament.uk
5. SIP/2.0 200 OK
   To: tony@parliament.uk
6. ACK sip:dcheney@wh.us.gov
   From: sip:tony@parliament.uk

Example: PSTN to IP Call



Example: IP to PSTN Call