Technologie-Zentrum Informatik

# SIP Tutorial

**Upperside SIP 2004** **Paris, 20 January 2004**

*Dr.-Ing. Jörg Ott* *{sip,mailto}:jo@tzi.org*

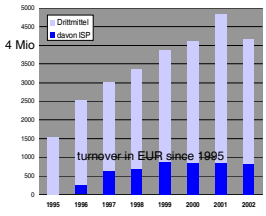




## Introduction: Who We Are

"Technologie-Zentrum Informatik"

- Founded 1995
- Non-Profit organization
- Part of the Dept. of Computer Science Universität Bremen, Germany
- ~100 Scientists (11 University Professors)
- Main focus is on technology transfer
  → Tech consulting, studies, workshops, cooperative projects, …
- 2002 turnover > € 4 Mio

# Research Group for Computer Networks

Architectures, Protocols and Interfaces for computer-supported communication and collaboration

- **Research Activities**
  - Multimedia-Conferencing, Media Transcoding
  - Data Distribution (Unicast/Multicast)
  - Content Transformation, XML Technologies
  - Call Signaling, Presence
  - Security in distributed systems (focus on secure real-time multimedia)

- **Standardization Activities**
  - IETF: MMUSIC, RoHC, (SIP)
  - TERENA: Task Force “Mobility”, “VoIP Cookbook”
  - Past: ITU-T (H.323)



→ Continuous work on MM Conferencing from early 1990s





# Prerequisites

- **Basic knowledge about computer networks**
  - packet switching, circuit switching
  - Some specific networking technologies such as Ethernet, serial, ATM
- **Familiarity with IP networks**
  - Internetworking “Architecture”, Internet Principles
  - Network Layer Protocols: IP, ICMP, …
  - Transport Protocols: UDP, TCP, …
  - Service Protocols: DNS, DHCP, …
  - Security Basics
- **Inclination to ask questions...**
- **Stamina :-)**

## This Week…

| Today | Wednesday | Thursday | Friday |
|---|---|---|---|
| SIP Tutorial | Status and Impact of SIP | SIP for small enterprise systems | Instant Messaging & Presence |
| | SIP Deployments | SIP Security | Public Safety |
| | | Mobile Networks | Application Building |
| The Real-World Impact of SIP | SIP – PSTN Interconnection | SIP Services | Charging & Billing |





## This Tutorial…

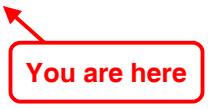
- **Proposed Schedule:**

| | |
|---|---|
| 1000 – 1100 | IP Multimedia, SIP Intro, Basics |
| 1100 – 1120 | Morning Break |
| 1120 – 1230 | Basic Call Flows, Registration, Location |
| 1230 – 1400 | Lunch Break |
| 1400 – 1515 | Security, SIP Building Blocks |
| 1515 – 1545 | Afternoon Break |
| 1545 – 1700 | Telephony, Deployment Issues, (3GPP) |
| 1700 – 1730 | The Real-World Impact of SIP |

- **We know that there is nothing like a 20min coffee break – but please let's try anyway...**

# Tutorial Overview

- Internet Multimedia Conferencing Architecture
- Packet A/V Basics + Real-time Transport
- SIP Introduction, History, Architecture
- SIP Basic Functionality, Call Flows
- SIP Security
- SIP Service Creation
- SIP in Telephony
- SIP in 3GPP








# IP Multimedia Applications (1)

- Packet multimedia experiments since 1980s
  - A/V tools + protocols for A/V over IP
  - Conference control protocols

<u>Internet broadcasting (Mbone)</u>

- First IETF Audiocast (1992)
- Broadcasts of IETF WG sessions
  - audio + video + whiteboard (transparencies)
  - enables remote participation (even talks)
- Broadcasting special events
  - talks, concerts, NASA shuttle missions, ...
- Broadcasting “radio” and “television” programs
  - numerous channels available today

# IP Multimedia Applications (2)

## Teleconferences

- Traditional Internet focus: large groups
- Small groups supported as well
- Audio + video + data (whiteboards, editors, ...)
- Multimedia gaming sessions
- Examples:
  - seminars and lectures
  - project meetings
  - work group meetings between IETFs
- Gatewaying where needed (PSTN, ISDN, cellular, …)





# IP Multimedia Applications (3)

## IP Telephony

- “Special case” of teleconferences
  - point-to-point + conference calls
- Gatewaying to PSTN / ISDN / GSM
  - also H.323, MGCP, MEGACO
- Include “Supplementary Services”
  - what users are used to from their touch tone phone or PBX environment
- Include “Intelligent Network (IN)” services
  - To some degree trivial in an IP environment

# IP Multimedia Applications (4)

**Multimedia retrieval services**

- "Video on demand"-style
  - including “VCR controls”: pause/restart/cue/review
- Access to multimedia clips from web browsers
  - Commercial examples: RealAudio/RealVideo, IP/TV, Microsoft
- Often: Internet- / web-based access to live streams
  - “Big Brother”, concerts, etc.
- Option: recording multimedia information





# Common Requirements

**Network infrastructure**

- Multicast routing
- Real-time-capable packet forwarding
- Resource reservation

**Transport protocols**

- Real-time (audio / video) information
- Non-real-time (data / control) information

**Media encoding standards**

**Security**

## Specific requirements

**Control protocols**
- **Setup / teardown of communication relationships**
- **Call (and conference) control**
- **Remote control of devices (e.g. media sources)**

**Naming and addressing infrastructure**

**User (and service) location**

**Billing and accounting (and policing)**

**(Legal requirements)**

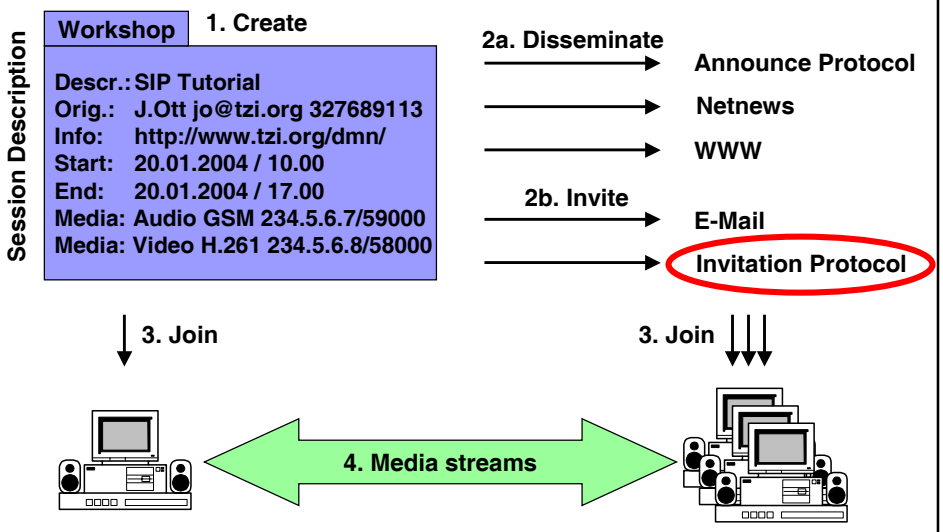




## Traditional IETF Conferencing Concept

- **Multicast-based**
- **Loosely-coupled conferences**
  - **no membership control**
  - **inexact information about participants**
    - **provided on a voluntary basis**
  - **security by encryption**
- **Public announcements and invitations**
  - **Convey session parameters, then get out of the way**
    - **Session Announcement Protocol (SAP)**
    - **Session Initiation Protocol (SIP)**
- **Conference control**
  - **Some need perceived; limited success over many years**

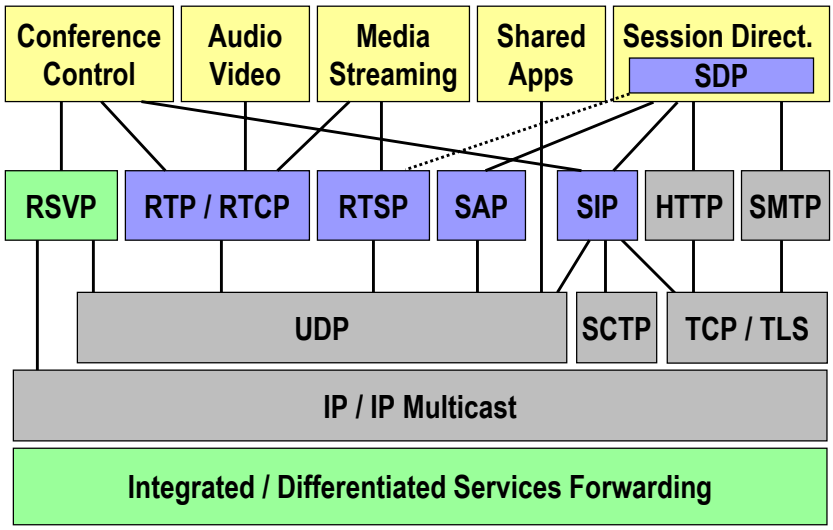
# Conference Establishment








# IETF Conferencing Architecture



# Conference Description

## Session Description Protocol (SDP, RFC 2327)

- All you need to know about a session to join
  - who? — convener of the session + contact information
  - what about? — name and informal subject description
  - when? — date and time, rep
  - where? — multicast addresses, port numbers
  - which media? — capability requirements
  - how much? — required bandwidth
  - …
- Grouped into three categories
  - 1 x session, m x time, n x media





# Conference Discovery

## Session Announcement Protocol (SAP)

- Advertise conference description regularly
  - announcement frequency depends on distribution scope
- Originally: avoid conflicts in multicast addresses
  - simple multicast address allocation (224.2.0.0/16)
- Optional security support
  - authentication header and encrypted payload

## Alternatives

- Use conventional means
  - e-mail (SMTP), web servers (HTTP), Netnews (NNTP)
  - MIME type defined for SDP specification: “application/sdp”
- Session Invitation Protocol (SIP)

## Tutorial Overview

- Internet Multimedia Conferencing Architecture
- Packet A/V Basics + Real-time Transport
- SIP Introduction, History, Architecture
- SIP Basic Functionality, Call Flows
- SIP Security
- SIP Service Creation
- SIP in Telephony
- SIP in 3GPP

You are here



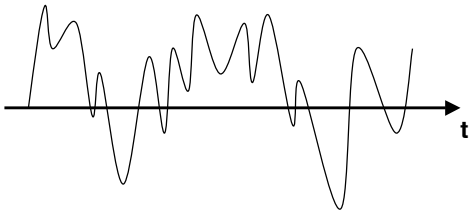


## Real-time Media over Packets

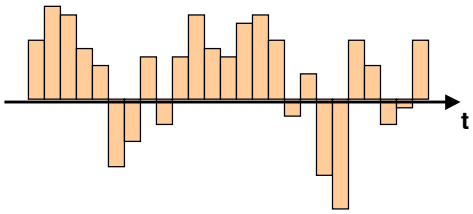
- Audio / Video are continuous media
- Packet networks transport discrete units
  - digitize media
  - compression
  - packetization
- No additional multiplex (beyond UDP/IP) needed:
  - no separate lines, bit allocations, etc.
  - transport different media in different packets
  - can give different quality of service to different media
  - allows different sites to receive different subsets
- Issue: Variable, partially unpredictable delay…

# Real-time Media over Packets (2)

1) analog input signal




2) sampled input signal (implicit compression)

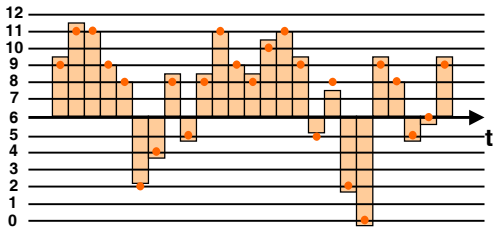




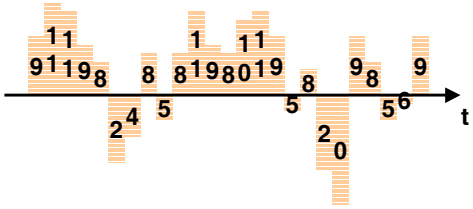



# Real-time Media over Packets (3)

3) Quantization (another step of implicit compression)



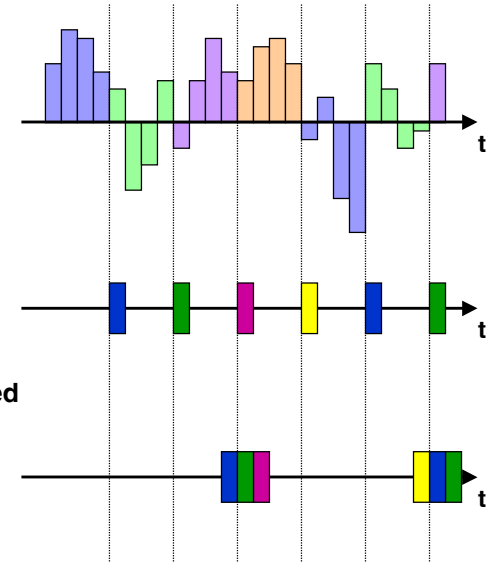

4) Digital data stream



# Real-time Media over Packets (4)

5) optional further compression yields small discrete frames

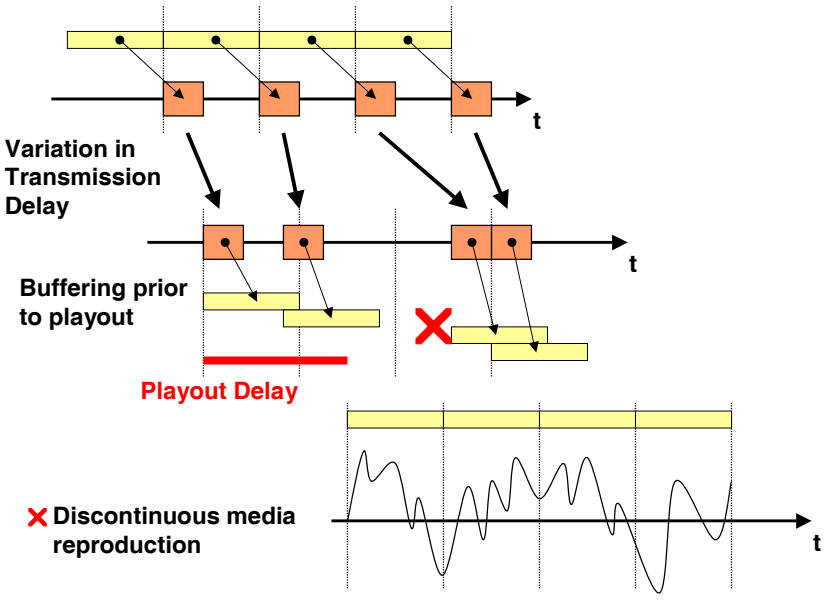
6) multiple frames or samples are collected to form packets








# Real-time Media over Packets (5)




✕ Discontinuous media reproduction

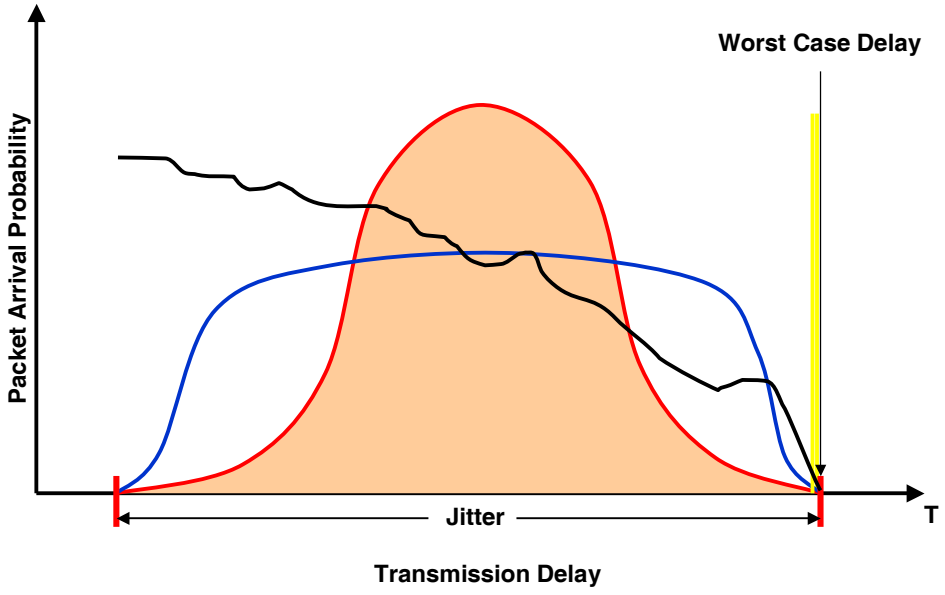
# Sources of Delay

- **Sender**
  - Capturing / digitizing delay (+ operating system)
  - Encoding / compression delay
  - Packetization delay
- **Network (potentially highly variable!)**
  - Link propagation delay (order of speed of light)
  - Serialization delay
  - Queuing delay
- **Receiver**
  - buffering delay + potential delay for repair
  - decoding / decompression delay
  - rendering / replay delay (+ operating system)

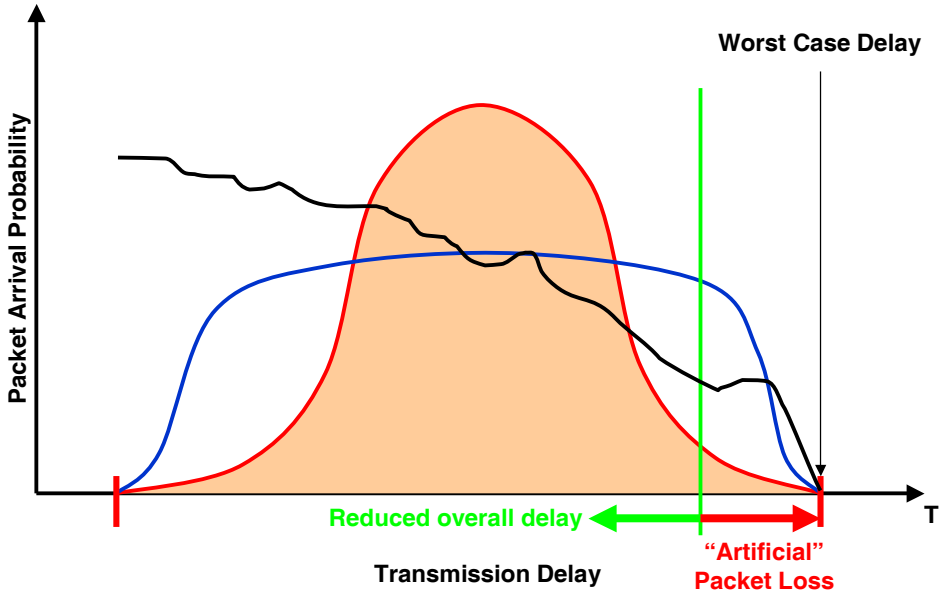




# Jitter vs. Delay



## Artificial Upper Bound on Delay








## Real-time Media over Packets (6)

Little help needed from transport protocol:

- Retransmission would take too long (interactivity!)

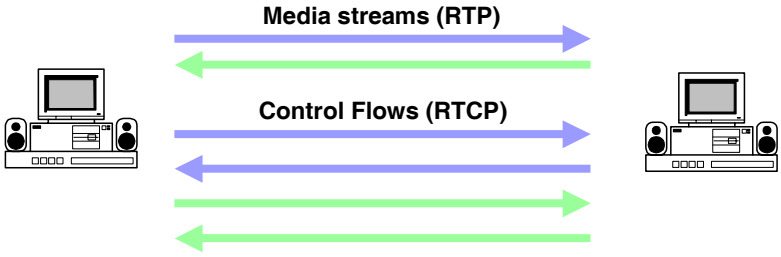
End systems must buffer before playout!

- Jitter in transmission delay due to queueing
- Packet A/V rule #1:
  - jitter is never a problem,
  - worst-case delay is!
- Need a timestamp in packet to be able to play at right time
  - intra-stream timing
  - optionally correlate for inter-stream timing (e.g. lip-sync)

# Real-time Transport Protocol (1)

- RTP Functionality (RFC1889)
  - framing for audio/video information streams
  - preserve intra- and inter-stream timing
  - mechanisms for awareness of others in a conference

  ➔ RTP sessions



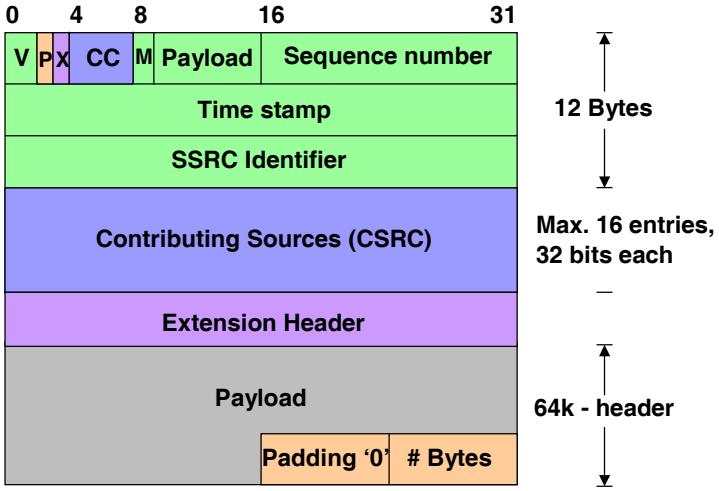





# Real-time Transport Protocol (2)

- Mechanisms
  - detect packet loss, cope with reordering
    - sequence number per media stream
  - determine variations in transmission delays
    - media specific time stamp (e.g., 8 kHz for PCM audio)
    - allows receiver to adapt playout point for continuous replay
  - source identification
    - possibly mixed from several sources
  - payload type identifier
- Standard RTP packet header
  - independent of *payload type*
  - possibly seconded by *payload header*

# RTP Header








# RTP Header Fields (1)

V: Version — version 2 defined in RFC 1889

P: Padding — indicates padding
# bytes indicated in last byte

X: eXtension bit — extension header is present

Extension header — single additional header (TLV coded)

CC: CSRC count — # of contributing sources

CSRC: contributing sources — which sources have been "mixed" to produce this packet's contents

## RTP Header Fields (2)

| | | |
|---|---|---|
| M: Marker bit | — | marks semantical boundaries in media stream (e.g. talk spurt) |
| Payload type | — | indicates packet content type |
| Sequence # | — | of the packet in the media stream (strictly monotonically increasing) |
| Timestamp | — | indicates the instant when the packet contents was sampled (measured to media-specific clock) |

SSRC: synchronization source — identification of packet originator





## Real-time Transport Control Protocol

**Mechanisms:**

- Receivers constantly measure transmission quality
  - delay, jitter, packet loss
- Regular control information exchange between senders and receivers
  - feedback to sender (receiver report)
  - feed forward to recipients (sender report)
- Allows applications to adapt to current QoS
- Overhead limited to a small fraction (default: 5 % max.) of total bandwidth per RTP session
  - members estimate number of participants
  - adapt their own transmission rate

**Obtaining sufficient capacity: outside of RTP!**

# RTP Payload Types

- 7-bit payload type identifier
  - some numbers statically assigned
  - dynamic payload types identifiers for extensions – mapping to be defined outside of RTP (control protocol, e.g. SDP “a=rtpmap:”)

**Payload formats defined for many audio/video encodings**

- Conferencing profile document RFC 1890
  - audio: G.711, G.722, G.723.1, G.728, GSM, CD, DVI, …
- In codec-specific RFCs
  - audio: Redundant Audio, MP-3, ...
  - video: JPEG, H.261, MPEG-1, MPEG-2, H.263, H.263+, BT.656, …
  - others: DTMF, text, SONET, …
- Generic formats
  - generic FEC, (multiplexing)



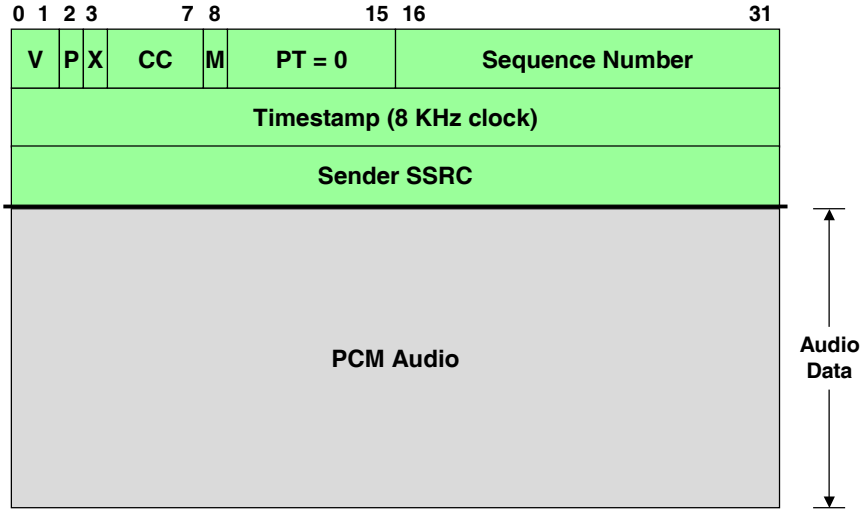


# Media Packetization Schemes (1)

**General principle:**

- Payload specific additional header (if needed)
- Followed by media data
  - packetized and formatted in a well-defined way
  - trivial ones specified in RFC 1890
  - RFC 2029, 2032, 2035, 2038, 2190, 2198, 2250, 2343, 2429, 2431 RFC 2435, 2658, 2733, 2793, 2833, 2862, 3016, 3047, 3119, 3189 RFC 3190, 3267, 3389, and many Internet Drafts
  - Guidelines for writing packet formats: RFC 2736
- Functionality
  - enable transmission across a packet network
  - allow for semantics-based fragmentation
  - provide additional information to simplify processing and decoding at the recipient
  - maximize possibility of independent decoding of individual packets
  - Typically, little to do for audio!

# Audio over RTP: PCM








# Video over RTP: H.261

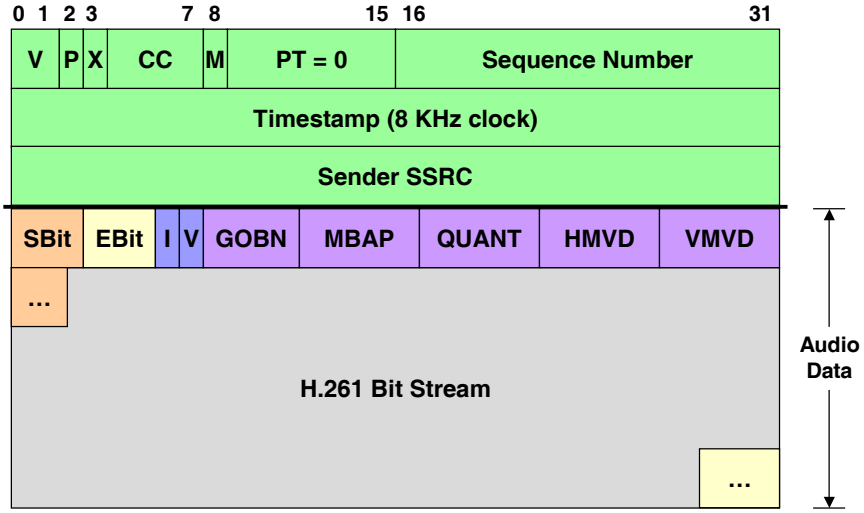
**Additional payload-specific header preceeds payload**

- To avoid expensive bit shifting operations
  - Indicate # invalid bits in first (SBit) and last (EBit) octet of payload
- Indicate Intra encoding (I bit)
- Indicate the presence of motion vector data (V bit)
- Carry further H.261 header information to enable decoding in the presence of packet losses

**Further mechanisms for video conferencing (in RTCP)**

- FIR: Full Intra Request
  - Ask sender to send a full intra encoded picture
- NACK: Negative Acknowledgement
  - Indicate specific packet loss to sender

# Video over RTP: H.261 (2)



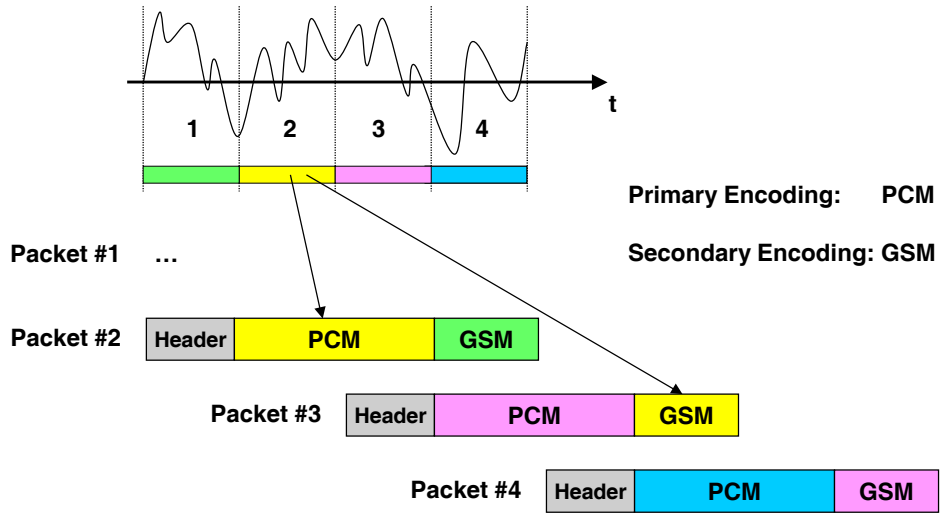





# Audio Redundancy Coding (1)

- Audio Packets are small!
  - have to be because of interactivity (avoid large packetization delay)
  - packet loss primarily depends on packet rate (rather than packet size)
- Payloads for multiple time slots in one packet
  - send redundant information in packet n to reconstruct packets k, ..., n-1 in packet n
  - redundant information typically sent at lower quality
  - details defined in RFC 2198
  - uses dynamic payload type
- Format specification, e.g. using SDP
  - m=audio 20002 RTP/AVP 96 0 0 0
  - a =rtpmap:96 red/8000/1

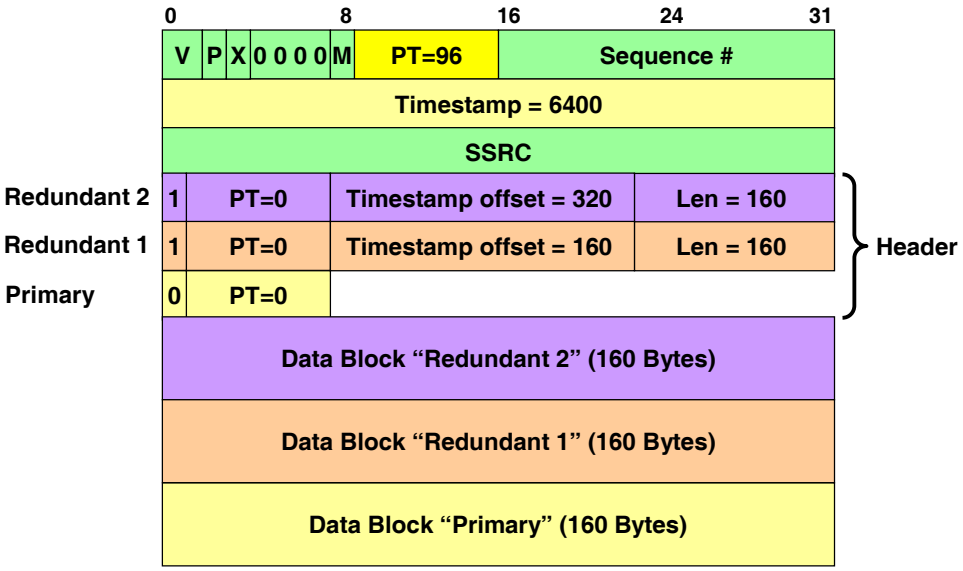
# Audio Redundancy Coding (2)








# Audio Redundancy Coding (3)



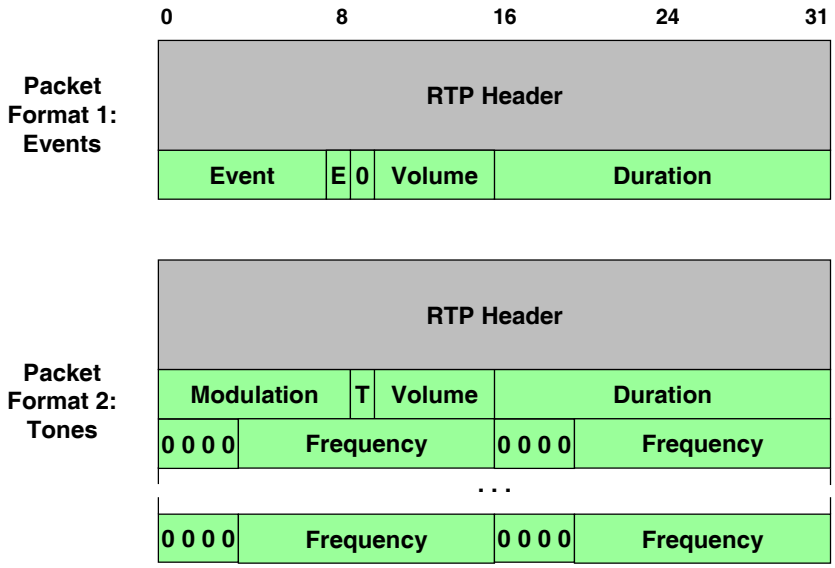
# DTMF over RTP (1)

- DTMF digits, telephony tones, and telephony signals
  - two payload formats
  - 8 kHz clock by default
  - audio redundancy coding for reliability
- Format 1: reference pre-defined events
  - 0 - 9 * # A - D Flash [17]
  - modem and fax tones [18]
  - telephony signals and line events [43]
    - dial tones, busy, ringing, congestion, on/off hook, …
  - trunk events [44]
  - specified through identifier (8-bit value), volume, duration
- Format 2: specify tones by frequency
  - one, two, or three frequencies
  - addition, modulation
  - on/off periods, duration
  - specified through modulation, n x frequency, volume





# DTMF over RTP (2)



# Media Packetization Schemes (2)

**Error-resilience for real-time media**

- Input: Observation on packet loss characteristics
- Generic mechanisms (RFC 2354)
  - Retransmissions
    - in special cases only (interactivity!)
  - Interleaving
  - Forward Error Correction (FEC)
    - media-dependent vs. media-independent
    - Generic FEC: RFC 2733
    - Audio Redundancy Coding (RFC 2198)
- Feedback loops for senders
  - based upon generic and specific RTCP messages
  - adapt transmission rate, coding scheme, error control, ...





# Tutorial Overview

- Internet Multimedia Conferencing Architecture
- Packet A/V Basics + Real-time Transport
- SIP Introduction, History, Architecture
- SIP Basic Functionality, Call Flows
- SIP Security
- SIP Service Creation
- SIP in Telephony
- SIP in 3GPP

You are here

# SIP: Session Initiation Protocol

**From HTTP and Session Invitation to Setup and Control for Packet-based Multimedia Conferencing**





# History of Mbone conference initiation

**Session Invitation Protocol**

(Handley/Schooler)

- Participant location
- Conference invitation
- Capability negotiation during setup

**Simple Conference Invitation Protocol**

(Schulzrinne)

- Participant location
- Conference invitation
- Capability negotiation during setup
- Changing conference parameters
- Terminate/leave conference

1996 ⇒ ***Session Initiation Protocol***

## Session Initiation Protocol (SIP)

**First draft in December 1996**

- Joint effort to merge SIP and SCIP
- IETF WG MMUSIC
  (Multiparty Multimedia Session Control)

**Application-layer call signaling protocol:**

- Creation, modification, termination of teleconferences
- Negotiation of used media configuration
- Re-negotiation during session
- User location → personal mobility
- Security
- Supplementary services

**RFC 3261**
- June 2002
- obsoletes RFC 2543





## SIP and Conferencing over Time…

- Origin: MMUSIC: Multiparty Multimedia Session Control

- From Invitation… to initiation, modification, and termination
- From Multiparty… to point-to-point-focused
- From Multimedia… to voice-centric

Now: Multiparty & multimedia rediscovered

But: Don't believe in multicast (anymore)!

# Timeline: 1996

**Initial Internet Drafts:**
**Session Invitation Protocol (SIP) – M. Handley, E. Schooler**
**Simple Conference Invitation Protocol (SCIP) – H. Schulzrinne**

**SIP: Setup + Caps Negotiation**

**SCIP: Setup + Caps Modify + Terminate**

**Presentations at 35th IETF, Los Angeles**

**Merged Draft: SIP -01**

**Main Features set: TCP/UDP, Forking, Redirection, addrs INVITE,CAPABILITY From: To: Path:**

22 Feb 1996 — 4-8 Mar 1996 — 2 Dec 1996





# Timeline: 1997

**Draft SIP -02**

**Formal syntax**
**CAPABILITY ➔ OPTIONS**
**Path: ➔ Via:**
**Ideas for Alternates:**

**IETF Action: Split SIP into base spec and extensions**

**Draft SIP -04**

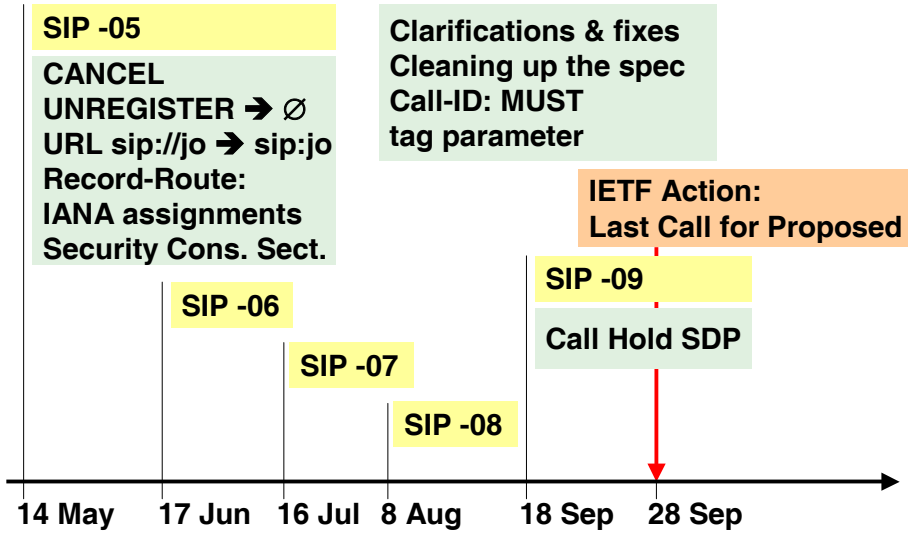
**CONNECTED ➔ ACK**
**UNREGISTER**
**Sequence: ➔CSeq:**
**Call-Disposition:**
**Require:**

**Draft SIP -03**

**SIP URL: sip://jo@…**
**CONNECTED, BYE, REGISTER**
**Call-ID: Sequence:**
**Allow: Expires:**

27 Mar 97 — 31 Jul 97 — 11 Nov 97 — Dec 97

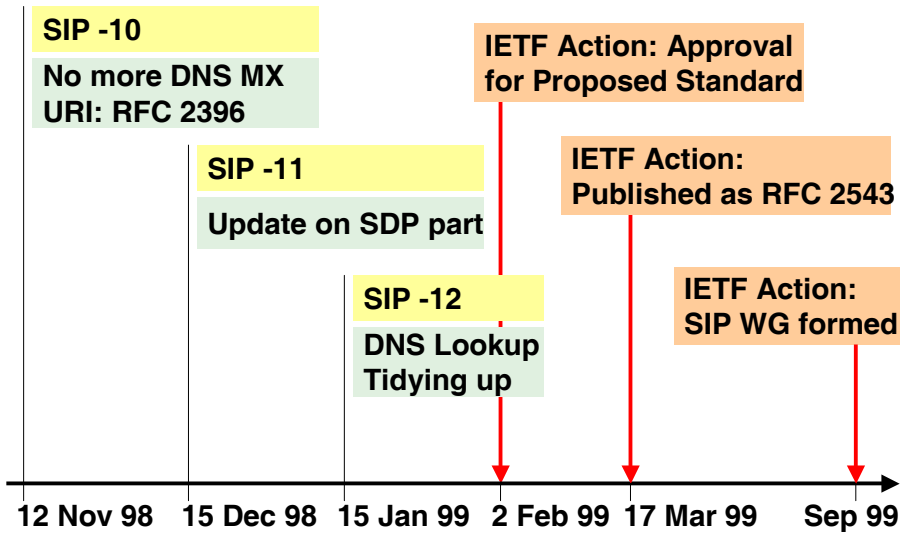
# Timeline: 1998



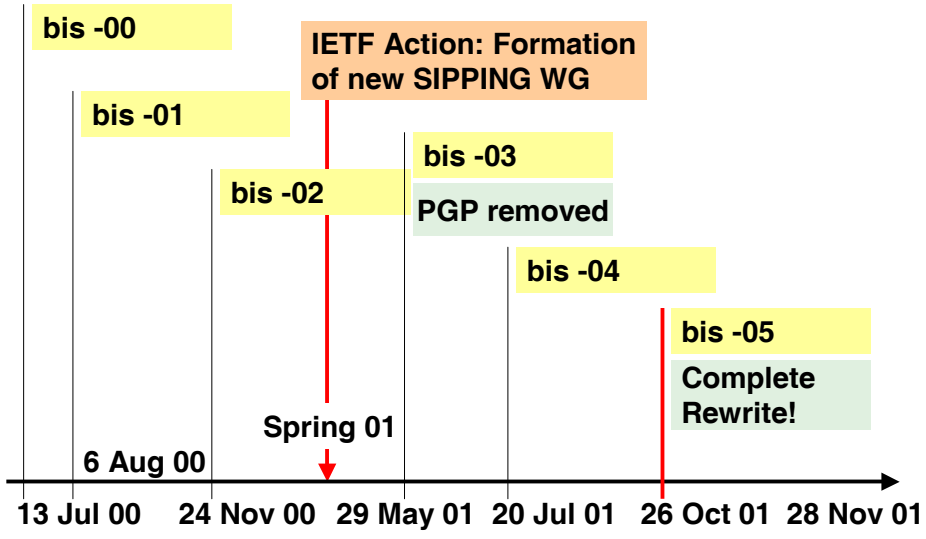





# Timeline: 1998/99



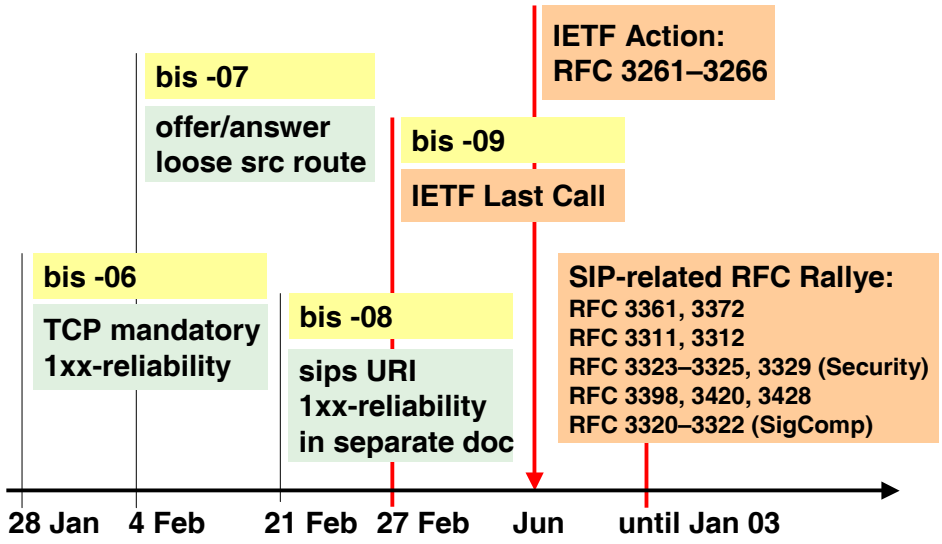
## Timeline: RFC2543bis (2000/2001)








## Timeline: RFC2543bis, RFC3261 (2002)



# RFCs related to SIP

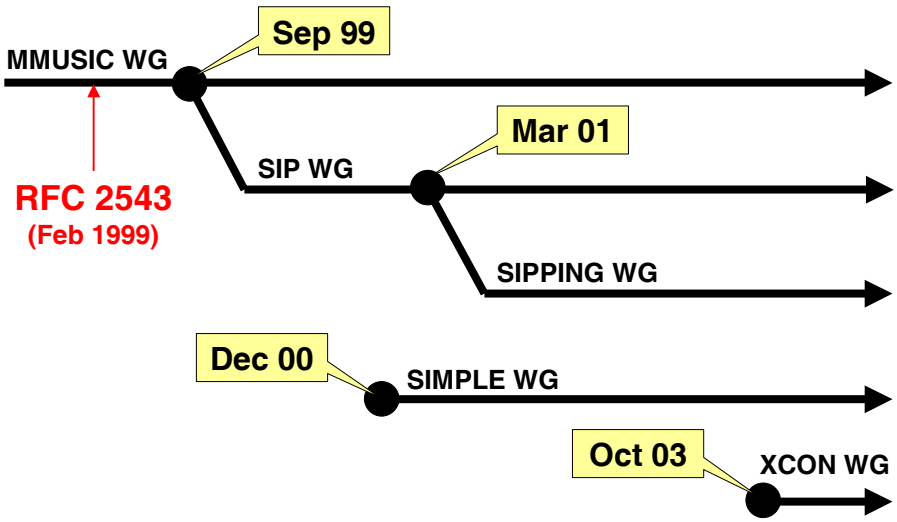
- Base spec
  - RFC 3261: SIP: Session Initiation Protocol
  - RFC 3263: Locating SIP Servers
  - RFC 3264: An Offer/Answer Model with SDP
- Extended Features
  - RFC 2976: The SIP INFO Method
  - RFC 3262: Reliability of Provisional Responses in SIP
  - RFC 3265: SIP-specific Event Notification
  - RFC 3311: SIP UPDATE Method
  - RFC 3326: Reason Header
  - RFC 3327: Registering Non-Adjacent Contacts
  - RFC 3428: Instant Messaging
  - RFC 3485, 3486: SIP Compression
  - RFC 3487: Requirements for Resource Priority
  - RFC 3515: SIP REFER Method
  - RFC 3581: Symmetric Message Routing
- Security
  - RFC 3323: A Privacy Mechanism for SIP
  - RFC 3325: Private Extension for Asserted Identity in Trusted Networks
  - RFC 3329: Security-Mechanism Agreement for SIP
  - RFC 3603: Proxy-to-Proxy Extensions
- Others
  - RFC 3665, 3666: SIP Call Flows
  - RFC 3312: Integration of Resource Management and SIP
  - RFC 3361: DHCP Option for SIP Servers
  - RFC 3608: Service Route Discovery
  - RFC 3398, 3578: ISUP and SIP Mapping
  - RFC 3420: Internet Media Type message/sipfrag
  - RFC 3427: SIP Change Process
  - RFC 3455: Header Extensions for 3GPP
  - SDP, RTP, ENUM, CMS, AKA, ...

Plus some 150+ Internet Drafts





# IETF SIP-related Working Groups (1)



## IETF SIP-related Working Groups (2)

| | |
|---|---|
| MMUSIC WG → | – SDP extensions<br>– SDPng |
| SIP WG → | – SIP core spec maintenance<br>– SIP protocol extensions |
| SIPPING WG → | – Requirements for SIP<br>– Specific SIP application services |
| SIMPLE WG → | – SIP for Presence and Instant Messaging |
| XCON WG → | – Centralized Conferencing |





## SIP is not …


- Intended for conference control by itself
  - No floor control
  - No participant lists
  - No policies, voting, …
- Designed for distribution of multimedia data
  - Some extensions allow for carrying images, audio files, etc.
- A generic transport protocol!
- Another RPC mechanism
  - SIP has no inherent support for distributed state information
- …
- (but proposals for “misuse” show up again and again)

# Tutorial Overview

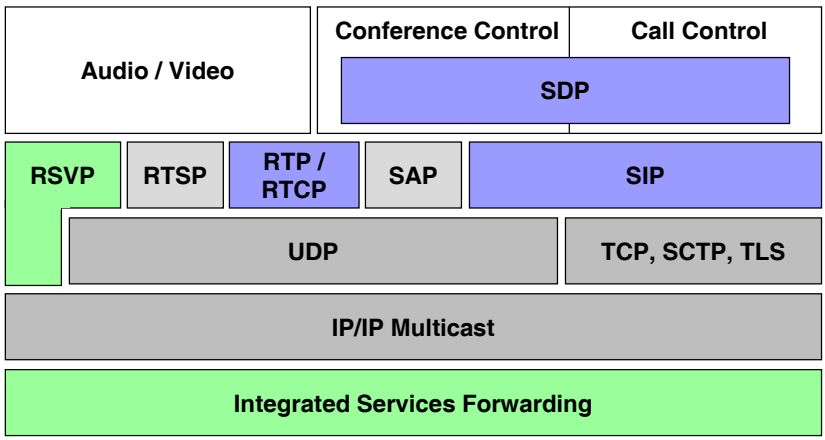
- Internet Multimedia Conferencing Architecture
- Packet A/V Basics + Real-time Transport
- SIP Introduction, History, Architecture
- SIP Basic Functionality, Call Flows
- SIP Security
- SIP Service Creation
- SIP in Telephony and Deployment Issues
- SIP in 3GPP

You are still here





# SIP and the Multimedia Conferencing Architecture



# Base Terminology

- **User Agent Client (UAC):**
  - Endpoint, initiates SIP transactions
- **User Agent Server (UAS):**
  - Handles incoming SIP requests

} User Agent

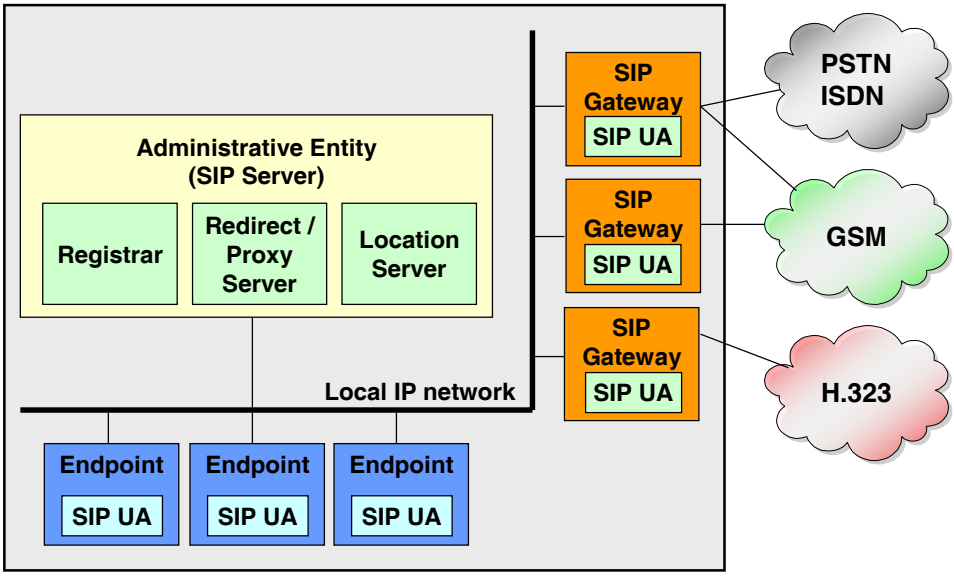
- **Redirect server:**
  - Retrieves addresses for callee and returns them to caller
- **Proxy (server):**
  - UAS/UAC that autonomously processes requests
    → forward incoming messages (probably modified)
- **Registrar:**
  - Stores explicitly registered user addresses
- **Location Server:**
  - Provides information about a target user's location

- **Back-to-Back User Agent (B2BUA)**
  - Keeps call state; more powerful intervention than proxy





# Local SIP Architecture



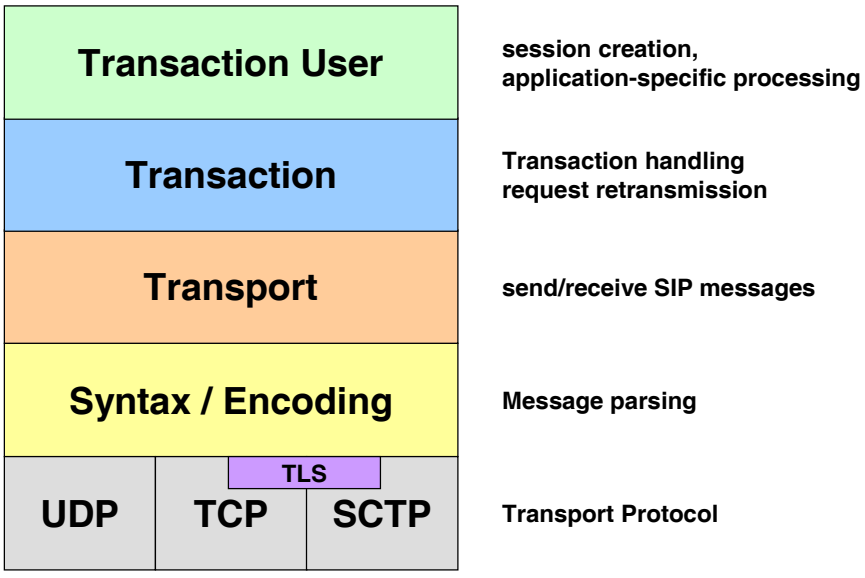
## Protocol Characteristics

- Transaction oriented
  - Request–response sequences
- Independent from lower layer transport protocol
  - Works with a number of unreliable and reliable transports
    - UDP, TCP, SCTP
    - Secure transport: TLS over TCP, IPSec
  - Retransmissions to achieve reliability over UDP
  - Optionally use IP multicast → anycast service
- Independent of the session to be (re-)configured
- Re-use syntax of HTTP 1.1
  → Text-based protocol (UTF-8 encoding)
- Enable servers maintaining minimal state info
  - Stateless proxies
  - Transaction-stateful proxies
  - Dialog (call) state in endpoints (optional for proxies)

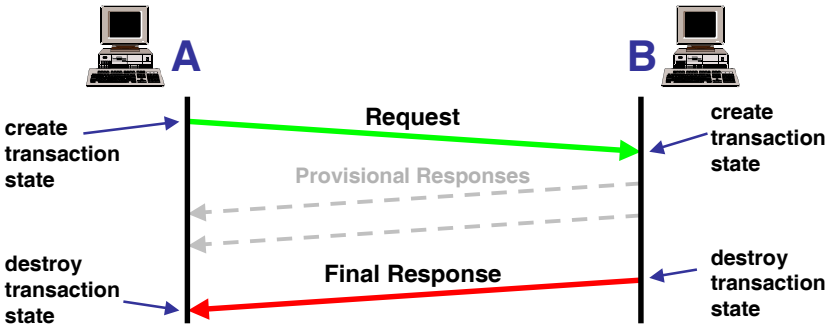




## Functional Layers



# SIP Transactions



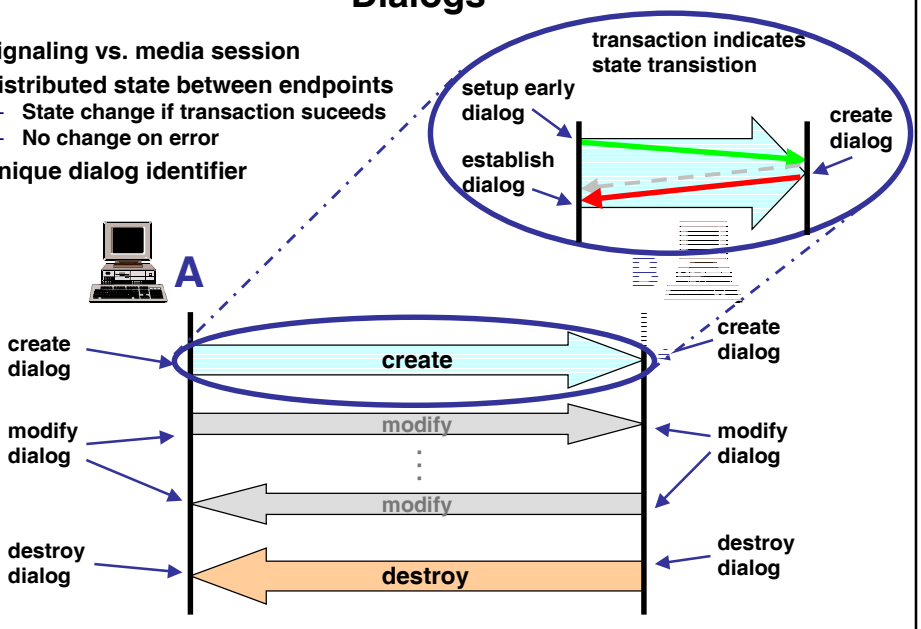

- **RPC-like approach:**
  - Initial request
  - Wait for final response
- **Provisional responses:**
  - Additional status information
  - May be unreliable
- **Unique identifier (transaction id)** (originator, recipient, unique token, sequence number, ...)
- **Independent completion**



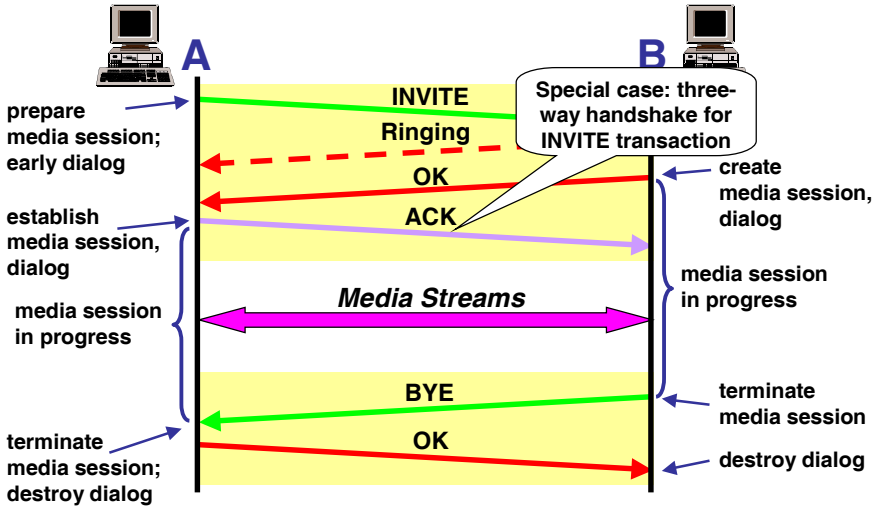


# Dialogs

- **Signaling vs. media session**
- **Distributed state between endpoints**
  - State change if transaction suceeds
  - No change on error
- **Unique dialog identifier**



# Dialog Example: Media Sessions








# SIP Message Syntax: Request

Start line

Message headers

Message body (SDP content)

```
INVITE sip:user@example.com SIP/2.0
To: John Doe <sip:user@example.com>
From: sip:jo@tzi.uni-bremen.de;tag=4711
Subject: Congratulations!
Content-Length: 117
Content-Type: applicaton/sdp
Call-ID: 2342344233@134.102.218.1
CSeq: 49581 INVITE
Contact: sip:jo@134.102.224.152:5083
          ;transport=udp
Via: SIP/2.0/UDP 134.102.218.1;
      branch=z9hG4bK776asdhds

v=0
o=jo 75638353 98543585 IN IP4 134.102.218.1
s=SIP call
t=0 0
c=IN IP4 134.102.224.152
m=audio 47654 RTP/AVP 0 1 4
```

# SIP Message Syntax: Response

Start line

Message headers

Message body (SDP content)

```
SIP/2.0 200 OK
To: John Doe <sip:user@example.com>;tag=428
From: sip:jo@tzi.uni-bremen.de;tag=4711
Subject: Congratulations!
Content-Length: 121
Content-Type: applicaton/sdp
Call-ID: 2342344233@134.102.218.1
CSeq: 49581 INVITE
Contact: sip:jdoe@somehost.domain
Via: SIP/2.0/UDP 134.102.218.1;
     z9hG4bK776asdhds

v=0
o=jdoe 28342 98543601 IN IP4 134.102.20.22
s=SIP call
t=0 0
c=IN IP4 134.102.20.38
m=audio 61002 RTP/AVP 0 4
```





# SIP URI Addressing Scheme

**sip: / sips:**

- **Separating names (permanent) and addresses (temporary)**
  - Basic mobility support
- **Two roles reflected in SIP**
  - Naming a user; typically sip:user@domain
  - Contact address of a user; typically contains host name or IP address, port, transport protocol, ...
- **URIs may carry additional parameters**

*´sip:´ [ user [ ´:´ passwd ] ´@´ ] host [ ´:´ port ] params [ ´?´ headers ]*

*params ::= ( ´;´ name [ ´=´ value ] )**
*headers ::= field ´=´ value? [ ´&´ headers ]*

- **URIs may also identify services**

# SIP Addressing Examples

```
sip:tzi.org
sip:192.168.42.1
```

Registration domain or IP address

```
sip:john@example.com
sip:john@example.com:5060
sip:foo@example.com:12780
```

Userinfo + domain/host optional port number

URI parameters may carry detailed information on specific URI components:

```
sip:john@Example.COM;maddr=10.0.0.1
sip:+1555123456@tel-gw.myitsp.com;user=phone
```

**Use URI scheme 'sips' to request secure transmission.**





# Service Description

Encapsulation

```
sip:sip%3Ajo%40192.168.42.5%3Bmaddr=134.102.3.99@example.com
```

Need to encode reserved characters

Service indication example

```
sip:voicemail.replay=ab1x817m@media-engine;msgid=78
```

Additional header fields (line breaks inserted for readability)

```
sip:sales@warehouse.com;method=INVITE \
?Subject=gw%20c2651&Call-ID=c239xa2-as921b%40warehouse.com
sip:jo@example.com?Replaces=abcd@example.com%3B   \
from-tag%3D2%3Bto-tag%3D234ab1&Accept-Contact=   \
%3Csip%3Ajo%40134.102.218.1%3C%3Bonly%3Dtrue
```

Separator characters

# Further Common URI Schemes

Telephony (RFC 2806)

```
tel:+1-555-12345678
tel:7595;phone-context=+49421218
```

ITU-T H.323 Protocol

```
h323:user@example.com
```

Instant Messaging

```
im:user@example.com
```

Presence

```
pres:user@example.com
```





# URIs in Header Fields

## URI-parameters vs. header parameters

```
Contact:   sip:bob@p2.example.com:55060
            ;methods="NOTIFY"
            ;expires=3600
```

?

→ angle brackets:

URI parameter

```
Contact:  <  sip:bob@p2.example.com:55060
            ;methods="NOTIFY" >
            ;expires=3600
```

✓

Header parameter

**Required if**

- URI contains comma, question mark or semicolon
- The header field contains a display name

# Tutorial Overview

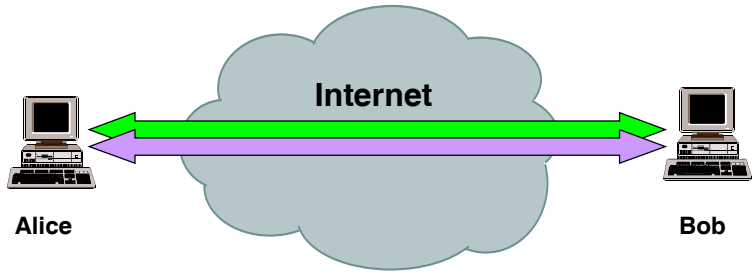
- Internet Multimedia Conferencing Architecture
- Packet A/V Basics + Real-time Transport
- SIP Introduction, History, Architecture
- SIP Basic Functionality, Call Flows
- SIP Security
- SIP Service Creation
- SIP in Telephony
- SIP in 3GPP

You are here



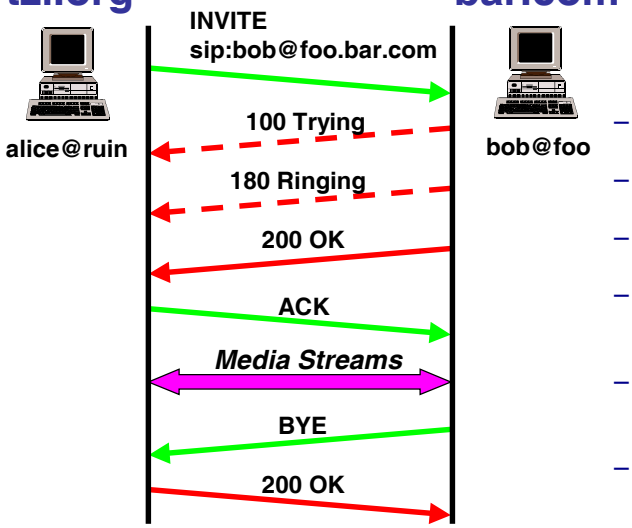


# Application Scenario 1: Direct Call UA–UA




Call signaling
Media streams

# Direct Call



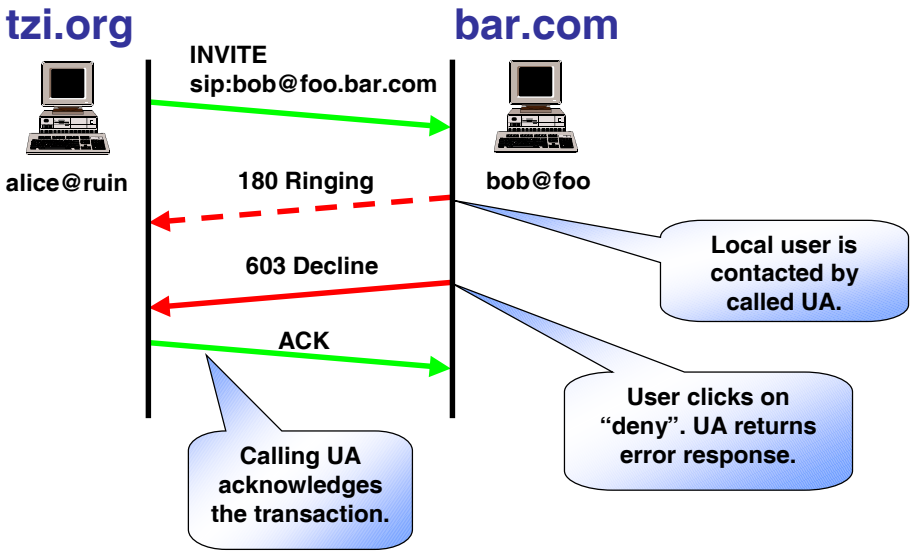

Note:
Three-way handshake is performed only for INVITE requests.

- Caller knows callee's hostname or address
- Called UA reports status changes
- After Bob accepted the call, OK is signaled
- Calling UA acknowledges, call is established
- Media data are exchanged (e. g. RTP)
- Call is terminated by one participant

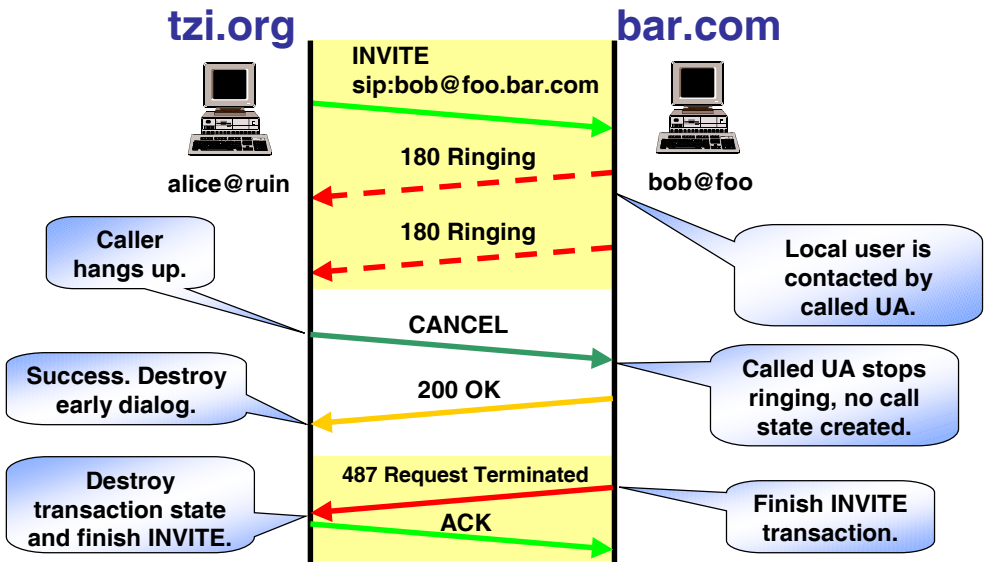




# Callee Declines Call



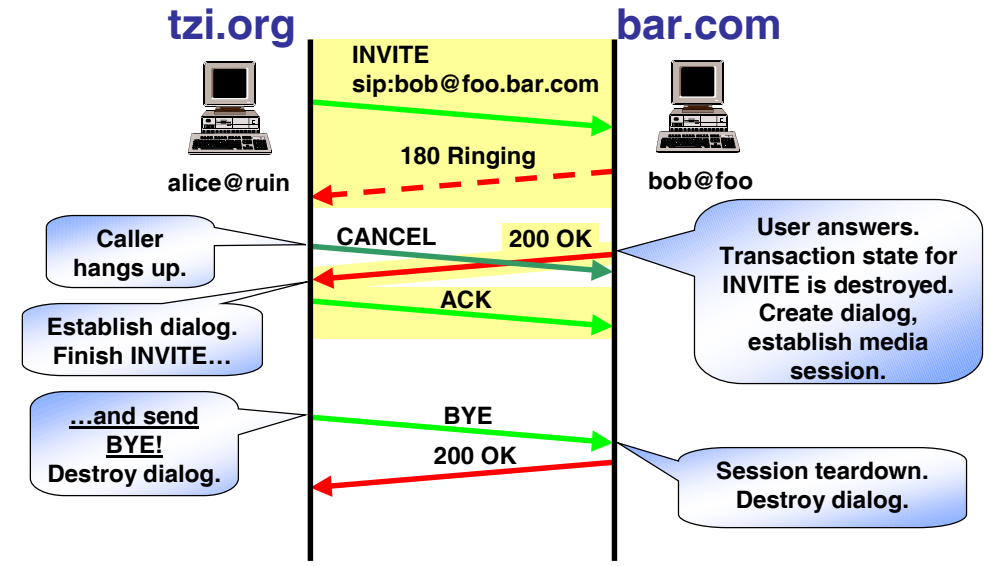
## Caller Gives Up



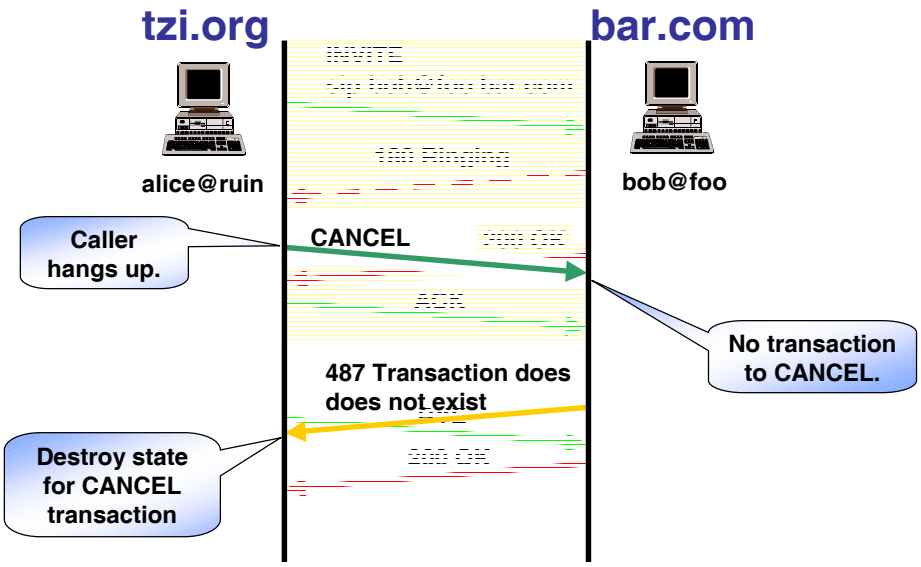





## Caller Gives Up While Call Established



## Caller Gives Up While Call Established



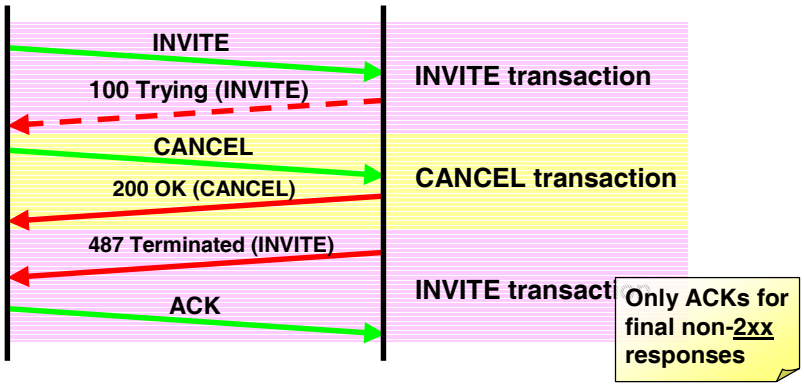





## Atomic Transaction Processing

**Transactions:**

- Created by initial (i.e. not retransmitted) requests
- Terminated by final response to initial request
- Identified by Method, Request-URI, To:, From:, Call-ID:, CSeq: and topmost Via:




**Transactions always complete independently!**

## How to Find The Callee?

- Direct calls require knowledge of callee's address
- SIP provides abstract naming scheme:

**sip:user@domain**

→ Define mapping from SIP URI to real locations:
  - Explicit registration:
    UA registers user's name and current location
  - Location service:
    Use other protocols to find potentially correct addresses
- Caller sends INVITE to any SIP server knowing about the callee's location
- Receiving server may either redirect, refuse or proxy





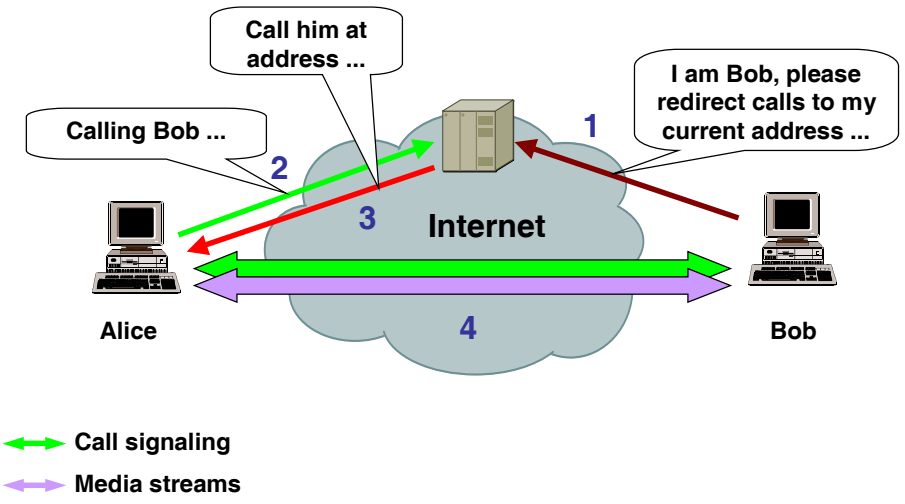
## Finding the Next Hop

- UAC may use a (manually) configured outbound proxy
  - Outbound proxy may also have be learned upon registration
- If request URI contains IP address and port, message can be sent directly

UDP/TCP 5060
TLS 5061

- Otherwise, determine far-end SIP server via DNS
  - Optionally use NAPTR RR
    eventually get ordered list of protocol, IP address, port
  - Query for SRV RR: _sip._udp, _sip._tcp
  - If entries found, try as specified in RFC 2782
- Last resort: query A or AAAA records
  - For specified domain name
  - (Deprecated: For specified *sip.domain* )

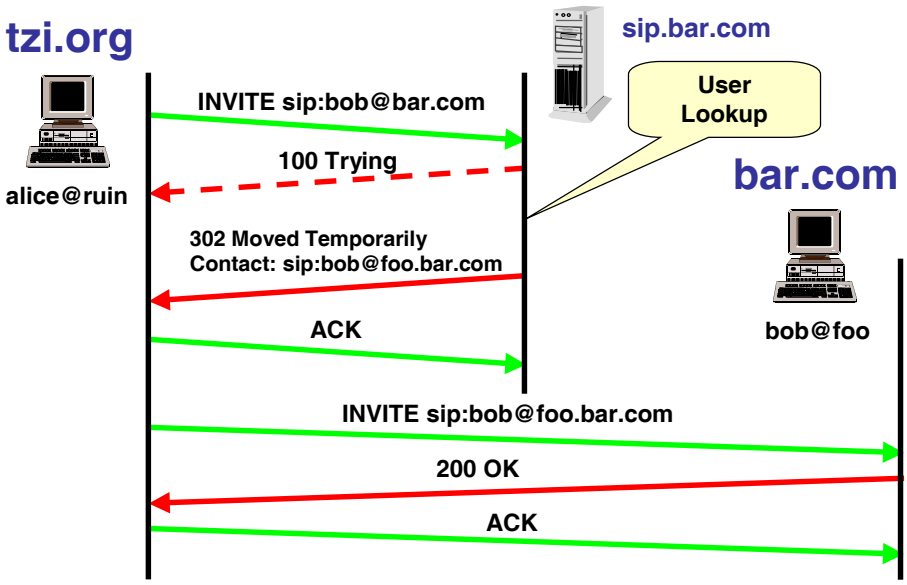
# Application Scenario 2: Redirected Call



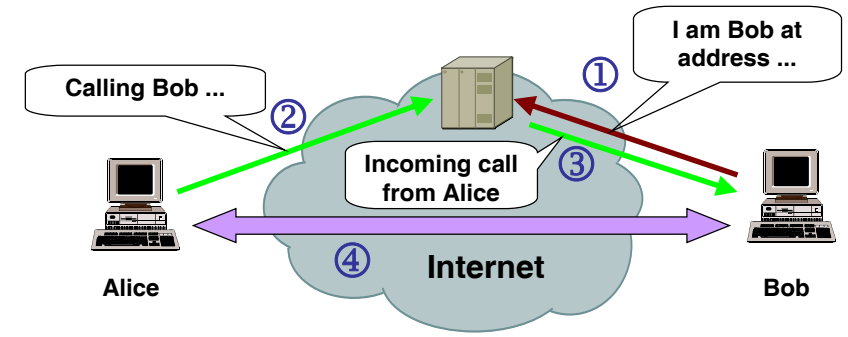





# Redirected Call



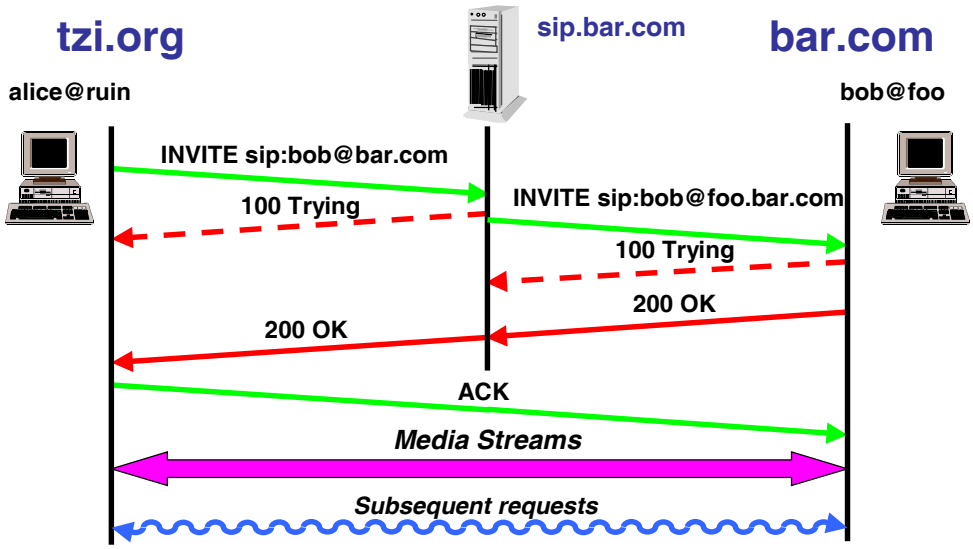
# Application Scenario 3: Proxied Call





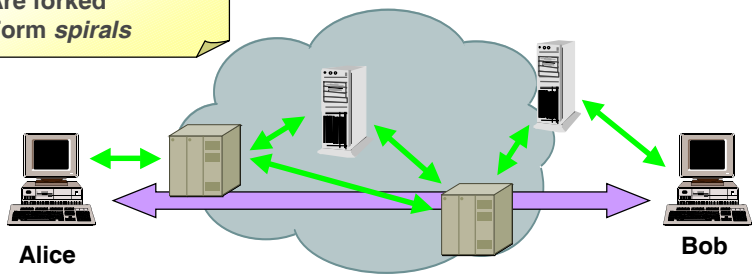



# Proxied Call



# Application Scenario 4: Proxied Call (Real World)

**Requests typically**
- Take different paths
- Are forked
- Form *spirals*




**Responses always** Take the reverse path of the originating request

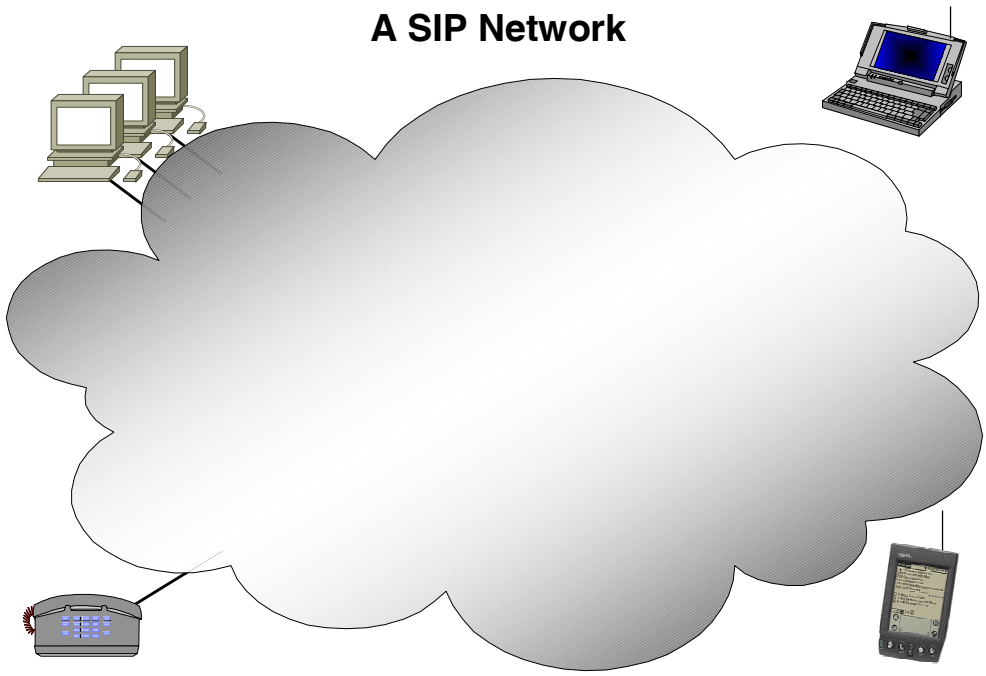
Call signaling

Media streams

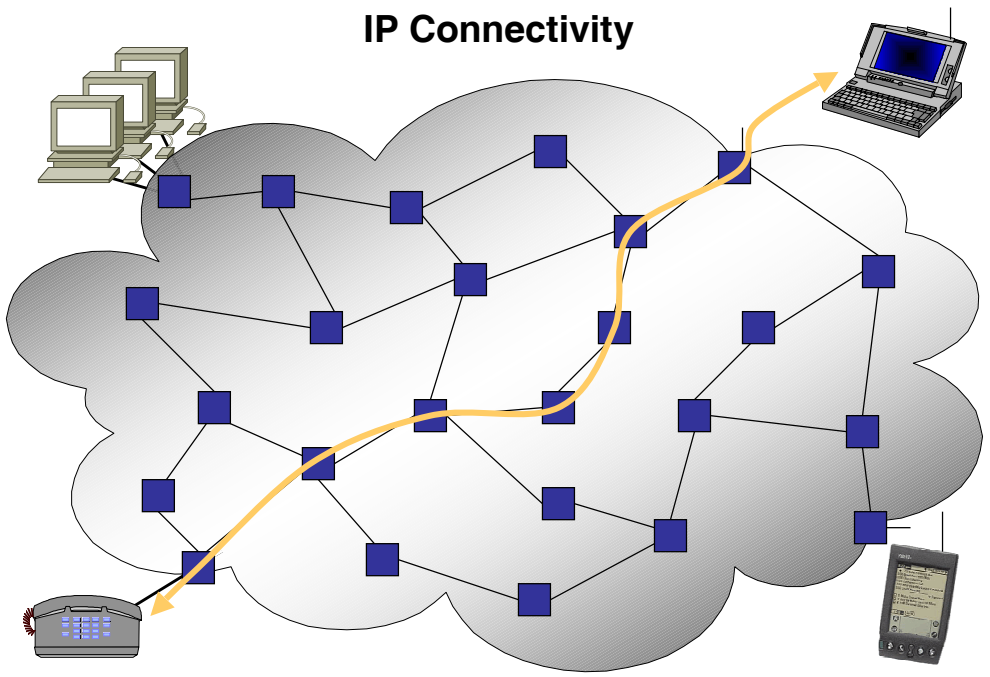




# A SIP Network



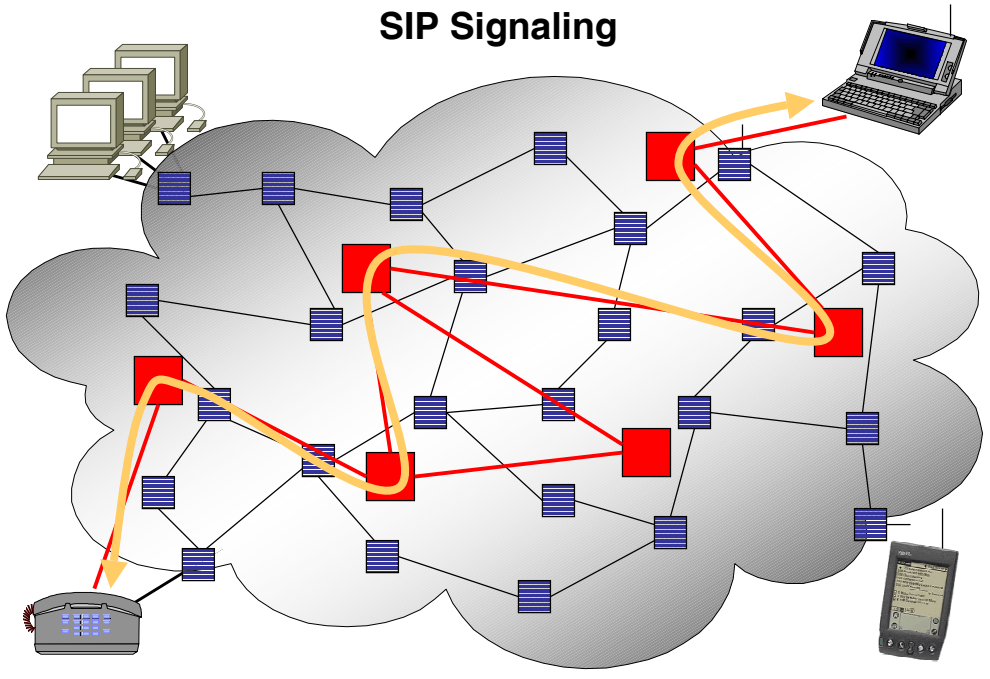
## IP Connectivity



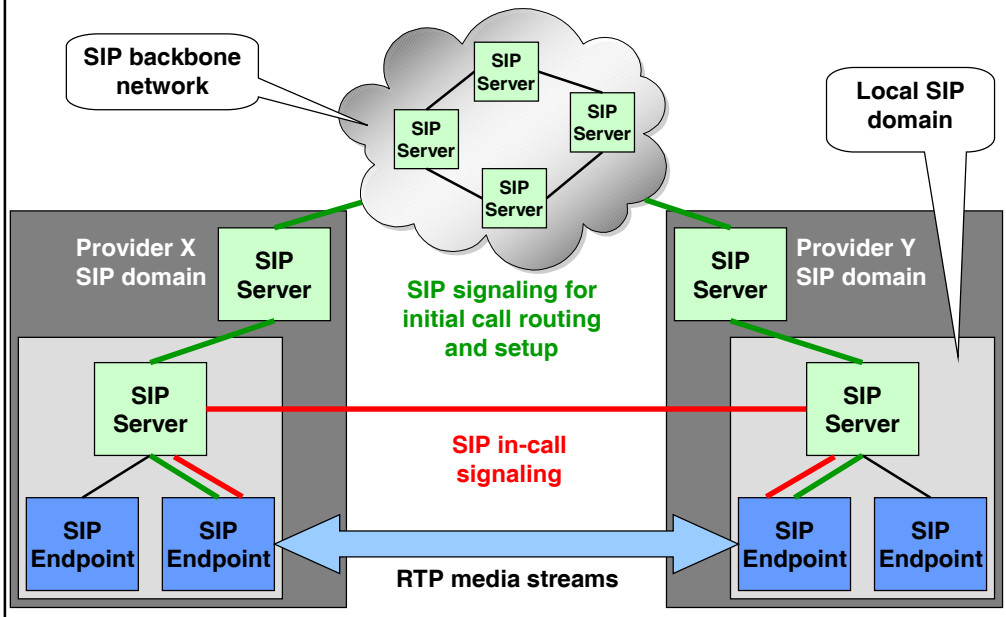




## SIP Signaling



# Global SIP Architecture








# SIP (Proxy) Server Functionality

- **Stateless vs. stateful**
  - Stateless: efficient and scalable call routing (backbone)
  - Stateful: service provisioning, firewall control, ...

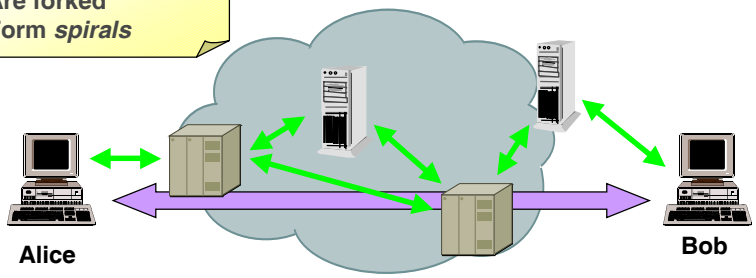
**Some roles for proxies**

- **Outbound proxy**
  - Perform address resolution and call routing for endpoints
  - Pre-configured for endpoint (manually, DHCP, ...)
- **Backbone proxy**
  - Essentially call routing functionality
- **Access proxy**
  - User authentication and authorization, accounting
  - Hide network internals (topology, devices, users, etc.)
- **Local IP telephony server (IP PBX)**
- **Service creation in general...**

More elaborate functions provided by Back-to-Back User Agents (B2BUAs)

# Application Scenario 4: Proxied Call (Real World)

Requests typically
- Take different paths
- Are forked
- Form *spirals*




Responses always Take the reverse path of the originating request

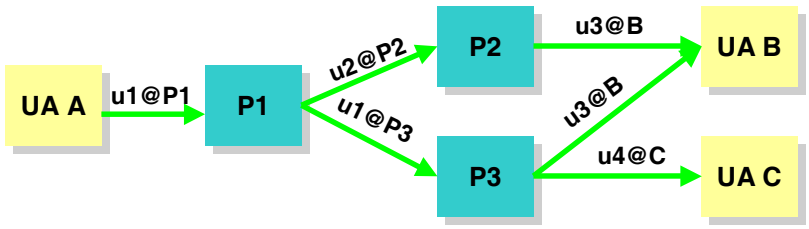
Call signaling

Media streams



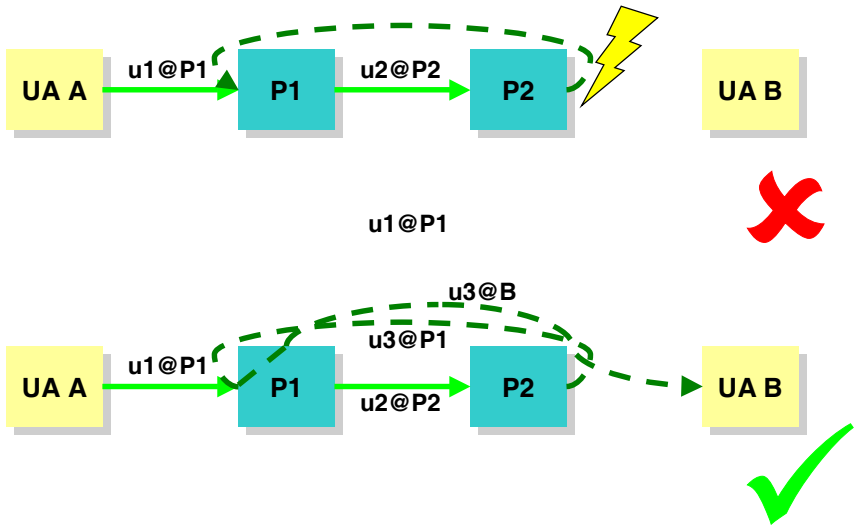


# Request Routing




- Every server determines next hop
- Several destination may be tried in parallel
  - Mark outgoing messages with branch tag
  - Use z9hG4bK to indicate unique branch tag
- Multiple requests can arrive at a single UAS
  - UAS tags To:-header to identify user presence
  - RFC3261: From:-tags mandatory for request merging at UAS

## Loops vs. Spirals





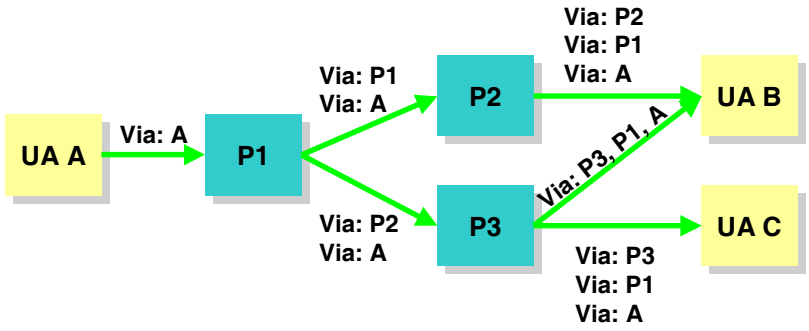



## Response Path

- Proxies may collect state for pending transactions
  - TCP connections
  - Accounting information
  - Parallel search

  → need response to clean up
- Insert Via:-header when forwarding request
- Send response along reverse path of originating request
- Subsequent requests may take different path!

# Creation of Via-path

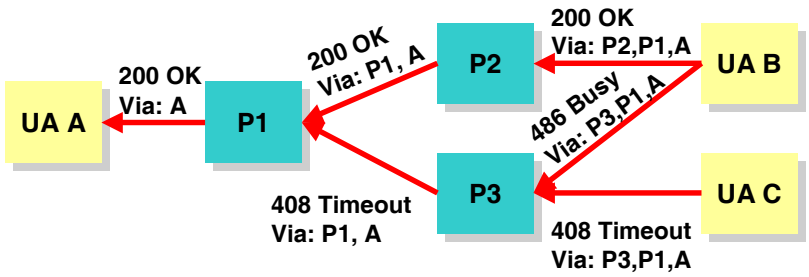



- Send outgoing responses to first Via-entry
- UAS drops responses with different first Via entry
- Add branch-tag to distinguish different search paths





# Response Merging




- Proxy gathers incoming responses until
  - No requests are pending, or
  - Request timers have expired
  - A user-initiated final response arrives (6xx or 2xx)
- Best response is returned to caller, others may be discarded or aggregated (“repairable errors”)
- OK-responses are returned whenever received

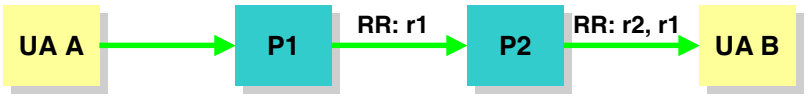
# Record-Route

- Proxies might depend on getting every request of established call
  - Update state information for accounting, call distribution, …
  - Firewalls and NATs
- Record-Route and Route headers for request routing
  - Prepend globally reachable request URI with proxy's address to Record-Route entries
  - Receiving UAS copies contents into response
  - Route header may be initialized with pre-existing route set
- Subsequent requests for this call will contain *source route* created from initial Record-Route header

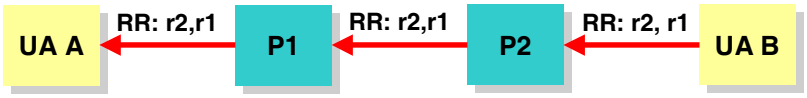




# Constructing the Route

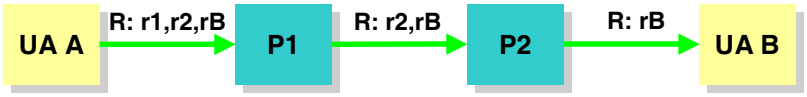



- UA A sends request for UA B to P1
- P1 and P2 add request URIs to Record-Route header
- UA B stores recorded route for later use



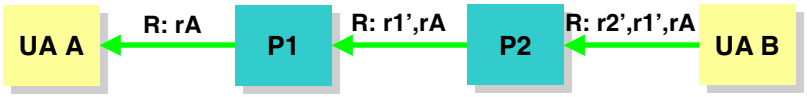

- UA B literally copies record route of request into response

# Using a Recorded Route




- UA A creates new request to B with *reversed* contents of Record-Route from received response in Route-header
- UA A appends B's contact from former response
- Every proxy forwards request according to Route

Significant change from RFC 2543

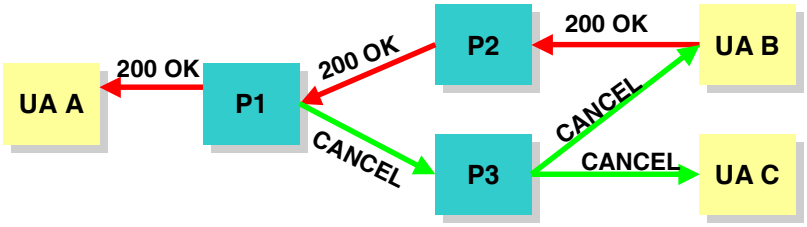



- UA B does not reverse ordering
- Request URIs for proxies are changed to UA A's contact





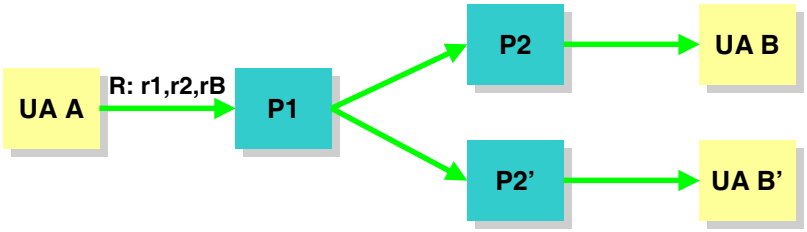
# Cancelling requests




- Cancel a previous request
- Does not affect complete transactions
  → UAS does nothing if final response was already sent
- Used by forking proxies to prune search tree if OK response arrived (see P1)

# Pre-loaded Routes

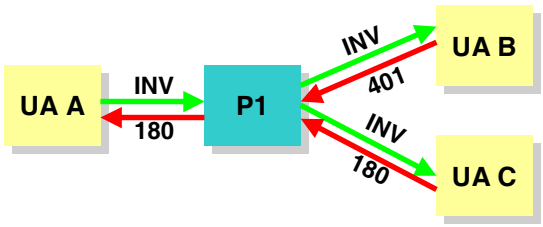
- Requests may contain pre-existing Route set
  → default outbound proxy, CFs in 3GPP
- Loose source routing:
  Forwarding proxy may ignore topmost Route entry
- Route entry rewriting:
  In a response Proxy may rewrite his own Route URI
- Deal with forked routes (due to DNS lookups):





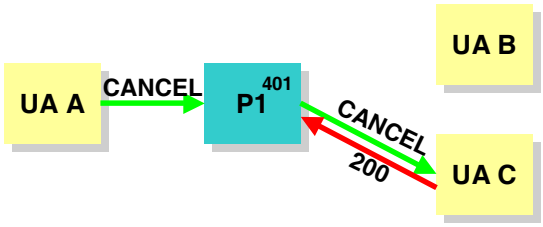



# Heterogeneous Error Response Forking Problem (HERFP) 1




- P1 forks INVITE request
  - UA B stays silent and solicits "401 Unauthorized"
  - UA C begins Ringing (180)
- Provisional response 180 is forwarded upstream
  - UAC sees ringing indication
- No handling for 401 response possible at this time

# Heterogeneous Error Response Forking Problem (HERFP) 2

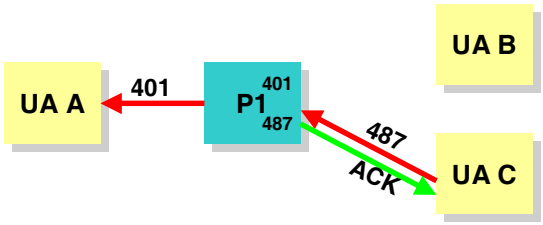



- No answer from C hence A hangs up → CANCEL





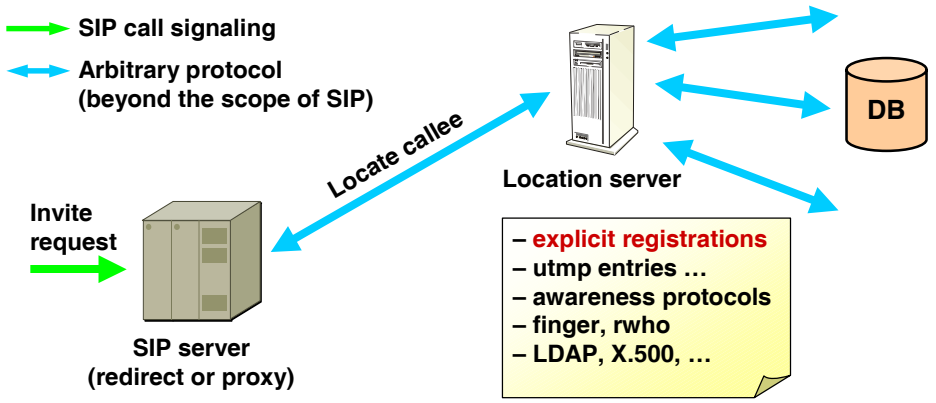
# Heterogeneous Error Response Forking Problem (HERFP) 3




- P1 forwards best response (401) to A
  - Resubmit INVITE with appropriate credentials for B
- Drawbacks:
  - C will ring again
  - large call-setup delay
  - Forked non-INVITE must be idempotent for sequential search

→ Heterogeneous error responses cause problems

# User Location

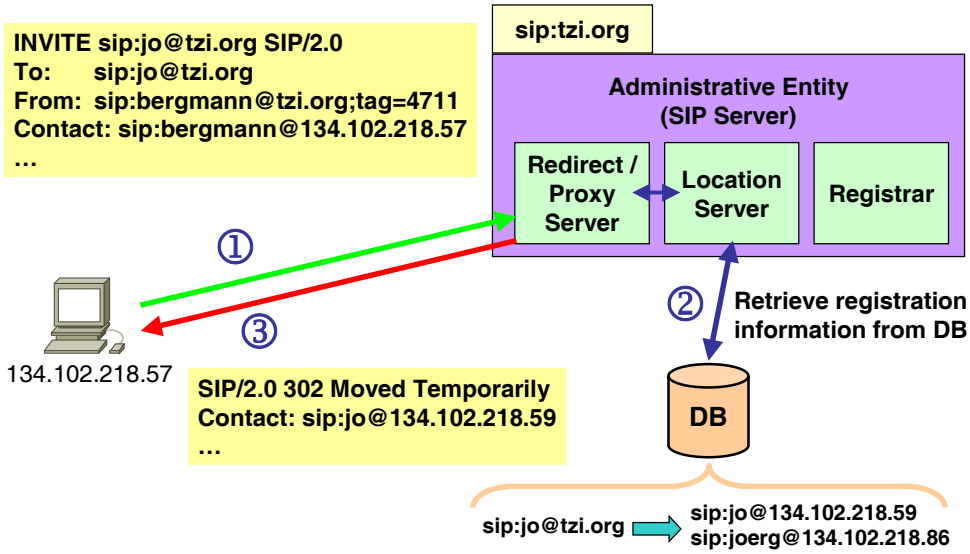



- SIP server asks location server where to find callee
- Location server returns list of contact addresses
- SIP server proxies or redirects request according to address list

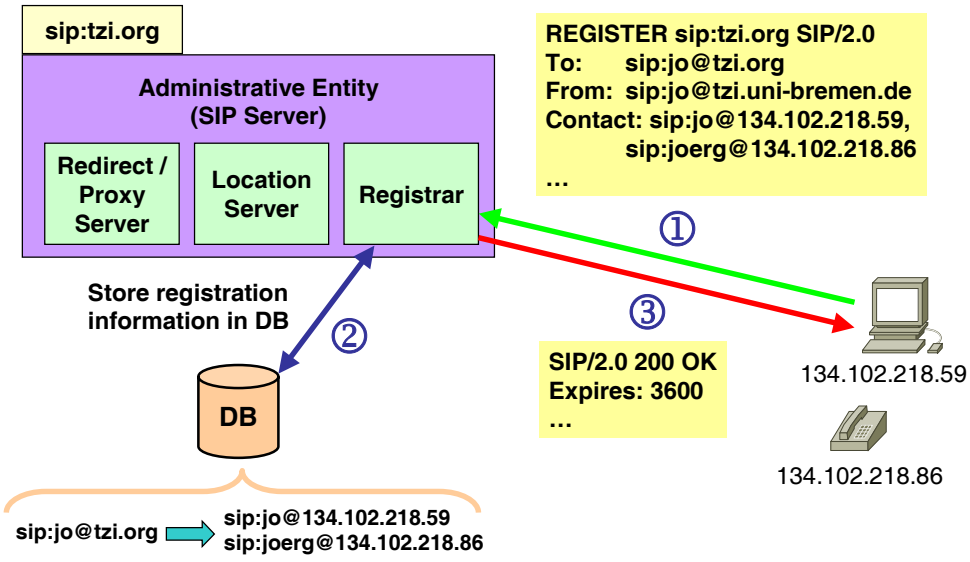




# User Location



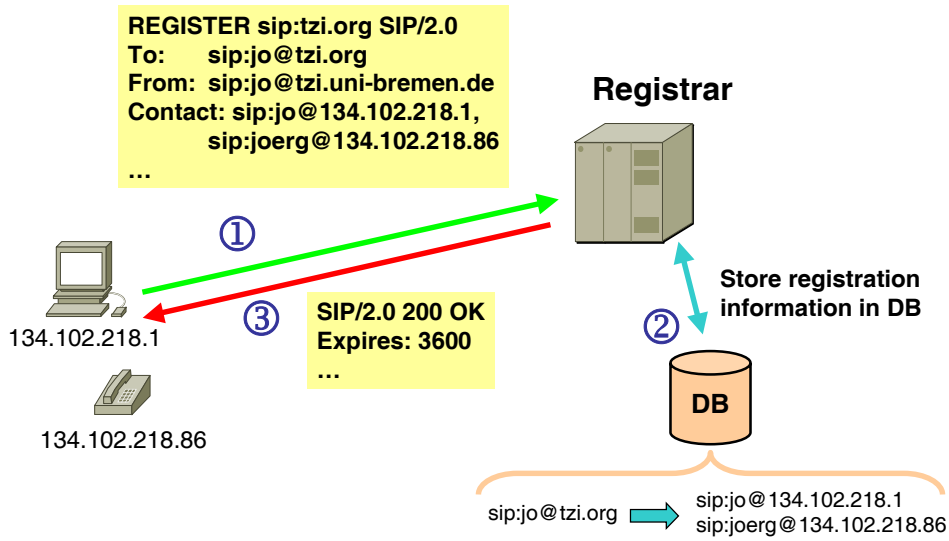
# User Registration








# User Registration (1)



# User Registration (2)

- Send REGISTER request to registrar
- Request URI sip:domain
  → registrar may refuse requests for foreign domains

```
REGISTER sip:tzi.org SIP/2.0
To:      sip:jo@tzi.org
From:  sip:jo@tzi.uni-bremen.de
Contact: sip:jo@134.102.218.1,
         sip:joerg@134.102.218.86
...
```

- To: canonical name for registered user (usually sip:user@domain)
- From: responsible person
  (may vary from To: for third party registration)
- Contact: contact information for the registered user
  → address, transport parameters, redirect/proxy, capabilities
- Specified addresses are merged with existing registrations
- Registrar denotes expiration time in Expires: header
- Client refreshes registration before expiry





# Registration Expiry

- Client requests lifetime
  - Contact:-header parameter expires
  - SIP message header field Expires:
  - Relative duration (seconds) or
    absolute date (RFC 1123, only GMT)
  - Default if no expiry time requested: 3600 seconds
- Registrar may use lower or higher value, indicated in OK response
  - Registrar must not increase expiry interval, may decline request with “423 Registration Too Brief” and Min-Expiry: header
- After expiration, registrar silently discards corresponding database entries

## Add/update Registration Entries

- Retrieve all entries for user@domain (specified in To:-header) from database
- Compare with Contact:-addresses according to scheme-specific rules:
  - Add addresses for which no entries exist
  - Entries being equal to a contact address are updated
- Otherwise return 200 OK response
  - Include all entries for user@domain





## Lookup and Delete

- Lookup entries:
  - No Contact:-header in request
    → Current registrations are returned in OK response
  - For further processing:
    - Client uses q-parameter to determine relative order
- Delete entries:
  - Expires: 0
    - Delete URIs specified in Contact:-Header from database
    - Delete all entries for user: Contact: *

# Extended Registration Functions (1)

- Registrar is the initial point of contact for a UA
- Idea: provide (user-specific) configuration and other information in REGISTER response
  - Allows for centralized control using standard SIP mechanisms

## 1. Service Route Discovery

- Dynamically configure “home service proxies” for UAs
  - Proxies to be traversed first for outbound requests
  - Routing of incoming requests can be enforced by infrastructure
  - Routing of responses implied from Via: headers
- Simple extension header returned by registrar

  Service-Route: <sip:sp1.example.com;lr>, <sip:sp2.example.com;lr>
  - Syntax similar to Route: header
- May be used by UA subsequently for loose source routing
  - Included in Route: header





# Extended Registration Functions (2)

## 2. Globally Routable UA URI (GRUU)

- Users may be registered at multiple UAs
- Incoming SIP requests may be forked to all UAs
- May need to direct a request to a particular UA
  - Examples: Call Transfer, Conferencing, Presence
- Additional parameter in Contact: header from registrar

  Contact: <sip:jo@client-42.tzi.org>;gruu="sip:dsiah-0=@tzi.org"
  - Valid for the lifetime of this UA’s registration
  - Local proxies ensure proper routing to the right UA
- UA may place GRUU in Contact: header in later requests
  - UA uses grid parameter to distinguish individual uses of a GRUU
- Remote UAs will use GRUU to send messages
  - grid parameter will be returned in the Request-URI from remote UA

# Finding a Local Registrar Server

- Multicast REGISTER request
  - Send link-local request to sip.mcast.net (224.0.1.75)
  - Use address of first server that responds with OK
  - If other OK responses, use sender addresses as fallback
- Alternatively, use configured registrar address

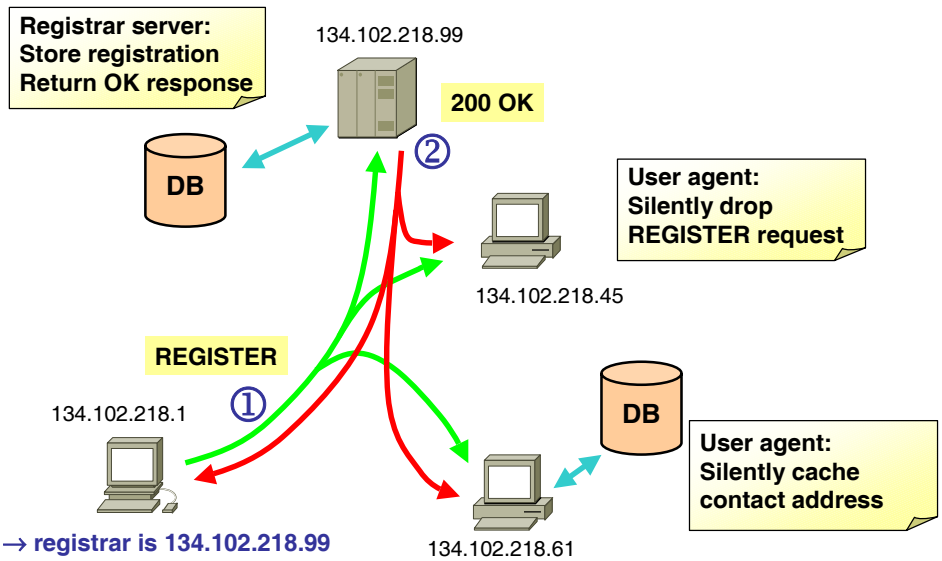
Other mechanisms out-of-scope of core spec. Examples:

- DHCP
  - DNS SRV lookup on domain name obtained via DHCP
  - If no SIP server found, query A or AAAA record
- Service Location Protocol (SLP)
- …





# Link-local Multicast Registration



# Media Session Setup Between Endpoints

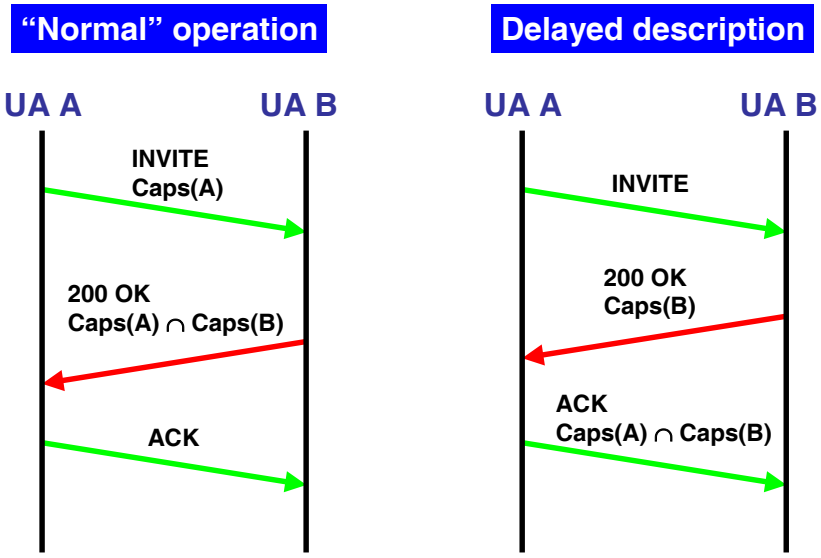




## Capability Negotiation

- SDP: Session Description Protocol, RFC 2327
- Caller includes SDP capability description in INVITE
  - Time information may be set to "t=0 0" or omitted
  - For RTP/AVT, use of *rtpmap* mappings is encouraged
- For each media stream (*m*-part of SDP message), callee returns own configuration in response
  - Indicate destination address in *c*=-field
  - Indicate port and selected media parameters in *m*=-field
  - Set port to zero to suppress media streams
- UA may return user's capability set in 200 OK response when receiving an OPTIONS request

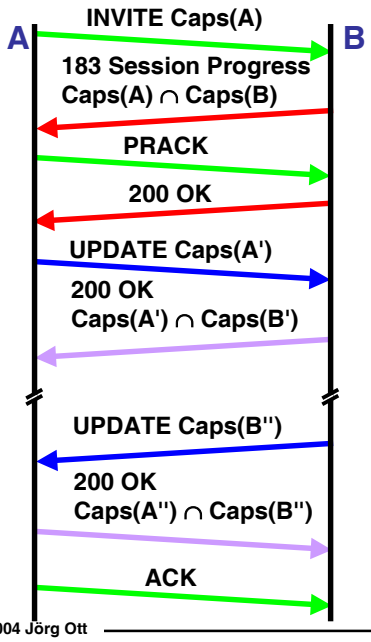
# Media Negotiation During Call Setup








# Use of UPDATE in Offer/Answer Model




- Offerer sends session description
- Answerer must match against local capabilities
- Mid-dialog UPDATE request:
  - establish dialog state first (final or reliable provisional response)
  - No effect on dialog state
- No offer allowed as long as pending offers exist
- Any party may send offer

# SDP Example

SDP Version

```
v=0
o=llynch 3117798688 3117798739 IN IP4 128.223.214.23
s=UO Presents KWAX Classical Radio
i=University of Oregon sponsored classical radio station KWAX-FM
u=http://darkwing.uoregon.edu/~uocomm/
e=UO Multicasters multicast@lists.uoregon.edu
p=Lucy Lynch (University of Oregon) (541) 346-1774
t=0 0
a=tool:sdr v2.4a6
a=type:test
m=audio 30554 RTP/AVP 0
c=IN IP4 224.2.246.13/127
a=ptime:40
```

Session Level

Media Level





# SDP Example

Originator + version

```
v=0
o=llynch 3117798688 3117798739 IN IP4 128.223.214.23
s=UO Presents KWAX Classical Radio
i=University of Oregon sponsored classical radio station KWAX-FM
u=http://darkwing.uoregon.edu/~uocomm/
e=UO Multicasters multicast@lists.uoregon.edu
p=Lucy Lynch (University of Oregon) (541) 346-1774
t=0 0
a=tool:sdr v2.4a6
a=type:test
m=audio 30554 RTP/AVP 0
c=IN IP4 224.2.246.13/127
a=ptime:40
```

Session Level

Media Level

# SDP Example

**Media Name and Transport Address**

```
v=0
o=llynch 3117798688 3117798739 IN IP4 128.223.214.23
s=UO Presents KWAX Classical Radio
i=University of Oregon sponsored classical radio station KWAX-FM
u=http://darkwing.uoregon.edu/~uocomm/
e=UO Multicasters multicast@lists.uoregon.edu
p=Lucy Lynch (University of Oregon) (541) 346-1774
t=0 0
a=tool:sdr v2.4a6
a=type:test
m=audio 30554 RTP/AVP 0
c=IN IP4 224.2.246.13/127
a=ptime:40
```

Session Level

Media Level





# SDP Example

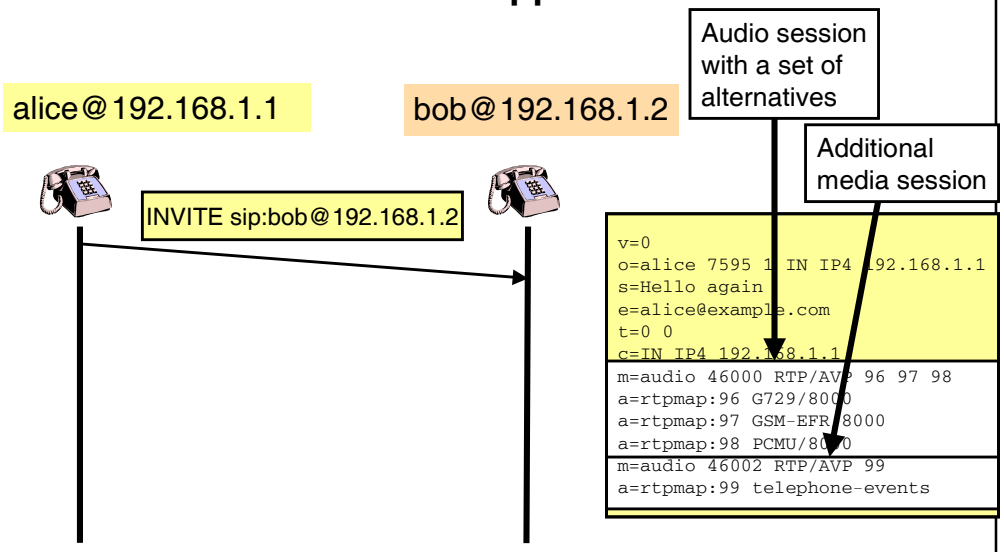
**Connection Information**
(In SIP: address where originator wants to receive data)

```
v=0
o=llynch 3117798688 3117798739 IN IP4 128.223.214.23
s=UO Presents KWAX Classical Radio
i=University of Oregon sponsored classical radio station KWAX-FM
u=http://darkwing.uoregon.edu/~uocomm/
e=UO Multicasters multicast@lists.uoregon.edu
p=Lucy Lynch (University of Oregon) (541) 346-1774
t=0 0
a=tool:sdr v2.4a6
a=type:test
m=audio 30554 RTP/AVP 0
c=IN IP4 224.2.246.13/127
a=ptime:40
```

Session Level

Media Level

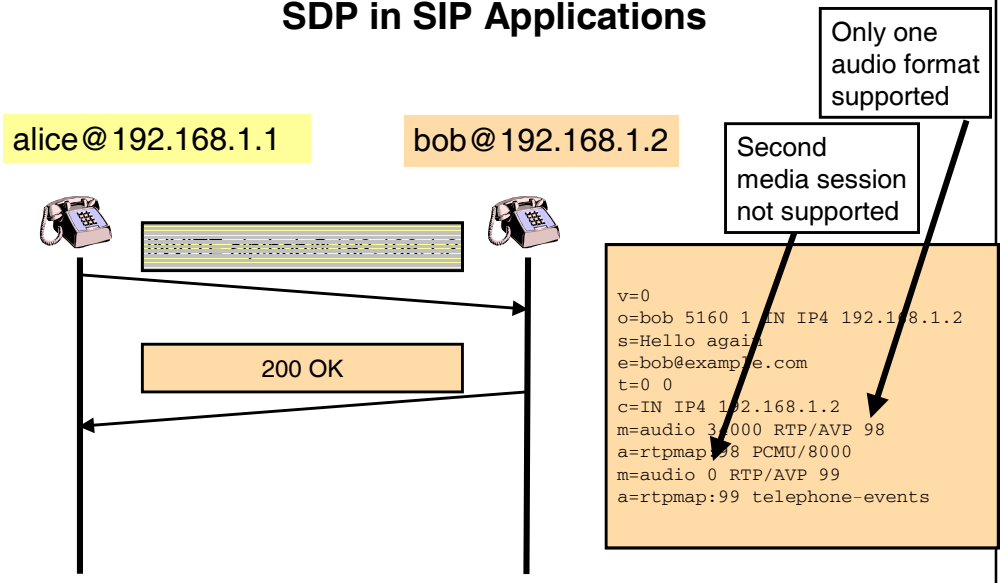
# SDP in SIP Applications





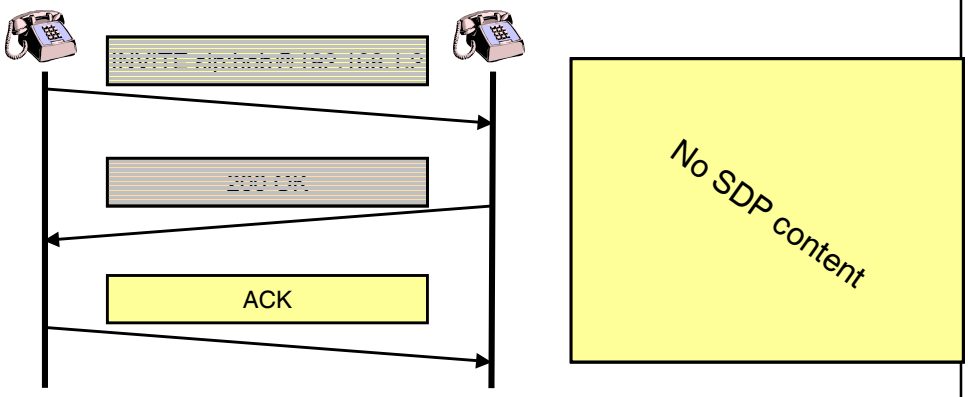



# SDP in SIP Applications



# SDP in SIP Applications

alice@192.168.1.1 bob@192.168.1.2





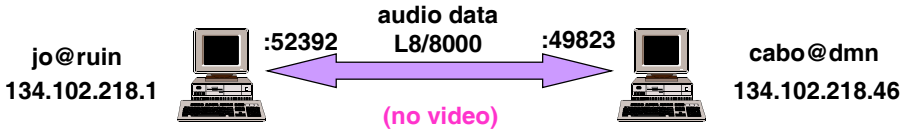



# Example SDP Alignment

```
v=0
o=jo 7849 2873246 IN IP4 ruin.inf…
s=SIP call
t=0 0
c=IN IP4 134.102.218.1
m=audio 52392 RTP/AVP 98 99
a=rtpmap:98 L8/8000
a=rtpmap:99 L16/8000
m=video 59485 RTP/AVP 31
a=rtpmap:31 H261/90000
```

```
v=0
o=cabo 82347 283498 IN IP4 dmn.inf…
s=SIP call
t=0 0
c=IN IP4 134.102.218.46
m=audio 49823 RTP/AVP 98
a=rtpmap:98 L8/8000
m=video 0 RTP/AVP 31
```

## Resulting configuration:



## Send/Receive Only

- Media streams may be unidirectional
  - Indicated by *a=sendonly*, *a=recvonly*
- Attributes are interpreted from sender's view
- sendonly
  - Recipient of SDP description should not send data
  - Connection address indicates where to send RTCP receiver reports
  - Multicast session: recipient sends to specified address
- recvonly
  - Sender lists supported codecs
  - Receiver chooses the subset he intends to use
  - Multicast session: recipient listens on specified address
- inactive
  - To pause a media stream (rather than deleting it)





## Change Session Parameters

- Either party of a call may send a re-INVITE that contains a new session description
  - Use other connection address, port
  - Add/remove codecs
  - Append new media streams at the message's end
- Recipient re-aligns session description with current values
  - Change media parameters
  - Delete media streams (port has zero-value)
  - Add new streams

→ Gateways may use re-INVITE when session parameters are unknown during call setup

# UPDATEs for Session State (RFC 3311)

- **UPDATE** as generic mechanism to modify session state
  - Align different proposals: early media, resource reservation, ...
  - Issues with heterogeneous error responses (HERFP) addressed
  - Even useful to establish security associations
  - Dialog state vs. session state

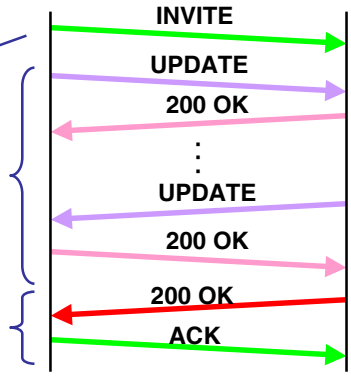
- Offer/Answer-Model

Request dialog state update

Request session state update

Multiple O/A cycles possible
No overlapping offers
O/A in 183, PRACK etc. allowed
→ Tight relationship between offer and answer necessary

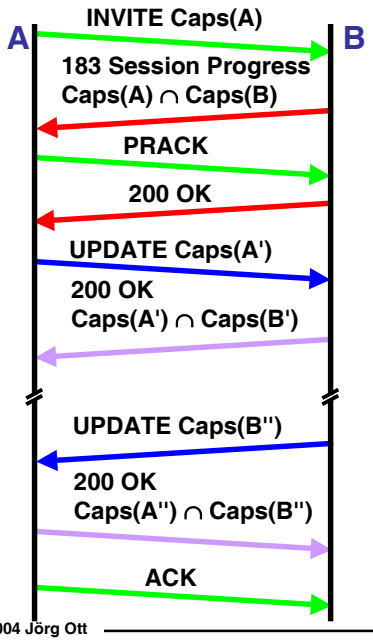
Finish dialog state update








# Use of UPDATE in Offer/Answer Model




- Offerer sends session description
- Answerer must match against local capabilities

- Mid-dialog UPDATE request:
  - establish dialog state first (final or reliable provisional response)
  - No effect on dialog state

- No offer allowed as long as pending offers exist
- Any party may send offer

## Tutorial Overview

- Internet Multimedia Conferencing Architecture
- Packet A/V Basics + Real-time Transport
- SIP Introduction, History, Architecture
- SIP Basic Functionality, Call Flows
- SIP Security
- SIP Service Creation

You are here

- SIP in Telephony
- SIP in 3GPP





## SIP and Security

SIP entities are potential target of a number of attacks, e.g.

- Spoofing identity
- Eavesdropping
  - Media streams
  - Call signaling
- Traffic analysis
- Theft of service
- Denial of service (DoS)

Typical threats for distributed applications using the public Internet

Some countermeasures:

- Client and server authentication
- Request authorization
- Encryption
- Message integrity checks + replay protection

Reuse existing security mechanisms.

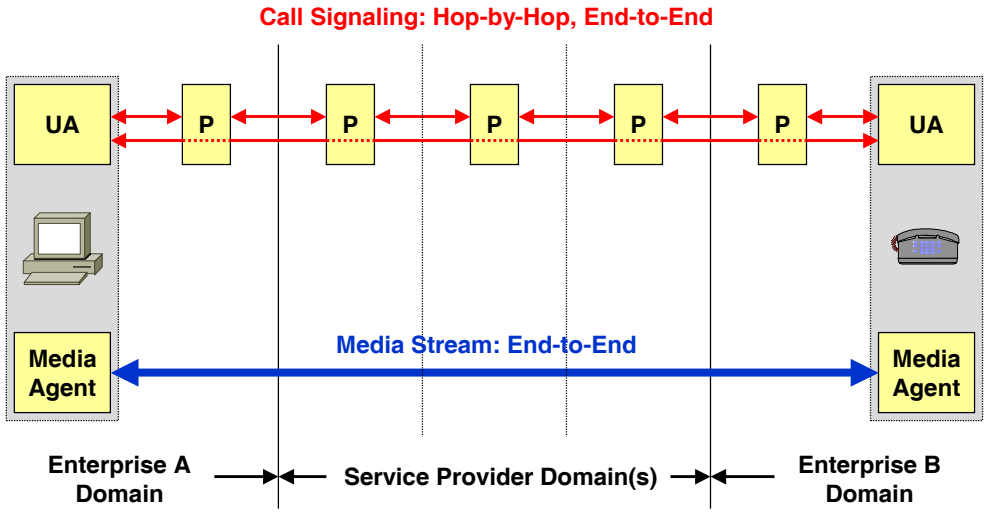
# Why SIP Security?

- **Ensure privacy**
  **(media encryption, anonymous calls, personalized services, ...)**
- **Billing and accounting**
  **(probably pay for services, assured bandwidth, etc.)**
- **Regulatory requirements**
  - Call id blocking
  - Prosecute abuse (call tracing facility)
  - Emergency call service
  - Multi-Level Priorization and Preemption





# SIP Security Overview



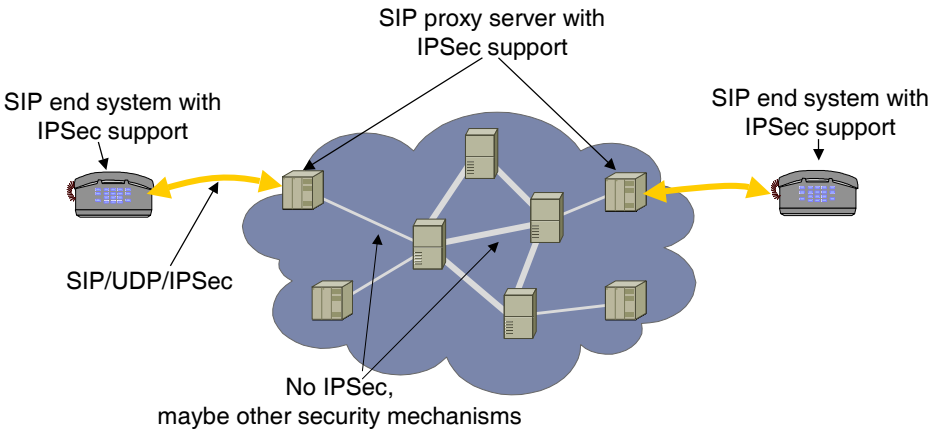
# Hop-by-hop Encryption of SIP Messages

- Lower layer mechanisms
  - Applicability depends on link layer technology
- VPN-like tunnel using IPSec
  - Suitable e.g. for coupling sites of a company
  - Need OS-support (required for IPv6 anyway)
- SIP over TLS (Transport Layer Security)
  - Access to outbound proxy
  - Call routing to ITSP
  - Call routing between neighboring ITSPs (agreements!)
  - In most cases, only servers have certificates
- Chain of trust: suitable also for authentication
  - e.g. in trusted networks



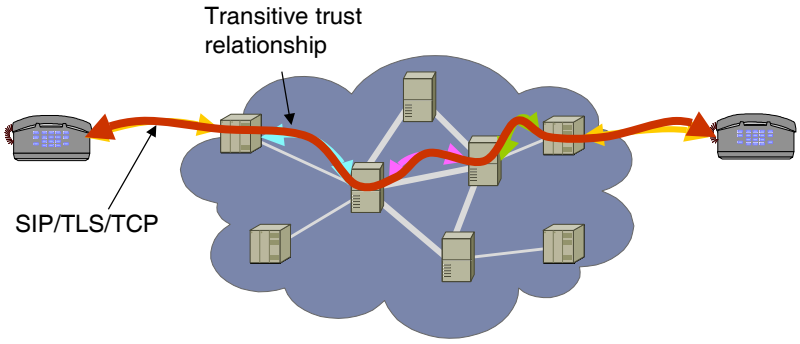


# Hop-by-hop encryption with IPSec




- Possible deployment scenario
  - IPSec between hosts inside an administrative domain
  - Established trust relationships, pre-shared keys
- Security functions independent of SIP layer

# SIP and TLS




- Hop-By-Hop security
  - TLS secures connections between SIP entities
  - TCP only!
- Useful for hosts with no pre-existing trust association
- Usage of TLS is coupled with SIP
  - Needs to be signalled, e.g, in Via headers
  - MUST be implemented by proxy servers, redirect server and registrars
- Basic model
  - A UA establishes a trust association with his outbound proxy (TLS connection)
  - The proxy build trust associations with other proxies/UAs
    - May require certificate exchange and validation
- Problems
  - Call setup delays
  - Resources for open TLS connections





# End-to-End Encryption of SIP Messages

- Impossible for entire message: proxies do call-routing
- Use SIP message bodies to carry confidential information
  - Can be used to achieve end-to-end encryption
    - E.g., when the network cannot be trusted or other mechanisms are not available
  - Does only protect the message body, not the header fields
- S/MIME
  - Allows to encapsulate arbitrary content for security purpose
  - Encryption and digital signature
- Issue: Key distribution problem *prior to call*
  - No global PKI
  - Certificates for SIP proxies, typically not for users
  - Typically no shared secrets
  - Validation of end user certificates will have to rely on public key infrastructures (or pre-configured certificates)

## SIP and S/MIME for End-to-End Enryption

- Carry (partial) SIP messages in S/MIME body
  - Content-Type: message/sipfrag (or: message/sip)
  - Apply MIME security to message body
  - Use multiparty/mixed if necessary
    - e.g. to also carry SDP bodies
- Include confidential headers in message body
  - E.g. From:, Contact:, Subject:
- Keep headers required for routing "outside" in the clear
  - Some outer header fields remain visible
    - To, From, Call-ID, Cseq, …
  - Some mandatory headers may be populated with dummy values
  - Some outer header fields may also be changed in transit
    - Request-URI, Via, Record-Route, …
- May also be used for end-to-end authentication





## SIP Media Privacy

- Encryption of (RTP) media streams
  - use old RTP encryption scheme
  - use Secure RTP (SRTP) profile
    - Currently finalized within the IETF
- Secure key distribution between endpoints *in a call*
- Original SDP allows only for one per media key field ("k=")
- SDP extensions for better keying support
  - Requires encrypted SDP in SIP message body
  - Requires protected communication path
- Further SDP extensions for secure media keying
  - MIKEY allows for end-to-end negotiation of keys
  - Protection of the exchanged information within SDP

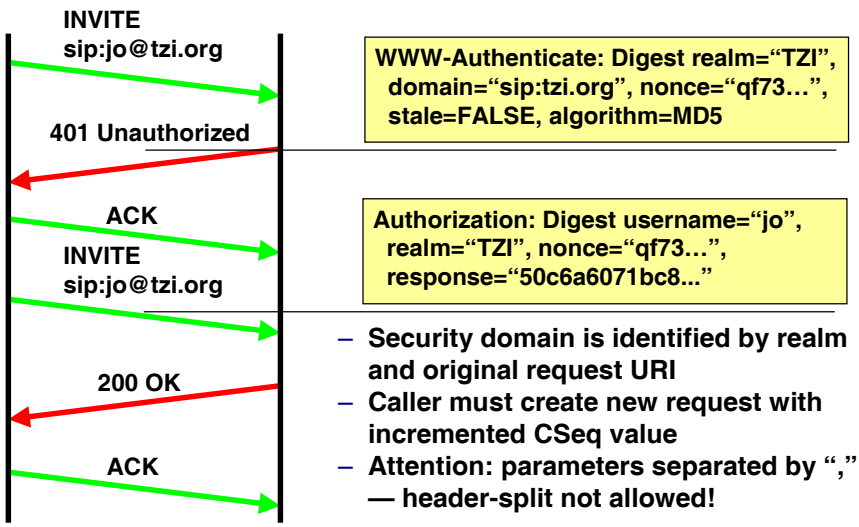
# Simple SIP Authentication

- HTTP *digest* authentication
  - basic authentication deprecated
- Challenge with WWW-Authenticate: header field
- Authorization: header field
  - End-to-end authentication of UAC
  - Digitally sign message body and header fields following the *Authorization:* header
  - Use canonical form
  - Fields to be changed must precede *Authorization*





# Authentication: Example Call Flow




- Security domain is identified by realm and original request URI
- Caller must create new request with incremented CSeq value
- Attention: parameters separated by "," — header-split not allowed!

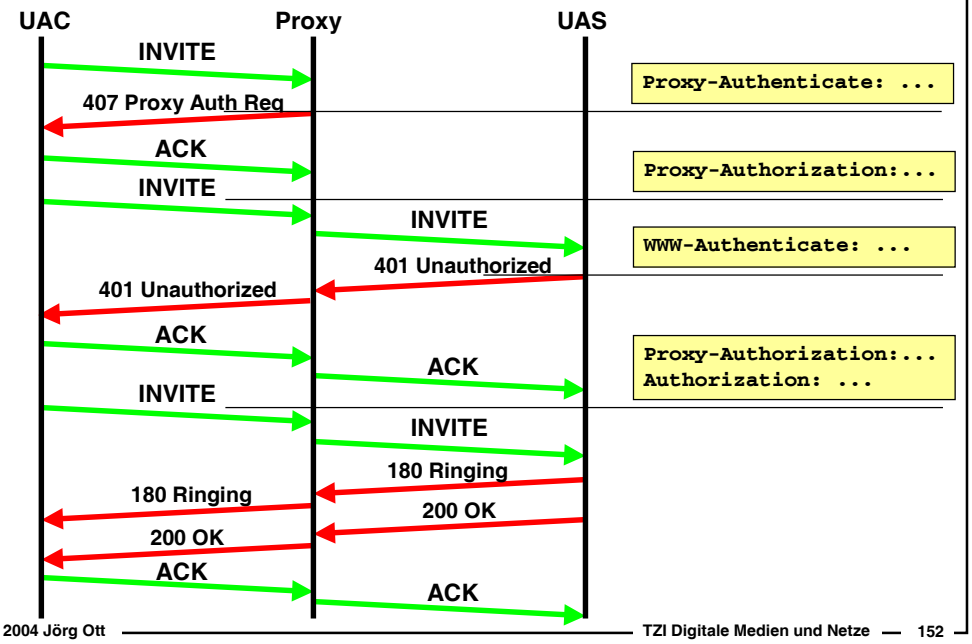
# Authentication for Proxies

- Similar to endpoints (HTTP Digest)
- Proxy rejects client request with “407 Proxy Auth required”
  - Proxy-Authenticate: header
  - Multiple proxies along the path may challenge
- Client resubmits request with credentials for Proxy
  - in Proxy-Authorization: header
  - Multiple headers with credentials may need to be included



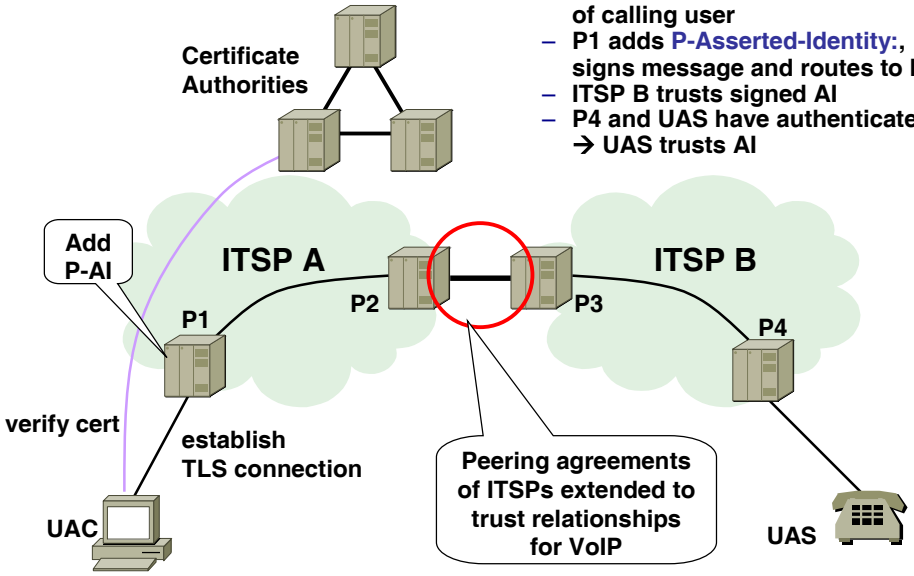


# Multiple Authentication Steps



# Transitive Trust Relationships

- UAC establishes TLS connection
  - Proxy provides certificate
  - UAC verifies cert
- Password-based authentication of calling user
- P1 adds P-Asserted-Identity:, signs message and routes to P2
- ITSP B trusts signed AI
- P4 and UAS have authenticated
  → UAS trusts AI








# Privacy of Calling Party Information

- Option-tag privacy to enforce treatment at proxies
- Extension header Privacy:
  - Initiator can specify a required level of privacy
  - Typically, intermediary has to obscure certain header fields
    → B2BUA
  - Different types of privacy information (header, session, user)
- Use asserted identity mechanism if authentication required
  → need trust relationships between ITSPs

## SIP and S/MIME for End-to-End Authentication

- Header fields are included in a signed S/MIME message body
  - Idea similar to encryption: message/sip, message/sipfrag
  - Include enough headers to prevent replay
    - From:, Date: Contact: (To:, Call-ID:, CSeq:)
- UA may sign the message itself: straightforward
  - Workable -- but receiver has to obtain UA's certificate
  - Work in progress to locate Certificate servers per domain
- UA may have an authentication server sign on its behalf
  - Requires (local) trust relationship between UA and auth server
  - Requires only auth server's certificate to be known
- Authenticated Identity Body (AIB)
  - Third party ensures receiver about sender identity
  - UA prepares and includes the above message body
  - Asks auth server to sign
  - Indicated by: Content-Disposition: message/aib





## Tutorial Overview

- Internet Multimedia Conferencing Architecture
- Packet A/V Basics + Real-time Transport
- SIP Introduction, History, Architecture
- SIP Basic Functionality, Call Flows
- SIP Security
- SIP Service Creation ← You are here
- SIP in Telephony and Deployment Issues
- SIP in 3GPP

## Basic Extension Mechanisms

- Proxies forward unknown methods and headers
- UAS' ignore unknown headers, reject methods

- Feature negotiation
  - Headers: Require, Proxy-Require, Supported
  - Option tags for feature naming (see below)
  - Error responses:
    - 405 Method not allowed
    - 420 Unsupported
    - 421 Extension Required

  } UAC and UAS/proxy must agree on common feature subset

- Option tags
  - Identified by unique token
  - Prefix reverse domain name of creator
  - IANA: implicit prefix *org.ietf.*





## Some Current SIP Extensions

- Reliable provisional responses
- Session Timers
- Early Media
- Adjusting session state: UPDATE
- INFO method
- REFERing peers to third parties
- SIP for subscriptions and event notifications
- Instant messaging
- SIP for presence
- ...

- Addressed later today where appropriate...

## Reliable Provisional Responses

- Signaling Gateways, ACD systems etc. may depend on provisional responses
  - Such as 180 Ringing

→ need optional reliability

- Option tag 100rel:
  - End-to-end reliability of provisional responses > 100
  - Retain order of reliable responses
- Implementation:
  - Windowing: RSeq/RAck headers
  - Explicit acknowledge: PRACK request

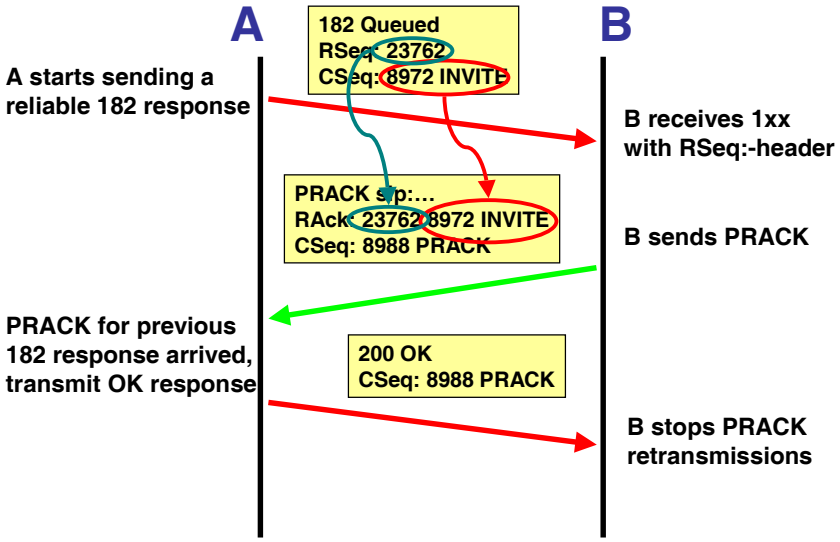




## Use of PRACK

- Insert RSeq:-header with random integer value
  - new responses must increment RSeq: by 1
- Retransmit with exponential backoff
- No subsequent reliable provisional responses until first PRACK
  - enables re-ordering at receiver side
- PRACK request
  - RAck: contains RSeq: and CSeq: values to acknowledge
  - May contain body with additional information

## Example Call Flow: PRACK



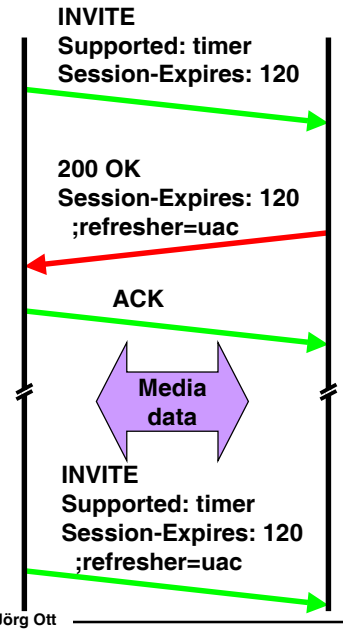





## Session Timers

- No keep-alive mechanism for SIP calls
  - Proxies may preserve outdated state information
  - Close dynamically created holes in firewalls
  - Need timeout-mechanism to perform cleanup
- SIP Extension for Session timers
  - Option tag timer
  - Negotiation mechanism for expiry time and refresh responsibility
    - Session-Expires: duration and role
    - 422 response with Min-SE: lower bound for expiry interval
  - *Responsible UA* sends re-INVITE before timer expires
- Session is terminated if no re-INVITE/UPDATE arrives in time
  - *Responsible UA* sends BYE
  - Proxies silently drop state information

# Example: Keep-alive

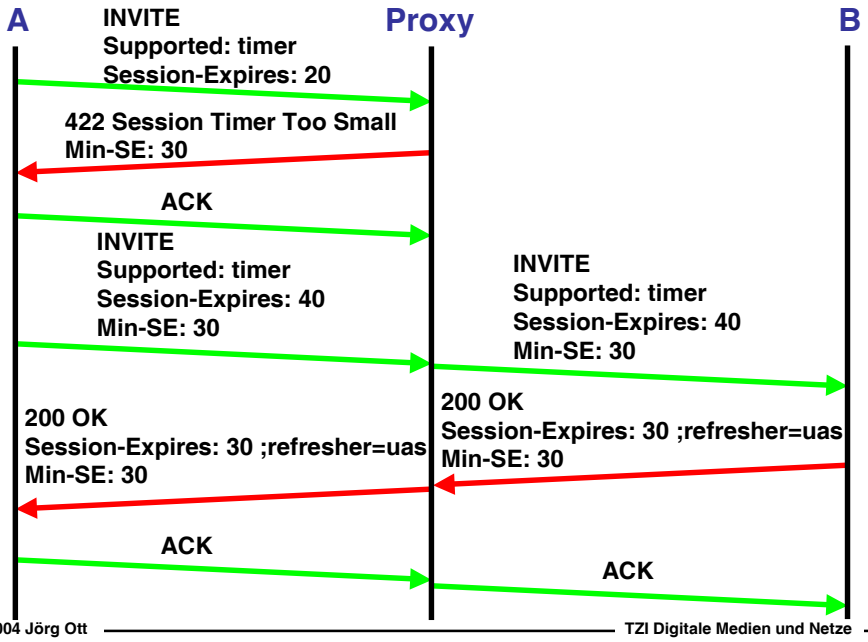



- Caller indicates support of session timers in INVITE
- Propose initial timer value
- Called UA supports timers, accepts proposed value, asks the caller to send refresh
- Setup complete, media streams are established
- After n/2•timer calling party sends re-INVITE
- Call continues after 200 OK and ACK





# Example: Interval Negotiation



# Event Notifications

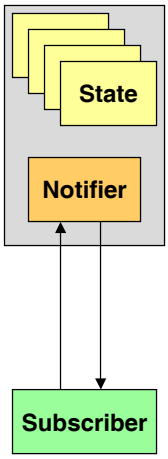
- Need for flexible event notification
  - Enable presence information
  - Better support for mobile SIP applications
  - Feedback about progress of other calls, conference state, etc.
  - NOTIFY method, Event:-header
- Event subscription
  - SUBSCRIBE, Event:
  - Events may be call-related or third-party generated
  - Security issues: Sensitive Events
    - Privacy
    - Authentication
- Used for personal presence applications
  - Used with PUBLISH method to update presence state
  - Augmented by MESSAGE method for Instant Messaging





# Event Concept

- Piece of state information S
  - Identified by some name (“package”)
- SIP entities interested in S
  - Query for the current state
    - “polling”
  - Be notified about changes to S
    - SUBSCRIBE
  - Subscriptions may be created implicitly
    - By means of other (SIP or non-SIP) protocol activity
- Information about S carried in message body
  - NOTIFY
  - Formats to be defined specific to S
- Protection of S
  - Keep control of who gains access, who has access; for how long



# SUBSCRIBE

- SUBSCRIBE used to registered interest in a piece of state
  - Event: identifies the state information to be retrieved
    - 489 Bad Event
    - Syntax reflects package concept:
      *package* ( '.' *template* )* → e.g. "Event: presence.winfo"
  - Allow-Events: used to indicate which event packages are supported
  - Expires: how long to subscribe
    - Subscriptions are soft-state; need to be refreshed periodically
    - May be shortened or lenthened by server
  - Expires: 0
    - Poll the state information once; do not estabilish a subscription
    - Terminate a subscription
  - Body may indicate desired notification policy (filters)
- Each SUBSCRIBE triggers at least one NOTIFY
  - May contain no (useful) information if access not yet authorized





# SUBSCRIBE Response

- SUBSCRIBE Responses
  - 200 OK: Everything's fine: event understood, client authorized, etc.
  - 202 Accepted : Request received, working on a final result
  - 401, 603, … if applicable
  - Acceptance or rejection to be reported in NOTIFY
- Accepting / rejecting subscription requests
  - Automated e.g. through configured black / white list
  - Ask user if acceptance cannot be determined automatically
- Watcher Info package used to monitor one's own presence
  - Generates notifications for subscription requests
  - User then needs to authorize subscription
    - Through arbitrary means (e.g. web page)

# SUBSCRIBE Response

- SUBSCRIBE Responses
  - 200 OK: Everything's fine: event understood, client authorized, etc.
  - 202 Accepted : Request received, working on a final result
  - 401, 603, … if applicable
  - Acceptance or rejection to be reported in NOTIFY
- Accepting / rejecting subscription requests
  - Automated e.g. through configured black / white list
  - Ask user if acceptance cannot be determined automatically
- Watcher Info package used to monitor one's own presence
  - Generates notifications for subscription requests
- 155 Update Request
  - Typically for authorization instead of 401
- Initiate dialog after receiving 155 or 2xx response
  - Mid-dialog SUBSCRIBE and UPDATE to refresh subscription

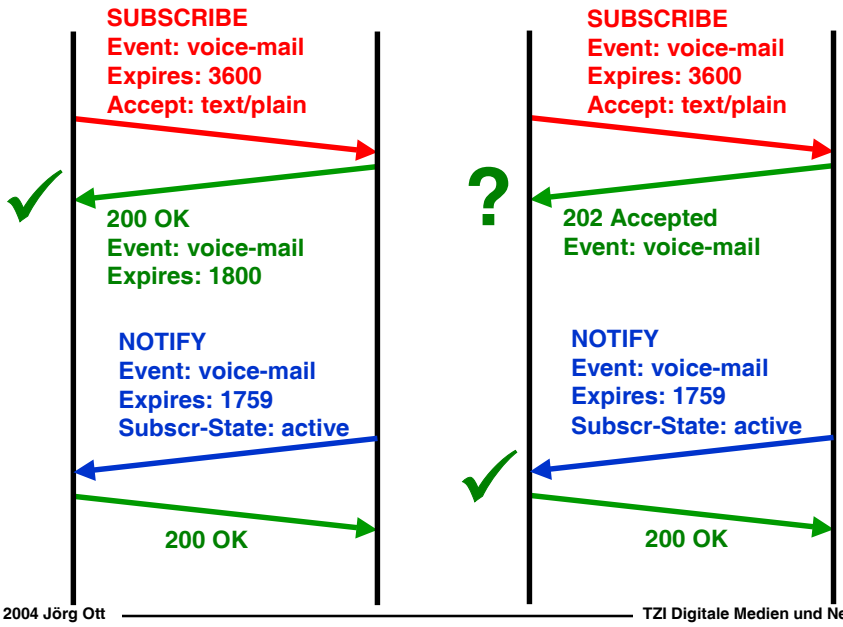




# NOTIFY

- NOTIFY carries state information in message body
  - Format depending on Accept: header from SUBSCRIBE
  - Full state vs. state deltas
  - Subscription-State: active/pending/terminated
- NOTIFY indicates acceptance or rejection of SUBSCRIBE
  - If no immediate response could be supplied
- NOTIFY may be used to terminate subscriptions
  - Initiated by the notifier
  - Includes reason code parameter
- NOTIFY may be sent without SUBSCRIBE
  - Implicit subscription
- Response to NOTIFY
  - 200 OK vs. 481 Subscription does not exist
- Notification rate: risk of network congestion
  - Event throttling to be specified in event packages

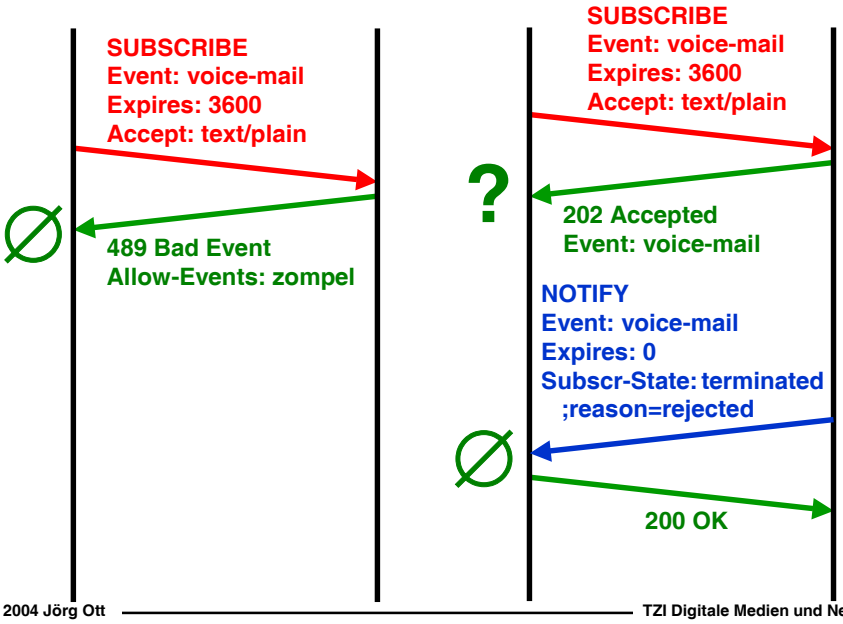
# Example: Successful SUBSCRIBE








# Example: Unsuccessful SUBSCRIBE



# REFER Method

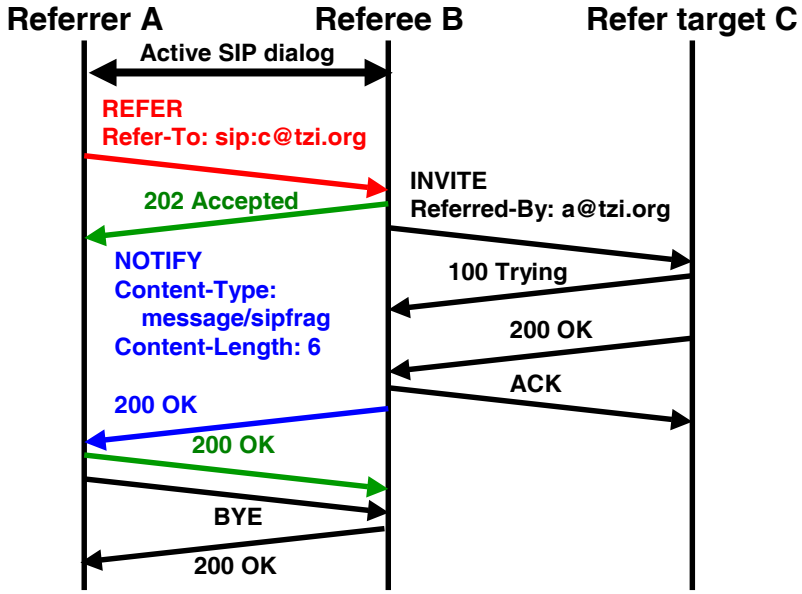
- Original motivation: Call Transfer
- General idea: make a SIP entity contact a third party

- Originator sends REFER method to peer
  - indicates *refer target* in Refer-To: header
- Referred party accepts or declines immediately
  - 202 Accepted or uses some SIP error code
  - Accepting establishes an *implicit subscription* on the new dialog
- Contacts *refer target* in an entirely independent dialog
  - May indicate who caused the call in Referred-By: header
  - New INVITE – 200 OK – ACK sequence, unrelated
- Reports outcome of new call to originator
  - (exactly) one NOTIFY
  - Message body contains SIP message fragments from new call, e.g.
    - 200 OK 486 Busy Here 503 Service unavailable





# REFER Example



# Securing REFER

- Refer Target C receives a request from Referee B
  - May be indistinguishable from any other request from B
  - May be acceptable to C only if A is known as initiator
  - Need to provide authentication information in the request
- Referred-By: header provides first indication
  - But may be faked by B
- Extension to securely pass information from A to C
  - Include signed S/MIME message body in REFER request
    - Content-ID: <sip-message-id>
  - To be copied by Referee into target request
  - Optional cid= parameter of Referred-By: points to signed body
    - Referred-By: <sip:referrer@example.com>;cid="<sip-message-id>"
- Refer target may refuse message without proper Referred-By
  - 429 Provide Referrer Identity





# REFER Semantics?

- Refer-To: supports arbitrary parameters
  - Including method= and response=
  - May populate various headers of a message
  - Enables remote control of a SIP UA
- Default: INVITE mehod for SIP URIs
  - Inside dialog: call transfer
  - Outside dialog: click-to-dial
- But generalized command: Apply <method> to <URI>
  - "Force" a remote UA to do something
  - In principle broad applicability but only little semantics "defined"
    - Fetch the content at http:// …
    - Subscribe to a certain resource
    - Send a certain reply to a request
  - Handle with care…

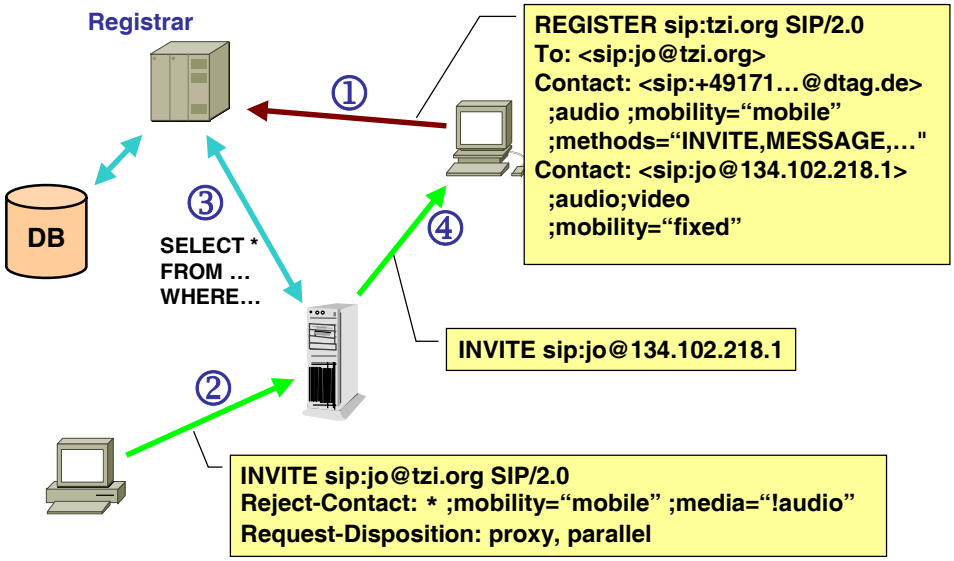
# Caller Preferences / Callee Capabilities

- Caller may provide preferences for request routing
  - URI-based Proxy or redirect
  - Refuse particular URIs
  - Forking
  - Recursive search
  - Parallel or sequential search
- SIP Extensions
  - Request-Disposition: Request routing in SIP servers
  - Accept-Contact: Change address ordering
    - Parameters: require, explicit
  - Reject-Contact: Reject particular URIs
  - Implied predicates from SIP method, events
  - Additional parameters for Contact: headers in REGISTER

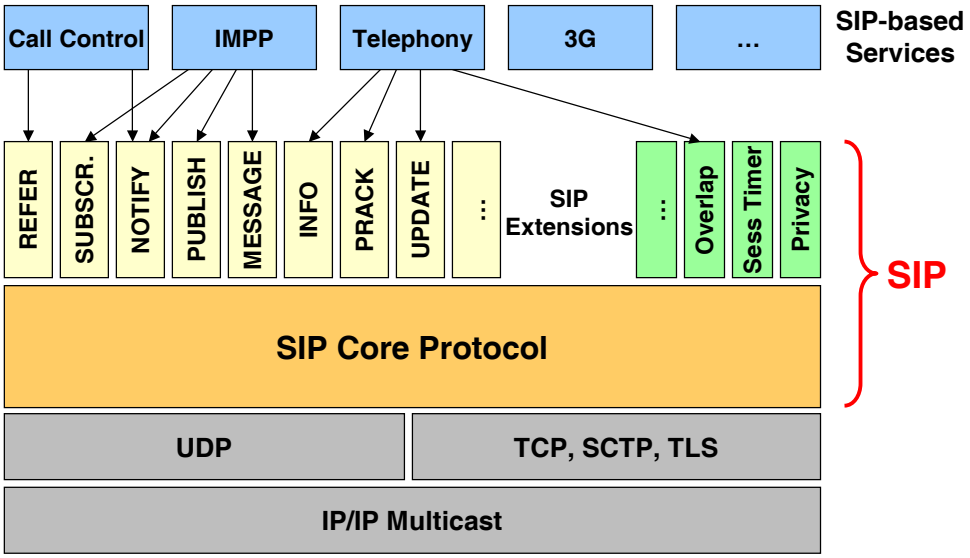




# Preference/Capability Matching



## SIP Service Creation Model








## Call Processing Language (CPL)

- Per-user configuration of server behavior
  - Idea of flexible service configuration (as on a PBX)
  - Specification of call-related actions
  - No side-effects
  - No loops or recursion
- Graphical representation, stored as XML document
- CPL scripts are associated with user's URI
  → registration; upload
- Design goals
  - Light-weight: minimal use of server's resources
  - Safety: do not break running server
  - Extensibility: add features without breaking existing scripts

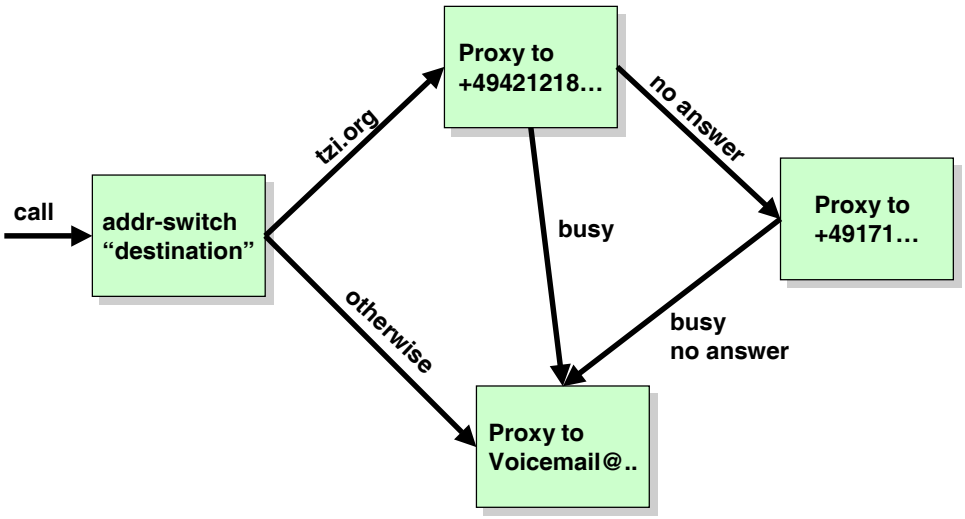
# Language Features

- Server provides information about incoming message
  - Destination
  - Originator
  - Caller preferences
  - Contents of several important header fields
  - Media description
  - Security parameters
  - …
- CPL scripts can specify several actions to take
  - Reject, redirect or proxy incoming message
  - Set timeout values for actions
  - Context-dependent choice of different actions
  - Perform location lookups
  - …





# Example CPL Script



```
<?xml version="1.0" ?>
<cpl>
  <subaction id="vm">
    <location url="sip:voicemail@dmn.tzi.org">
      <proxy />
    </location>
  </subaction>
  <incoming>
    <address-switch field="destination" subfield="host">
      <address subdomain-of="tzi.org">
        <location url="tel:+49421218…">
          <proxy timeout="10">
            <busy> <sub ref="vm" /> </busy>
            <noanswer>
              <location url="tel:+49171…">
              […]
            </noanswer>
          </proxy>
        </location>
      </address>
    </address-switch>
  </incoming>
</cpl>
```



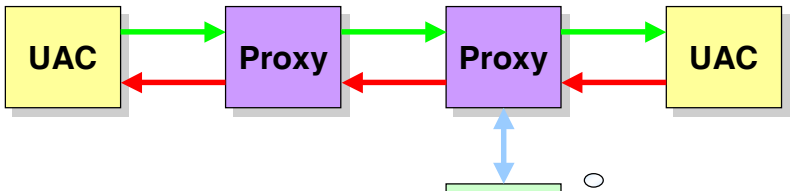


# Service Creation

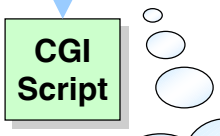
- **Flexible processing of SIP messages in *proxy***
  - Rapid development of supplementary services
  - Pass incoming requests to external script for processing

→ **SIP Common Gateway Interface (CGI)**
  - Adaptation of HTTP CGI
  - Support SIP's idiosyncrasies:
    - Transport protocols UDP, TCP
    - Central role of proxy servers
    - Persistent transaction state
    - Registrations
    - Request forking, recursive search, timeouts
    - …

## SIP CGI Architecture




- Fork script on incoming messages
  - Pass message on STDIN
  - Provide additional information in environment vars
- Process commands from script's STDOUT
  - Message forwarding
  - Create new messages
  - Add/delete message headers
- Use cookies to preserve state information




Script is invoked for subsequent messages of same transaction only if explicitly requested.





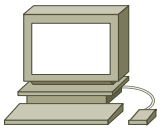
# SIP for Presence and Instant Messaging

(a somewhat brief introduction)

# A Role for Presence in "VoIP"

**Awareness of other users: availability, location, …**

- **To perform existing functions**
  - User location, call routing, follow-me, ...



- **To improve existing services**
  - Call completion ratio
  - Indicate availability

- **To enable new services**
  - Presence per se: Simplify meeting people
  - Messaging per se: "SMS"
  - Presence and Messaging as basis for other applications
  - Location-based services

- **To rescue the "VoIP" industry…**

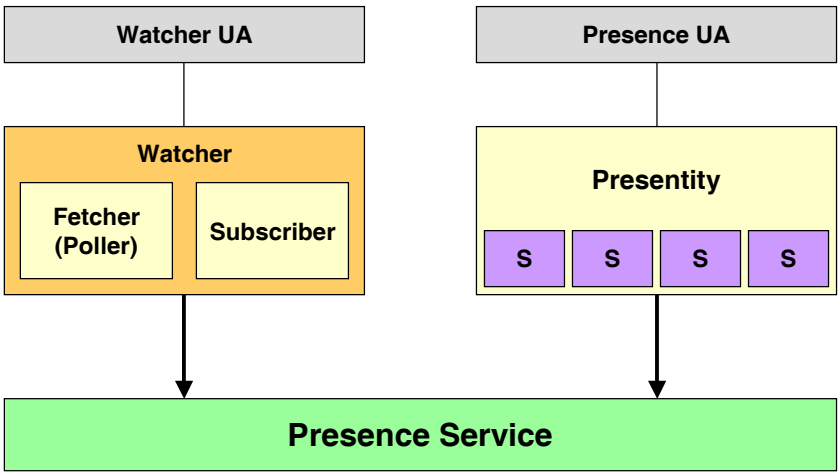




# SIP: Personal Presence and Instant Messaging

- **"Buddy Lists + Chat"**
- **Idea: re-use SIP infrastructure**
  - Maintain user locations
  - Route messages
  - Contact users
- **SIP Event package for "presence"**
  - SUBSCRIBE / NOTIFY fit well
  - Define presence format
- **Define a new method for Instant Messaging**
  - MESSAGE
  - Define basic message content format (text/plain)

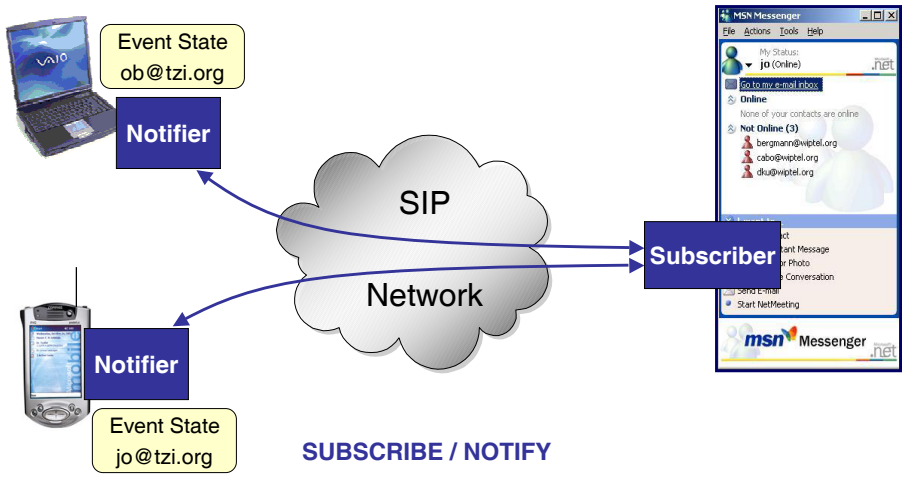
# IMPP Presence Model



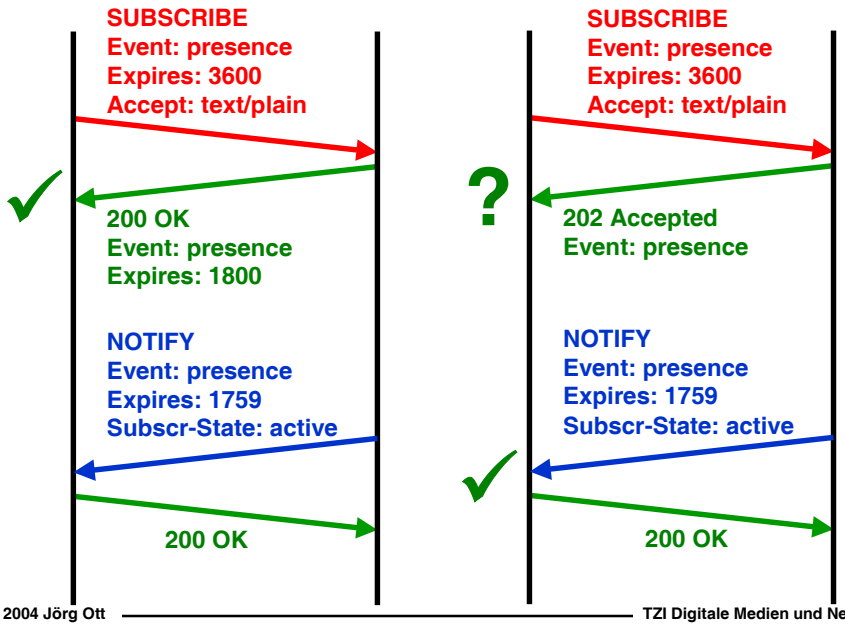





# SIP Entities for Presence (1)



# Example: SIP for Presence








# SIP Presence Notification (PIDF)

```
<presence ... entity="pres:jo@tzi.org">
  <tuple id="mobile-im">
    <status>
      <basic>open</basic>
    </status>
    <contact priority="0.8">im:jo@sms-gw.tzi.org</contact>
    <note xml:lang="en">Don't Disturb Please!</note>
    <note xml:lang="fr">Ne dérangez pas, s'il vous plaît</note>
    <timestamp>2003-01-14T10:49:29Z</timestamp>
  </tuple>
  <tuple id="interactive-mm">
    <status>
      <basic>closed</basic>
    </status>
    <contact priority="1.0">sip:jo@tzi.org</contact>
  </tuple>
  <note>I'll be in Paris next week</note>
</presence>
```

# Rich Presence Information Data Format (RPID)

- Introduces additional (more descriptive) elements
  - Ideally to be derived automatically from user activity
  - Organized in tuples
  - May be grouped into classes

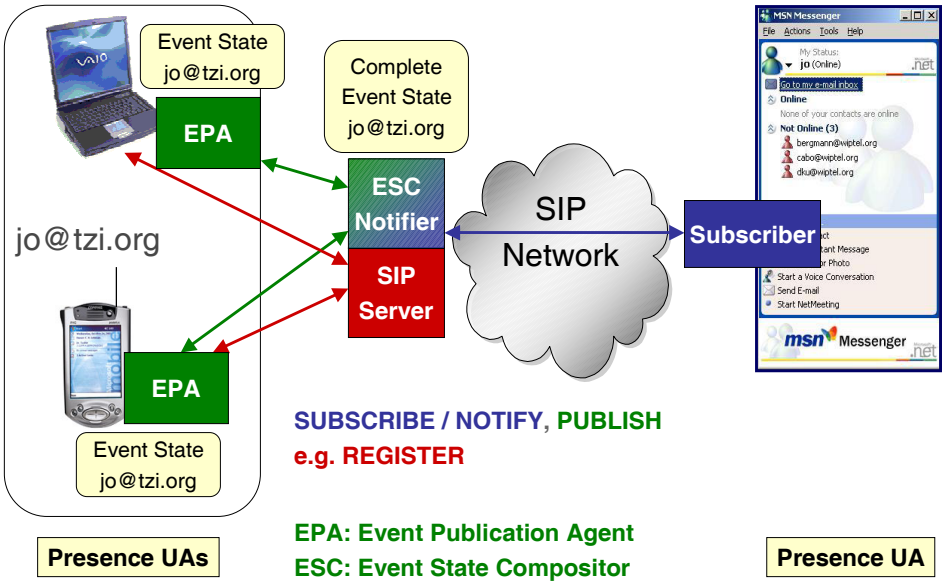
**Elements and value examples:**

- Activity: on-the-phone, away, meeting, travel, vacation, ...
- Contact-Type: device, in-person, services, ...
- Idle: inactivity time of a user (e.g. not typing)
- Place: home, office, public, street, train, ...
- Privacy: public, private, quiet





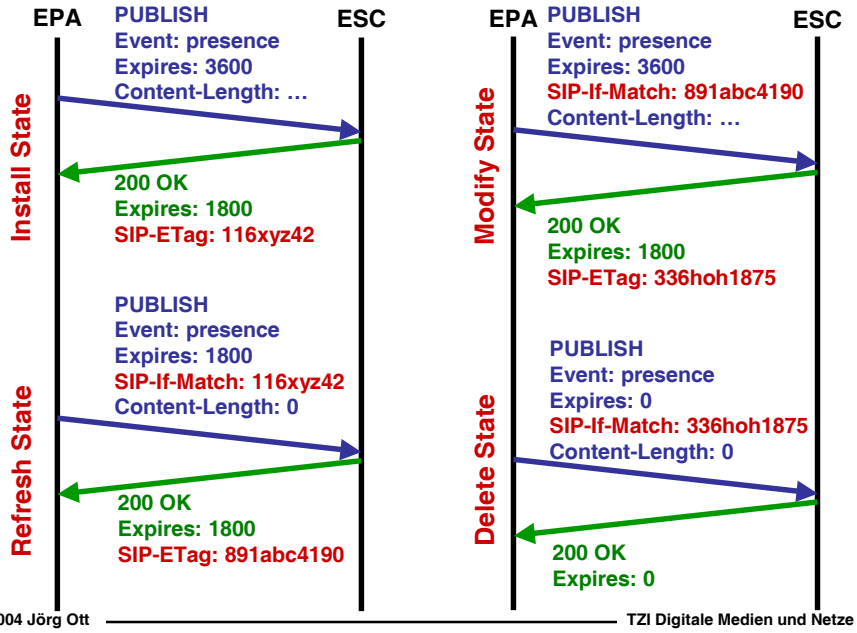
# SIP Entities for Presence (2)




SUBSCRIBE / NOTIFY, PUBLISH
e.g. REGISTER

EPA: Event Publication Agent
ESC: Event State Compositor

# Example: PUBLISHing State



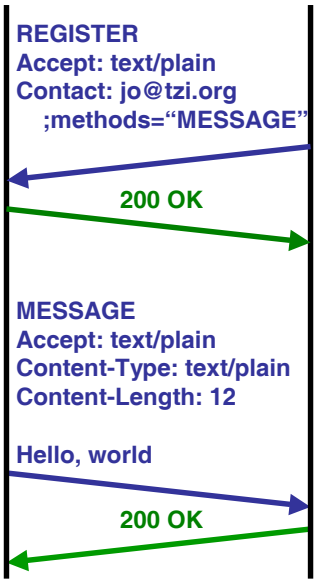





# SIP for Instant Messaging (IM)

- UAs may send and receive messages
  - Similar model to Presence
- Receivers indicate support when registering

  Contact: sip:jo@tzi.org;methods="MESSAGE"
- Senders just send
  - Use sip: or im: URLs
  - Congestion control is important!
- For longer "chat sessions"
  - Create a persistent IM session
    - Just another media type
  - May not be able use TCP or TLS
  - Use SIP-based session instead?
    - Lightweight SIP just as transport protocol



## Presence & IM Security

- Authentication of (Subscription) Requests
  - Standard SIP mechanisms: 401/407 responses
- Authorization: Users must stay in control of subscriptions
  - May subscribe to their own subscription state ("watcher info")
  - Receive notifications for every subscription attempt
  - May authorize each subscription
  - May cancel existing subscriptions
  - May retrieve lists of subscribers
- Authentication of Instant Messages
  - May include contents to be automatically acted upon
  - User / system needs to validate originator
- Meta-issue: end-to-end authentication





## Issue: Congestion Control

- PUBLISH, NOTIFY: Throttling of events
  - Keep event rate under control: one PUBLISH / NOTIFY per RTT
  - To be defined on a per-package basis
- MESSAGE: Message frequency
  - Only one outstanding message: one MESSAGE per RTT
    - But: messages are stand-alone; no dialog context to check against
- MESSAGE: Large messages
  - UDP is an acceptable transport for SIP: no congestion control
  - Endpoints can't see beyond next hop
  - Artificial limit on message size?
  - Request end-to-end congestion control?
  - Content indirection: convey only pointers (URLs) to contents in message?
- Approach: Message Session Relay Protocol (MSRP)

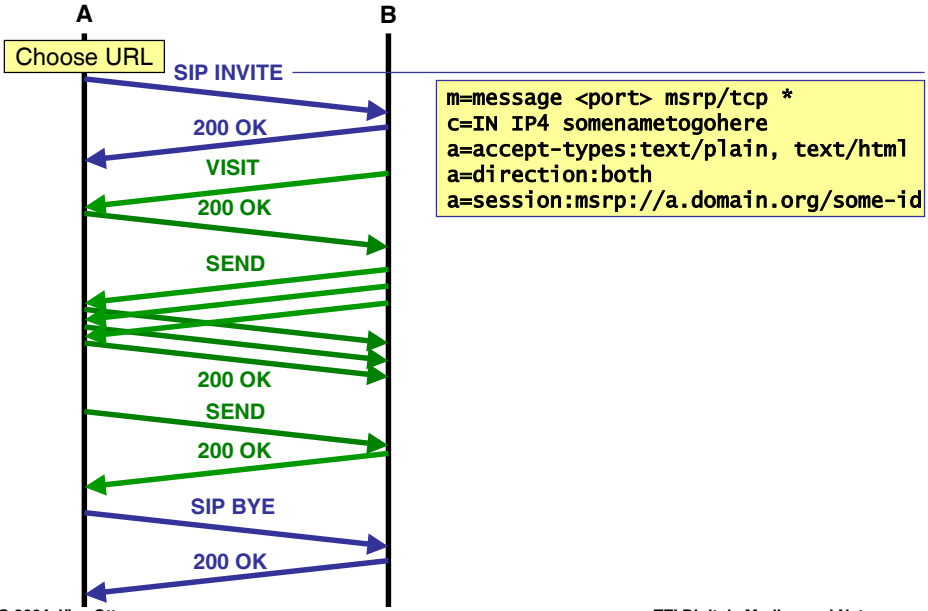
# Message Session Relay Protocol (MSRP)

- **Protocol for Messaging Sessions**
  - Uses TCP or another reliable and congestion controlled transport
  - Message encoding similar to SIP and HTTP
- **Just another media protocol**
  - Messaging sessions require explicit setup and teardown
  - MSRP URLs to identify sessions
  - SDP to describe sessions (m=message)
  - Uses SDP Offer/Answer to convey parameters
    - e.g. in SIP INVITE / 200 OK / ACK
  - VISIT method to establish session
  - SEND method to convey messages
  - Session termination out of band (e.g. SIP BYE)
- **Two modes of operation**
  - Direct communication between peers (simple case)
  - Communication via relays (NATs, firewalls, policy)
    - BIND method to work with relays

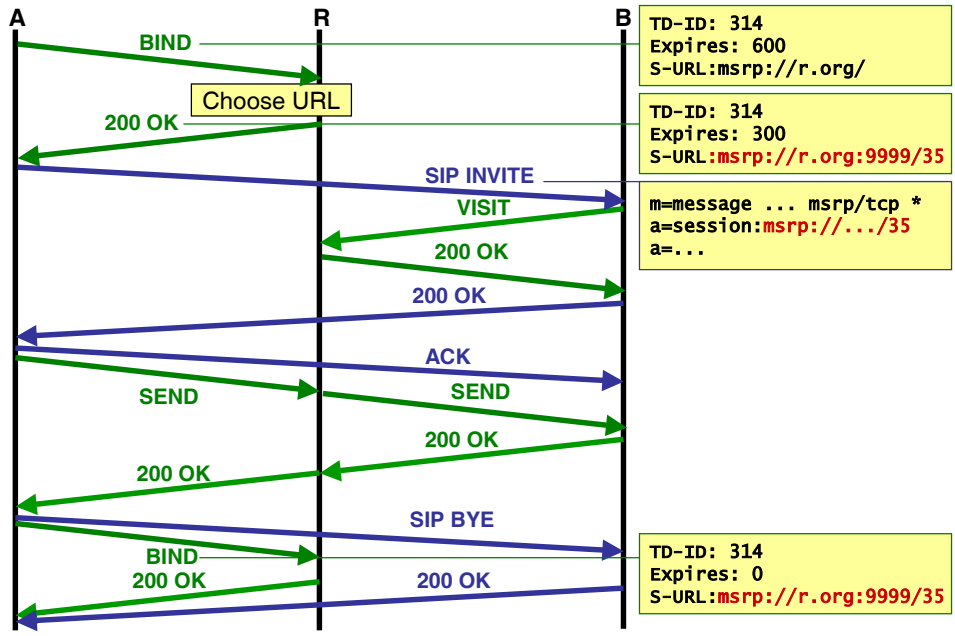




# Direction Communication between Peers



## Communication via a Relay








## Tutorial Overview

- Internet Multimedia Conferencing Architecture
- Packet A/V Basics + Real-time Transport
- SIP Introduction, History, Architecture
- SIP Basic Functionality, Call Flows
- SIP Security
- SIP Service Creation
- SIP in Telephony ← You are here
- SIP in 3GPP

# SIP for Telephony

…yet another set of SIP services…



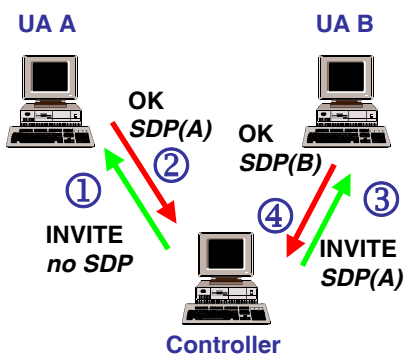


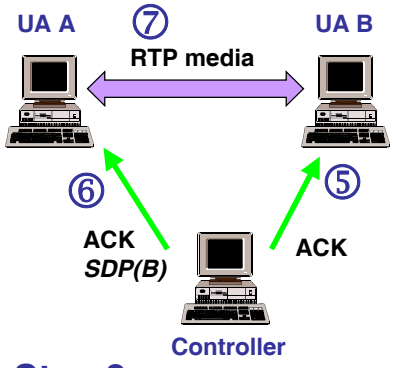
## "Supplementary Services"

- Call Diversion
  - Intrinsic support (301/302 redirection)
- Call Park & Pickup
  - Distributed state of user agents; embed call state in cookies
- Call Hold and Retrieve
  - User agents sends re-INVITE to mute other parties (a=inactive)
- Call Waiting
  - Implemented in endpoints
- 3rd party call control
- Call transfer
  - Use new REFER method to indicate a party to place a call to
- Message waiting
  - Specific event package: message-summary
- Conferencing
  - Various extensions; supports base and enhanced SIP UAs

# Third-Party Call Control (3PCC)

- Examples: “Click-to-dial”, conference bridge control, …
- Several approaches with different advantages / drawbacks
- Simplest call flow:



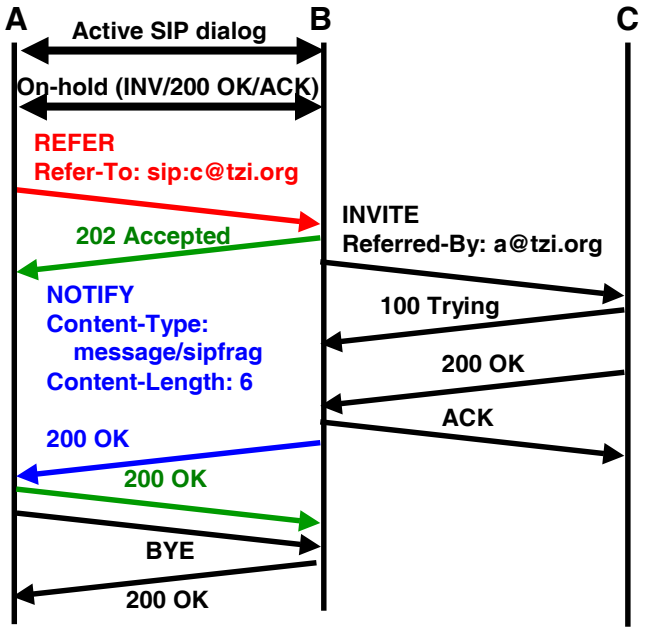





# Call Transfer

- Part of Call Control Framework
- Uses basic SIP protocol features and extensions
  - REFER method to invoke another (INVITE) transaction
  - NOTIFY (with implicit subscription) to indicate success or failure
  - New Replaces: header to indicate substitution of an existing call
- Supports numerous variants
  - Attended
  - Unattended
  - Intermediate three-way calling
  - Optional protection of transfer target

## Simple Unattended Transfer








## Message Waiting Indication

- Asynchronously notify endpoint(s) about messages
  - Voice, video, image, text, ...
- Define a new SIP event package
  - Subscribe to one or more mailboxes
  - Results from many sources may be merged
- Content-Type: application/simple-message-summary
  - General indicator for new messages
  - Message type followed by new/old and (new-urgent/old-urgent)
  - Encoding as plain text
- Example:

```
Messages-Waiting: yes
Voicemail: 4/8 (1/2)
Email: 238/42116 (0/1)
```

## SIP Conferencing

- Motivators: n-way calling, video conferencing, …
  - Tightly coupled conferences
- Different conferencing models preserved
  - Except for “fully meshed” conference: complexity just not worth it!
- Terminology
  - Trying to avoid already overloaded terms as much as possible
- Functional entities in a “system model”
  - Plus implementations examples
- Set of protocols
  - Definition of basic building blocks (SIP and other)
  - Sample combinations to implement conferencing services





## Conference Participants

- MUST work with basic SIP support only
  - No awareness of conference
  - Just a point-to-point call with minimal means for control
  - Possibly augmented by out-of-band control (e.g. HTTP)
- Member types (SIP UA)
  - Conference-unaware: plain-old SIP device
  - Conference-aware: supports conferencing features
  - Focus: (one) center of a conference
  - Anonymous: Visible but unidentified participant
  - Invisible: Participant whose presence is not known

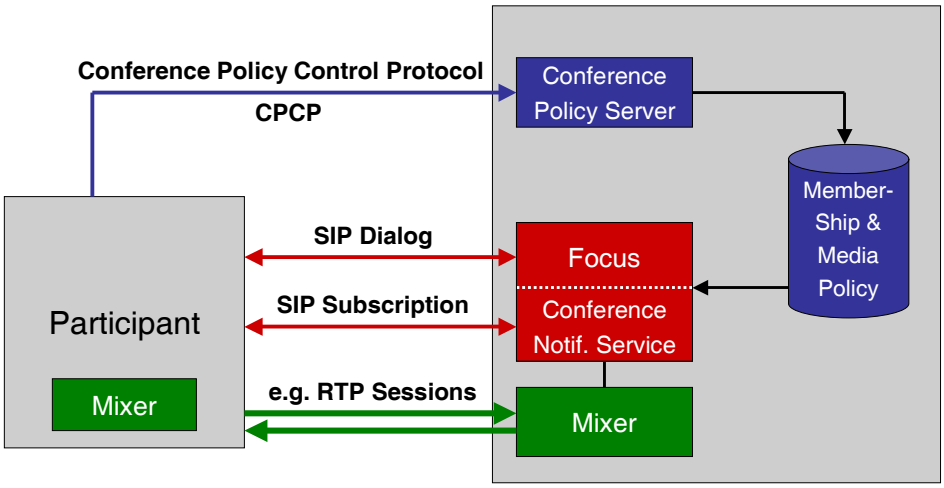
# Conferences

- **Conference types**
  - Basic: just plain SIP, no further means for control
  - Complex: some conferencing features provided
  - Cascaded: several foci concatenated in a conference
  - Sidebar: conference as (logical) part of another
- **Focus** — Signaling center of a conference
- **Conference URI** — Identifies a focus (isFocus parameter may indicate this)
- **Factory URI** — for automated conference creation
  - Yields a dynamically generated conference URI in return

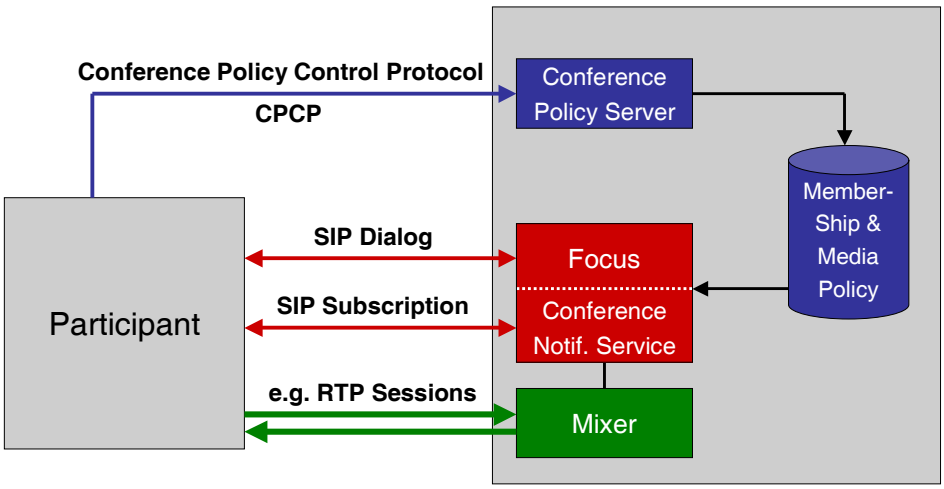




# Terminology and Model



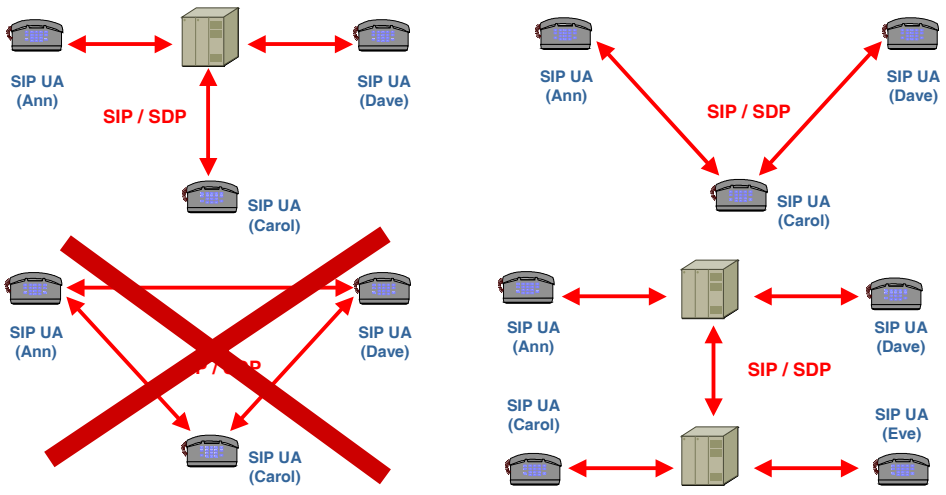
# Terminology and Model



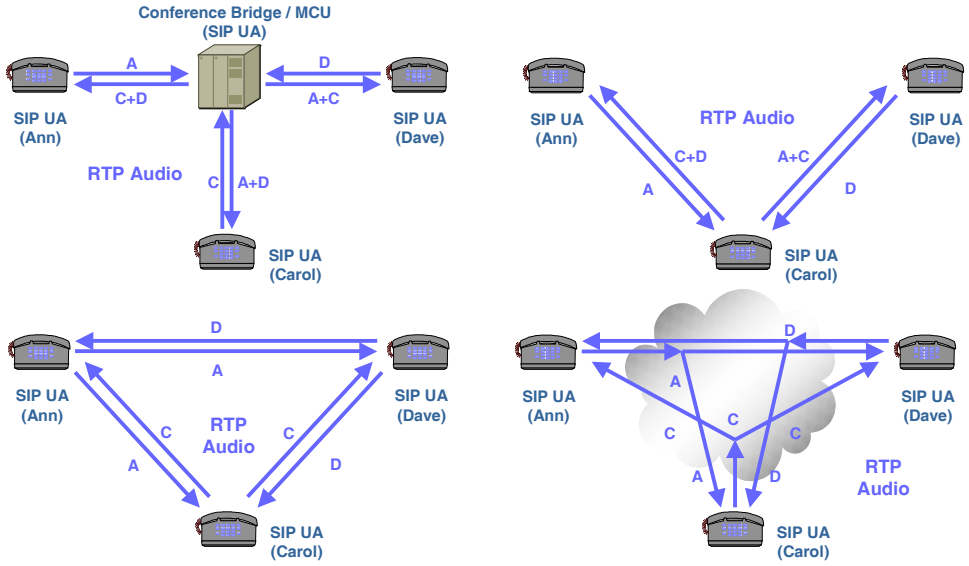





# SIP Signaling Relationships



# (RTP) Media Sessions








# Conferencing Scenarios (1)

- Simple conferencing scenarios
  - Plain SIP only (RFC 3261, 3264)
- Extend point-to-point call
  - Works only with local focus; otherwise, new call required
- Ad-hoc conference
  - Automated creation at focus
  - IVR / DTMF for control
  - Audio for information about the conference and its members
- Reserved conference
  - Same as ad-hoc
  - Use external means for reservation and configuration (e.g. web)

## Conferencing Scenarios (2)

- Advanced conferencing scenarios:
  - Support for Call Transfer
  - Support means to communicate information from focus to UA
  - Optional: means to manipulate conference and media policy
- Extend point-to-point call
- Join / create a conference based upon an existing dialog
- Ad-hoc conference
- Reserved conference
- Make use of additional conferencing features





## Sample Conferencing Features

- Invite participants (dial-in, dial-out), expel participants
- Authenticate new participants by members
- Obtain conference and media policy information
- Manipulate membership policy
  - Participant privileges, participant management (black list, white list)
  - Floor control
- Explicit media control (media policy)
  - Configure media distribution
  - Add / remove media sessions
- Create, control, and terminate sidebars
  - Separate conference vs. media policy
- …

# Membership Policy

**Define, retrieve/notify, modify, and act upon…**

- **Formal rules for the conference**
  - Conference creation, termination, (policy) modification
  - Access control: black list, white list, rules for authentication
  - Privileges of individual participants
  - Visibility of the conference and its members
  - Access to floor and media policy (defined separately)
- **General conference attributes**
- **Participant management**
  - Invite, expel
- **…**





# Media Policy

- **Mixer model**
  - Input switch: collecting & selecting input streams from partcipants
    - Possibly transcoding and other per-media functions
  - Mixing topologies describing mixing policies
  - Output switch: selecting & distributing output streams to participants
    - Possibly transcoding and other per-media functions
- **Mixer may be centralized or not**
- **Media policy defines how incoming streams are processed, combined, and then distributed**
  - Individual mixing functions may be defined per participant
  - Common mixing functions may be defined for the conference
  - Mixing function may take into account “events” from other components

# SIP Signaling Building Blocks

- **Membership control**
  - Initiation of conferences: INVITE
  - Inviting / adding to conferences: INVITE, REFER
  - Leaving a conference: BYE
  - Expelling from a conference: REFER (method="BYE")
- **Conference control**
  - State change notifications: SUBSCRIBE / NOTIFY
    - Dialog package, conference package, ...
  - Conference / media policy control
    - Might use data manipulation framework discussed in SIMPLE context
- **Other**
  - Determine focus URI OPTIONS
  - ...
- **Augmented by other protocols (CPCP for conference control)**

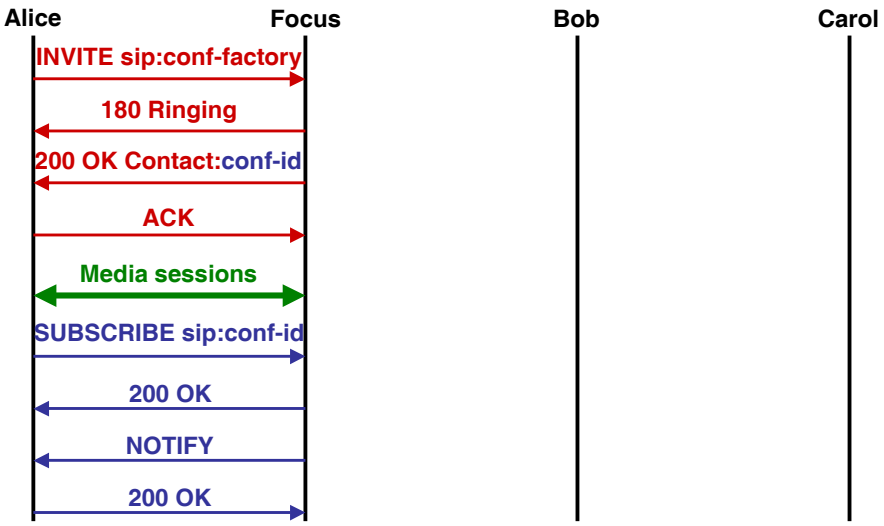




# Signaling Building Blocks using other Protocols

- **Conference Policy Control Protocol (CPCP)**
- **Floor control protocol**
- **Common baseline:**
  - Request-response protocol (RPC-style)
  - May be strictly client – server
  - Asynchronous notifications provided by SIP
  - HTTP/HTML access to a web page for human interaction
  - SOAP RPCs e.g. over HTTP

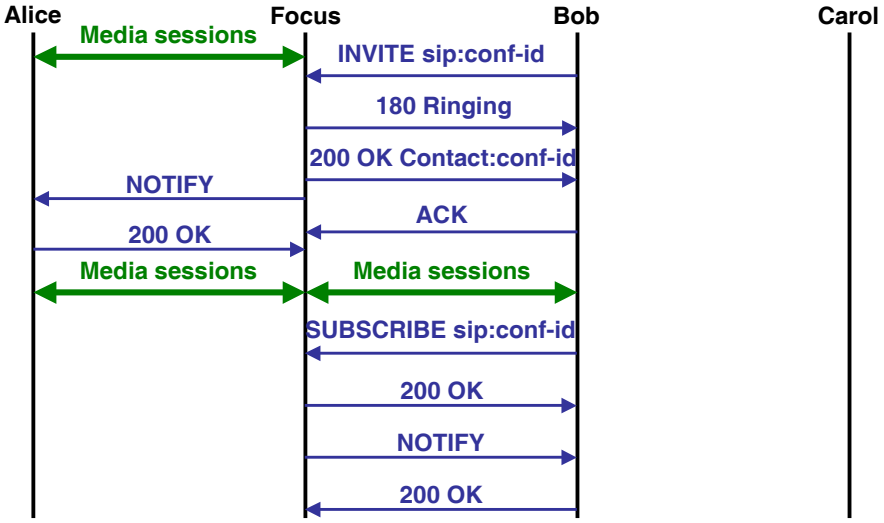
# Call Flow Example: Conference Creation



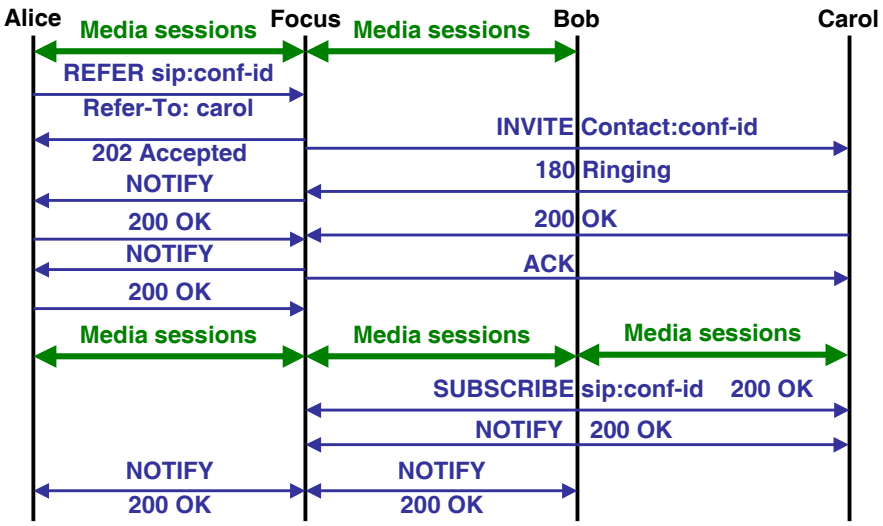





# Call Flow Example: User Joining



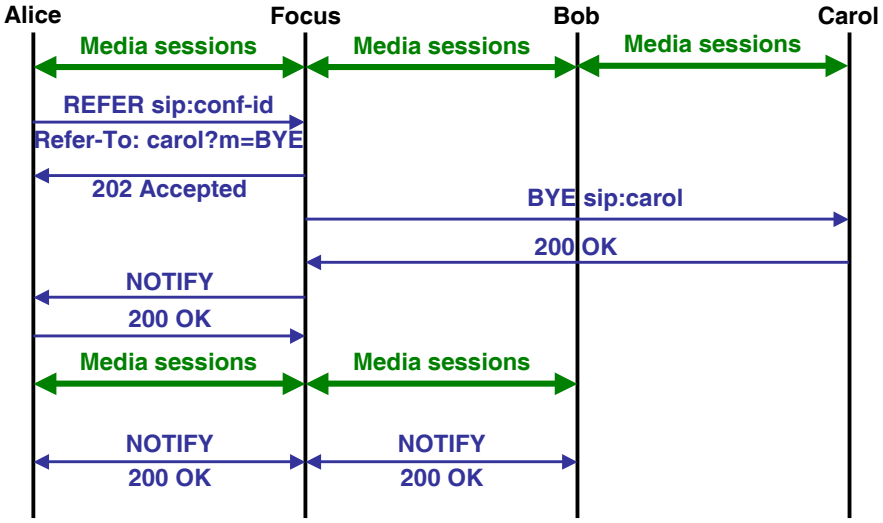
# Call Flow Example: Adding a User



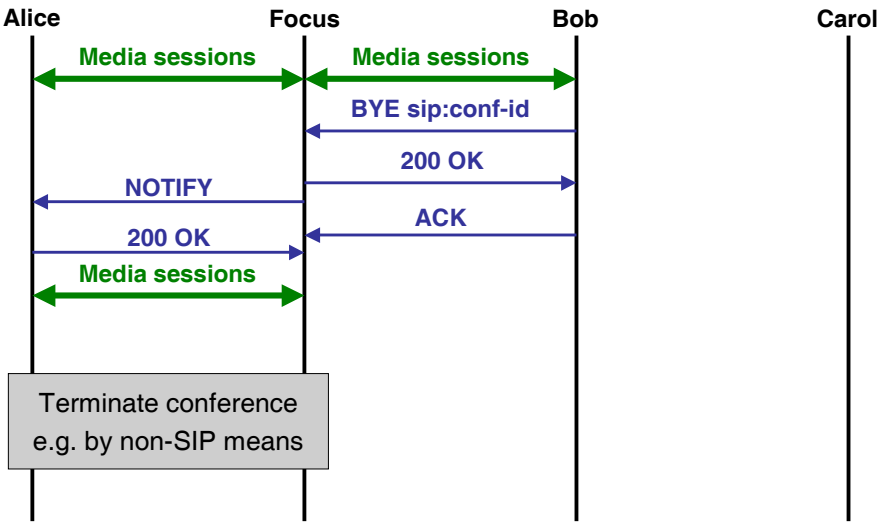





# Call Flow Example: Removing a User



# Call Flow Example: User Leaving

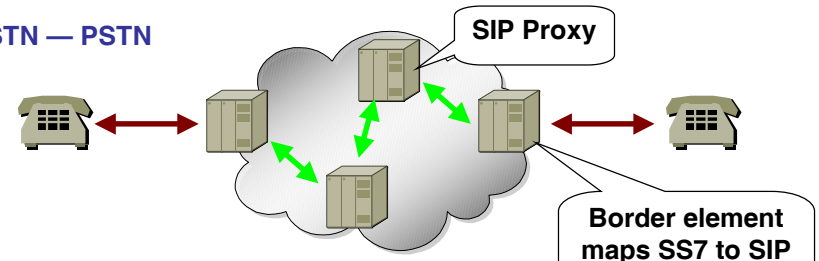




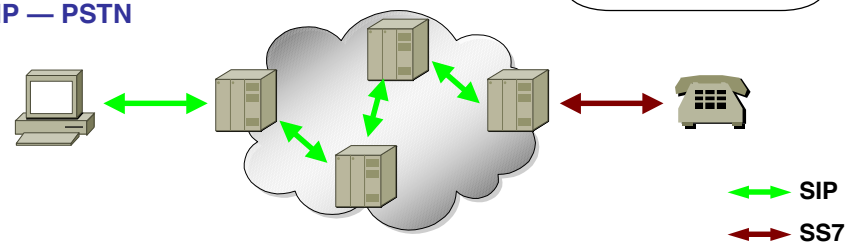



# Interfacing to the PSTN

1. PSTN — PSTN




2. SIP — PSTN



# Interfacing to the PSTN

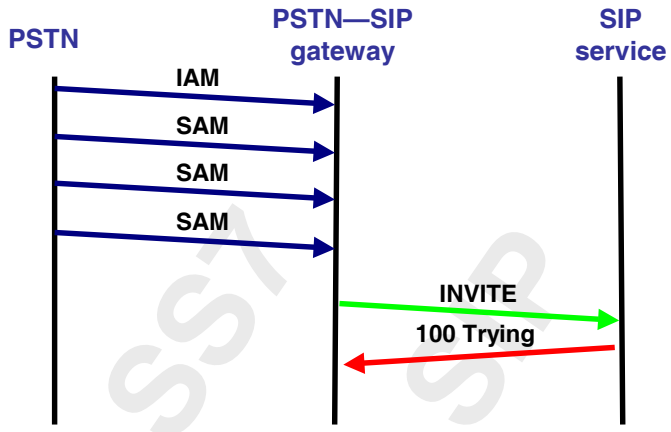
- Interface to the PSTN
- Preserve feature transparency
  → transport SS7 information (ISUP MIME type)
  - Eventually convert between different ISUP versions
- Provide enough routing information to find callee
  → (partially) translate ISUP to SIP
- Support for tel:-URLs to indicate Called Party Number
  - Address resolution using ENUM or TRIP
- Convey additional information during call
  - PSTN – SIP – PSTN case
  → INFO method (RFC 2976)
- SIP to PSTN mapping
  - Call flows for basic interoperation (RFC 3666)
  - Support for *Overlap Sending*





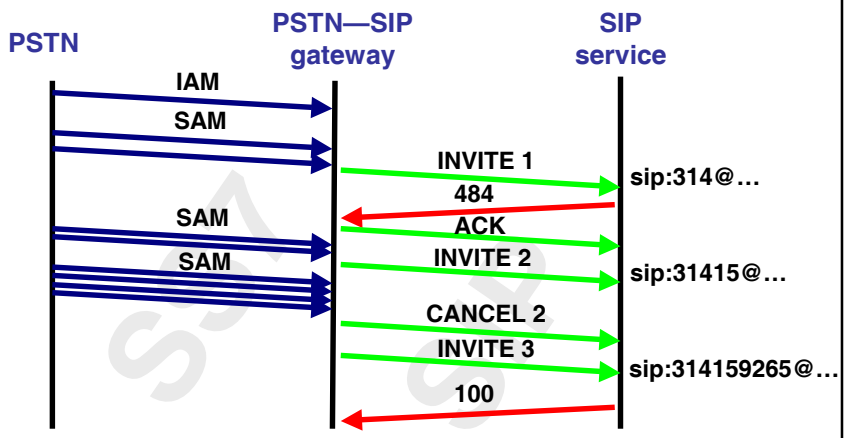
# Example for SIP and Overlap Sending (1)

- Approach 1: Collect digits in gateway
  - Use timeout
  - collect minimal number of digits



## Example for SIP and Overlap Sending (2)

- Approach 2: Send multiple INVITEs
  - Use timeout, possibly collect minimal number of digits
  - Wait for feedback from the SIP network
  - CANCEL obsolete INVITEs

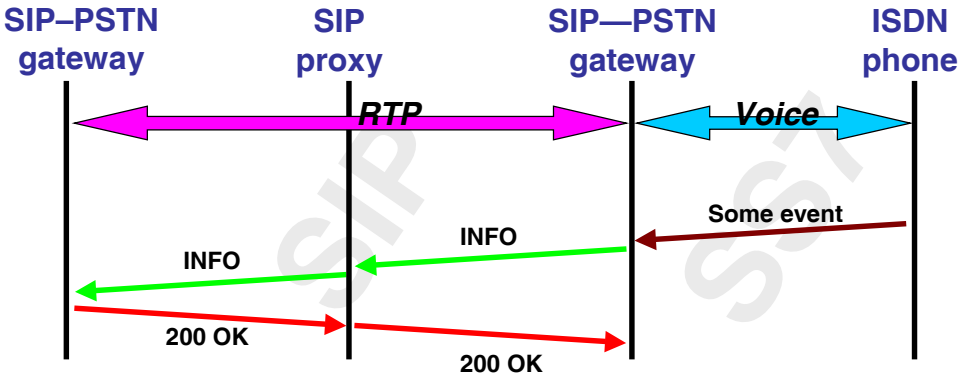







## INFO Method

- Transmit application-layer information during call
  - Use SIP signaling path of current session
  - Information is carried in message headers or body
  - No change of (SIP-related) call state



## 183 Session Progress Message

- INFO not applicable *before* call is established
- ISUP mapping requires inband data prior to final response

→ provisional response 183

- Additional information in message body
- Session:-header
  - Indicate reason: media, qos, security
- Optionally made reliable (100rel)
  - PRACK message to confirm receipt of 183

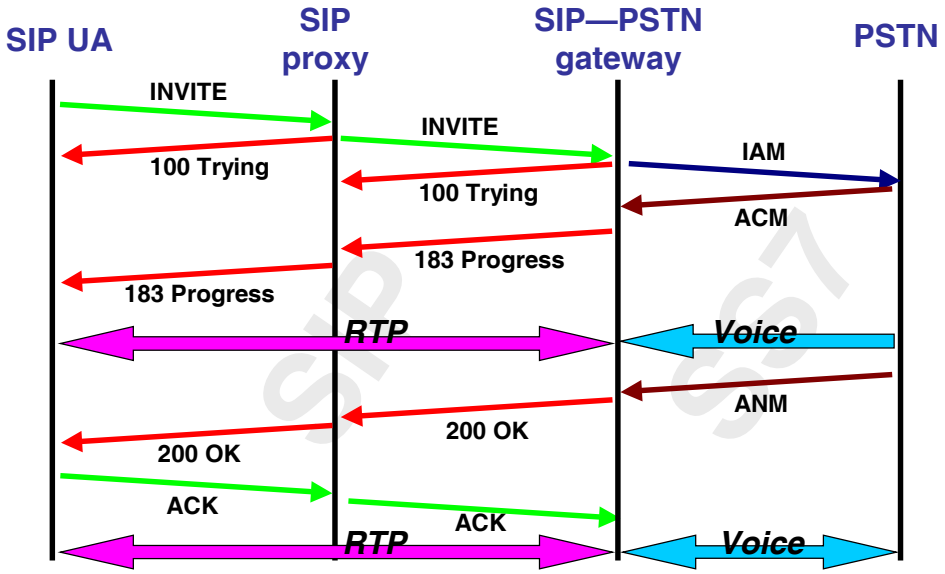




## Early Media Support

- Early media during call setup (SIP INVITE)
  - One-way transmission to report progress
  - Announcements, specific dial tones, …
  - No charge
  - Inhibit local alerting at calling user agent
- Problem: Media negotiation
  - Send SDP message in provisional response
    → *fast setup*
  - Create SDP from initial INVITE's capability set
    → What if not suitable for early media session?
  - Calling UA will establish *early* media session
    → Cannot decline session or change codecs

## SIP—PSTN Call With In-band Alerting








## Quality of Service

- Best-effort quality potentially too poor for IP telephony
  → need support for resource reservation in SIP
- Some applications need multi-phase call-setup
  - Resource reservations to assure certain QoS
  - Establish particular security relationships
    - Extend security services to media channels
    - Avoid fraud and theft-of-service

→ Establish call only if preconditions are met
- No charge for defect calls
- No inconsistent state of involved endpoints
  (alert user only if resource reservation succeeded, …)
- Fallback options: e.g. use codecs with lower bandwith

# SIP Preconditions (RFC 3312)

- SDP extension handles negotiation of QoS parameters
  - SDP attributes describe preconditions to be met
  - Current status (curr:) vs. desired status (des:)
  - Either party may request confirmation on current state (conf:)

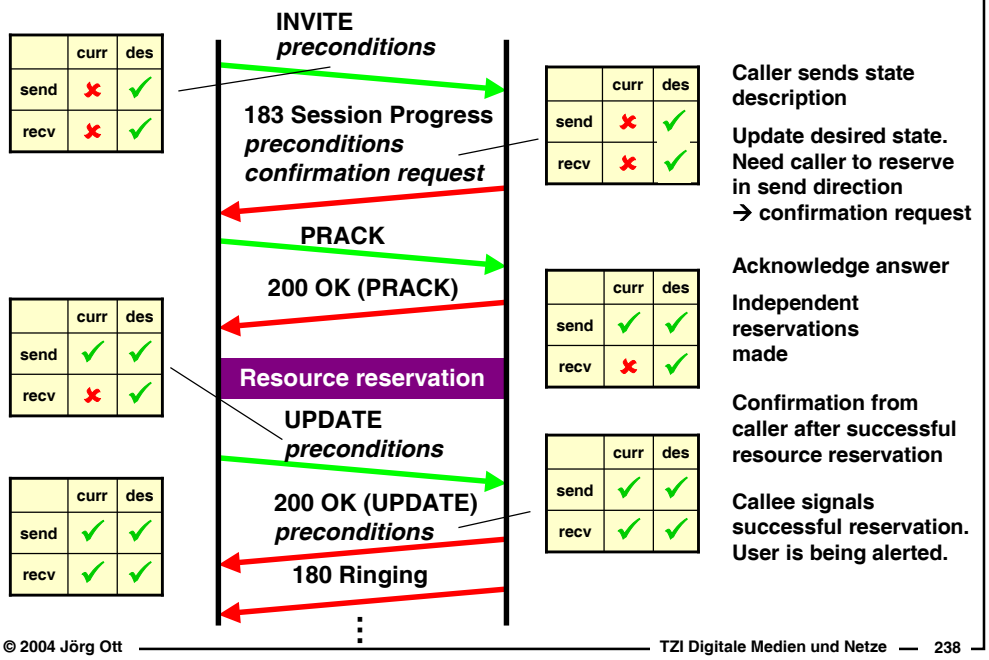
```
'a=curr:' type status direction
'a=des:'  type status strength direction
'a=conf:' type status direction
```

- Offer/answer to create common view on both UAs state
  - Typically INVITE, UPDATE, PRACK
  - Option tag precondition
  - New response code 580 Precondition Failure
  - Mark failed and unknown preconditions in SDP answer





# QoS Parameter Negotiation (Overview)



# Example Negotiation of QoS Parameters

## Confirmation request

```
v=0
o=jo 7849 2873246 IN IP4 ruin.inf…
s=SIP call
t=0 0
c=IN IP4 134.102.218.1
m=audio 52392 RTP/AVP 98 99
a=rtpmap:98 L8/8000
a=rtpmap:99 L16/8000
a=curr:qos e2e none
a=des:qos mandatory e2e sendrecv
m=video 59485 RTP/AVP 31
a=rtpmap:31 H261/90000
a=curr:qos local send
a=curr:qos remote none
a=des:qos optional local sendrecv
a=des:qos optional remote sendrecv
```

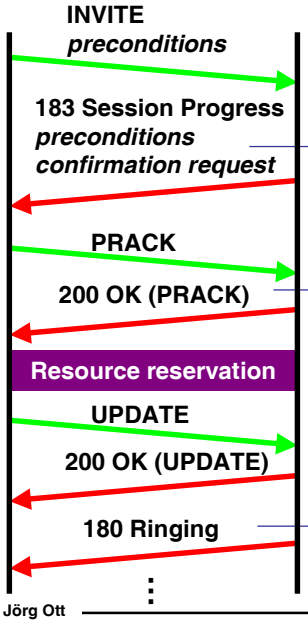
## Confirmation response

```
v=0
o=cabo 2552 892834 IN IP4 dmn.inf…
s=SIP call
t=0 0
c=IN IP4 134.102.218.46
m=audio 50239 RTP/AVP 98 99
a=rtpmap:98 L8/8000
a=rtpmap:99 L16/8000
a=curr:qos e2e none
a=des:qos mandatory e2e sendrecv
a=conf:qos e2e recv
m=video 56112 RTP/AVP 31
a=rtpmap:31 H261/90000
a=curr:qos local none
a=curr:qos remote send
a=des:qos failure local sendrecv
a=des:qos optional remote sendrecv
```



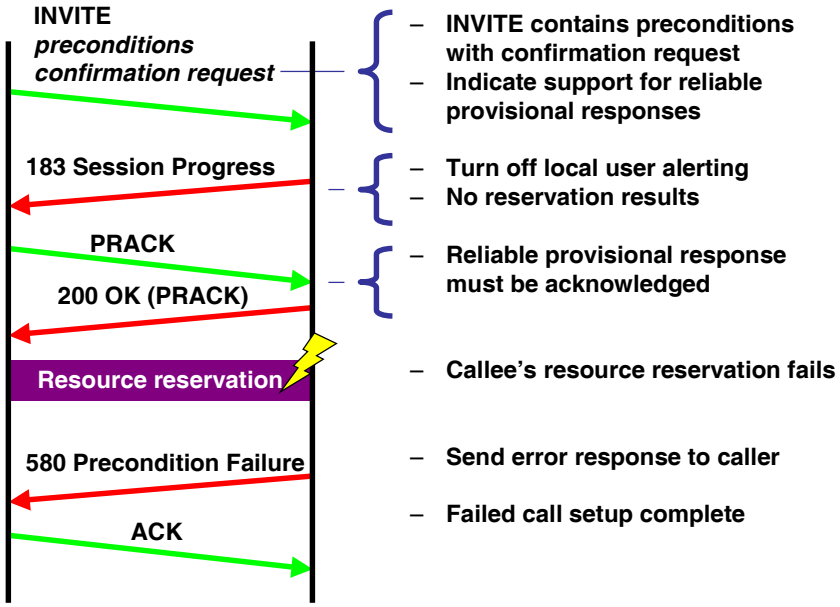


# Successful Reservation




- INVITE indicates support for reliable provisional responses
- Turn off local user alerting
- Choose appropriate subset of preconditions offered by sender
- Include confirmation request
- Reliable provisional response must be acknowledged
- Initiator of confirmation request awaits UPDATE message or CANCEL
- Continue call setup

# Failed Reservation








# Interaction with Application Services

- Traditional PSTN / PBX services often controlled via DTMF
  - Stimulus signaling with semantics defined by the application
  - Assumes that signaling and media paths are identical
- Does not apply to SIP
  - Media and signaling may go separate paths
  - Application servers may be on neither path
- SIP defines Application Interaction Framework
  - Taxonomy for interactions between users and application servers
  - Focus on stimulus-based signaling
- KPML Approach → Thursday
  - SUBSCRIBE to register interest of an application server in user input
  - XML document to describe input expected from user
  - NOTIFY to inform server about key events

# Operational Issues

- Regulatory Stuff ➔ Friday
- Deployments ➔ Wednesday, Thursday
- Accounting / Billing ➔ Friday
- NATs / Firewalls ➔ Thursday
- Devices ➔ …





# Regulatory Issues

- Public telephony networks must fulfill certain functions
  - Legal interception
  - Prohibit abuse or malicious use
  - Availability for disaster recovery
  - → Need feature transparency in hybrid networks
- Multilevel Priorization (MLP)
  - Increase probability for call-setup on overloaded networks
    ISUP IAM precedence parameters and calling party category
    → Resource-Priority: header in SIP
  - Avoid abuse
    → authentication and authorization
- Preemption
  - Make resources available for high-priority traffic
    (release existing resource reservations if necessary)
  - Legal issues in many countries

# Regulatory Issues

- Public telephony networks must fulfill certain functions
  - Legal interception
  - Prohibit abuse or malicious use
  - Availability for disaster recovery
  - → Need feature transparency in hybrid networks

e.g.: E.106, GETS, GTPS

- Multilevel Priorization (MLP)
  - Increase probability for call-setup on overloaded networks
    ISUP IAM precedence parameters and calling party category
    → Resource-Priority: header in SIP
  - Avoid abuse
    → authentication and authorization

- Preemption
  - Make resources available for high-priority traffic
    (release existing resource reservations if necessary)
  - Legal issues in many countries

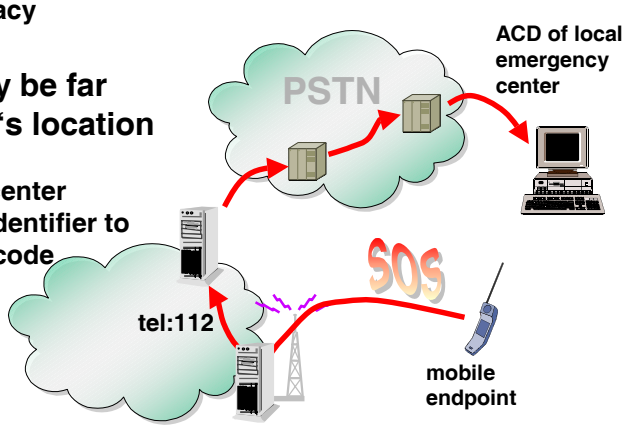




# SOS Calls

- Emergency calls for individuals
  - No calling party categorization
  - Geographic location of caller
  - Prohibit abuse: authentication, no CLIR possible
  - Need to respect privacy

- Access network may be far away from endpoint's location
  - Find geographically nearest emergency center
  - Map symbolic SOS-identifier to suitable emergency code
  - Add geographic info



# Firewalls, Proxies, NATs

- Intermediate devices in the Internet (aka *Middle Boxes*)
  - Can be transparent or non-transparent to hosts
- Different functions:
  - Packet-filters (firewalls)
  - Application layer gateways (proxies)
  - Network address translators (NATs)

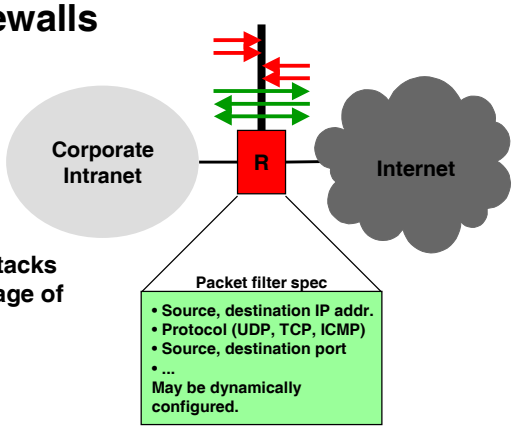
Middlebox communication





# Firewalls

- Packet filters, enforcing packet filtering/forwarding policies
- Usually transparent to hosts and applications
- Different motivations
  - Security: protect a network from attacks
  - Resource management: restrict usage of the network
- Different filtering criteria
  - Source and Destination IP address
  - Protocols: ICMP, TCP, UDP etc.
  - TCP and UDP port numbers (applications)
- Filter specification
  - Usually statically configured
  - Most configurations disallows packets for “non-standard ports”



# Firewalls and SIP

- Default configuration of many firewalls:
  - Allow outbound TCP and outbound UDP for DNS
  - Allow inbound TCP for well-known TCP ports only (e.g. HTTP)
- SIP communication disallowed in most cases
  - Port 5060 (UDP, TCP)
  - Relatively easy to fix: re-think filter firewall policy and update filter spec. to allow port 5060
    - Still a problem with SIP communication on other ports…
- The real problem:
  - Media sessions
  - Transport parameters are signalled by SIP messages
    - Negotiable, not known in advance
    - Different transport mechanisms for different applications
- Solution approaches
  - Have firewalls be controlled by ALGs ⇒ SIP proxy server
  - Adapt SIP to make it more firewall-friendly ⇒ SIP firewall traversal
  - Define firewall-control protocols to open firewalls dynamically
    - Issues: Authentication and authorization ⇒ MIDCOM

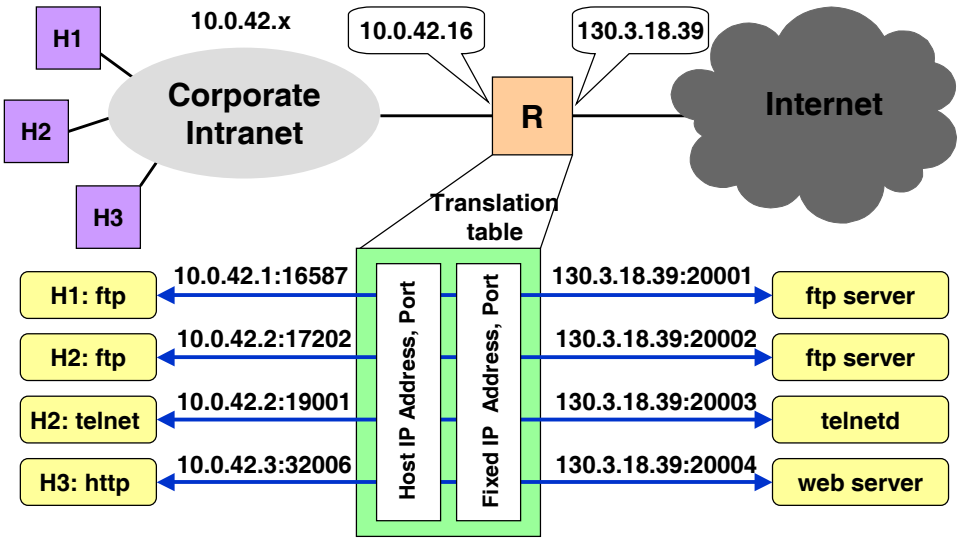




# Network Address Translators

- Intermediate systems that can translate addresses (and port numbers) in IP packets
  - Often used to map global addresses to address/port number combination of hosts in a corporate network
- Different motivations
  - Security
    - Make internal host inaccessible from the public Internet
  - Efficient usage of address space
    - Share one globally unique address
- NATs create address/port number mappings
  - Mappings are usually created dynamically, e.g. on connection setup
  - Static configurations also possible
  - Works best with connection oriented communication
  - Most common case: TCP connection from client-server sessions
    - Client in private address space, server in public Internet
  - NATs have to keep state for mappings that are tied to connections

# Network Address Translators








# NAT Traversal for SIP

- **Network Address and Port Translators**
  - Temporary address and port mappings
  - Private IP addresses for SOHOs (e.g. multiplexing single dialup-connection)
  - First line of defense in common firewall configurations
- **Problems**
  - Violation of the Internet's end-to-end-principle
  - NATs break many existing protocols
- **Solutions**
  - Midcom: control protocol for *middle boxes* (long term)
  - ALG-functionality in NATs (unlikely for most protocols)

→ **Solution: ICE**

# SIP Devices





## Sample SIP Phone Functionality

- Somewhat resemble a phone
  - More or less futuristic design
  - Two-line to color graphics display
  - Sometimes line power
- Basic SIP functionality
  - Registrations, voice calls (G.711, G.729, G.723, …)
- Expected "supplementary services"
  - Address book, short dials
  - Several "lines", speaker
  - Call hold, call transfer, conferencing, …
  - N-way conferencing for a few participants (local mixing)
- Some kind of CTI support
  - APIs, 3rd party call control, …
- HTTP server for manual user configuration
- Sometimes web browsers
- Autoconfiguration: DHCP, (t)ftp, …

# Autoconfiguration / Administration

- Key to large scale SIP deployments in enterprises
- Key to real plug & play products & broad deployment

- Aspects include
  - IP layer management
  - User and Device Profile Management
  - Software updates

- SIP Device Config Framework
  - Locate profile provisioning server
  - Register (authenticate) and retrieve user / device profiles
  - Receive notification of updates to profiles

- SIP Device Requirements
  - Collects expected functionality and behavior
  - Users, service providers, network admins, manufacturers, integrators
  - Representation of configuration data





# Some SIP Phones from our Lab









## Tutorial Overview

- Internet Multimedia Conferencing Architecture
- Packet A/V Basics + Real-time Transport
- SIP Introduction, History, Architecture
- SIP Basic Functionality, Call Flows
- SIP Security
- SIP Service Creation
- SIP in Telephony
- SIP in 3GPP ← You are here





# SIP and 3G

## SIP Support for Mobility

**Personal Mobility**

- SIP URIs
- SIP registration and authentication
- SIP call routing
- SIP routing to services

**Terminal Mobility**

- Ubiquitous IP connectivity (+ roaming)
- Mobile IP (orthogonal to SIP)
- Dynamic SIP session re-routing
- 3G link layer roaming mechanisms

**Service Mobility**

- Dynamic SIP session re-routing
- 3G Hand-off procedures

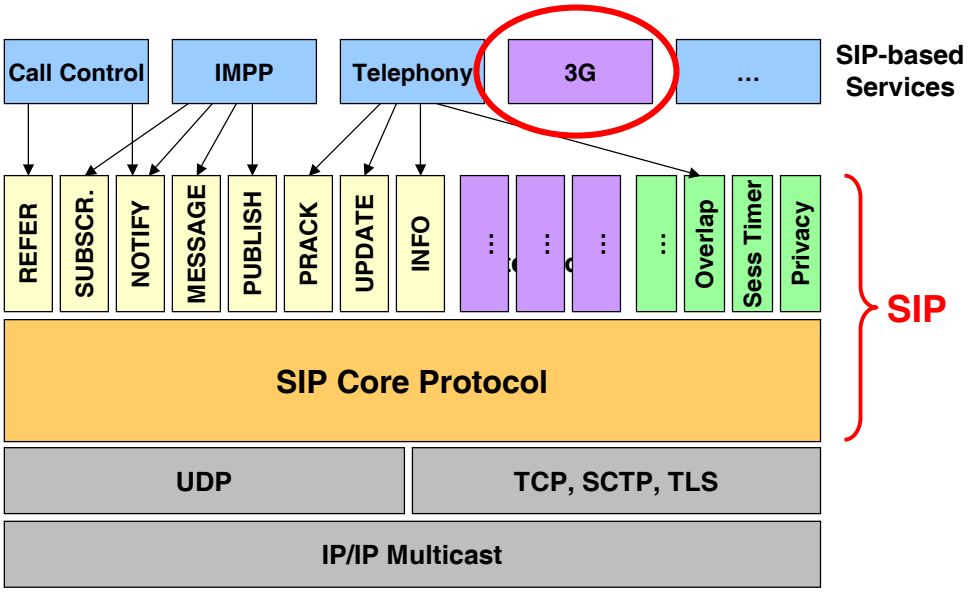




## SIP and 3GPP

- 3G networks offer two modes of operation:
  - circuits and packets
- Multimedia functionality based on packets
  - IP Multimedia Subsystem (IMS) in the Core Network (CN)
  - (exception: H.324 used for video telephony)
- IMS uses SIP for signaling
  - one piece out of a number of IETF protocols
- In the long run, all services shall converge to IP
- Release 5: SIP-based multimedia calls
- Release 6: Further SIP services to come (e.g. Presence, IM)

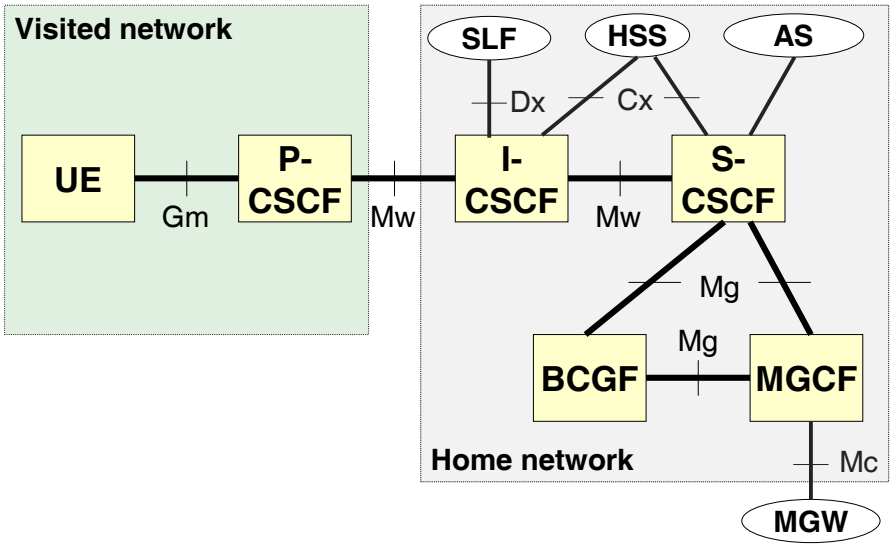
## SIP and 3G








## SIP-related Components in 3GPP



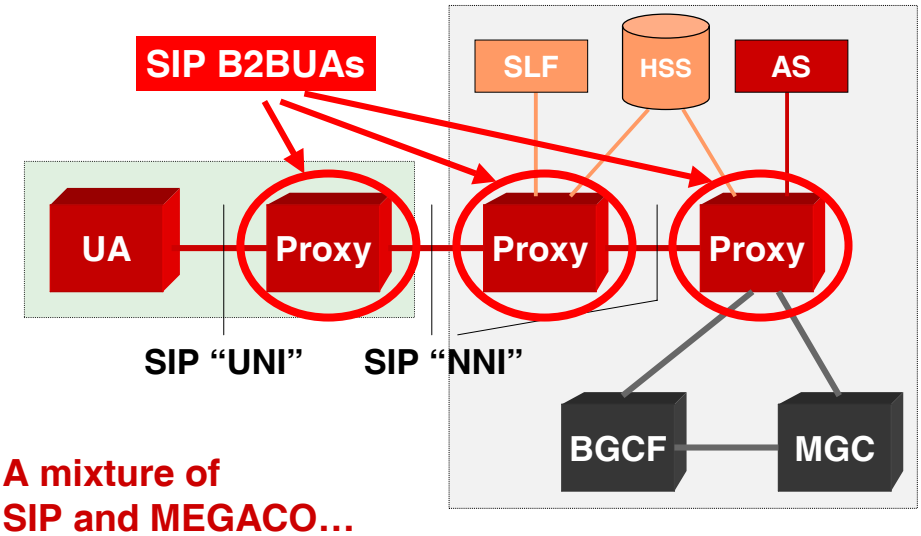
# A Collection of Acronyms

- UE — User Equipment
- P-CSFC — Proxy Call Session Control Function
- I-CSFC — Interrogating Call Session Control Function
- S-CSFC — Serving Call Session Control Function
- HSS — Home Subscriber Server
- AS — Application Server
- SLF — Subscription Locator Function
- BGCF — Breakout Gateway Control Function
- MGCF — Media Gateway Control Function
- MGW — Media Gateway



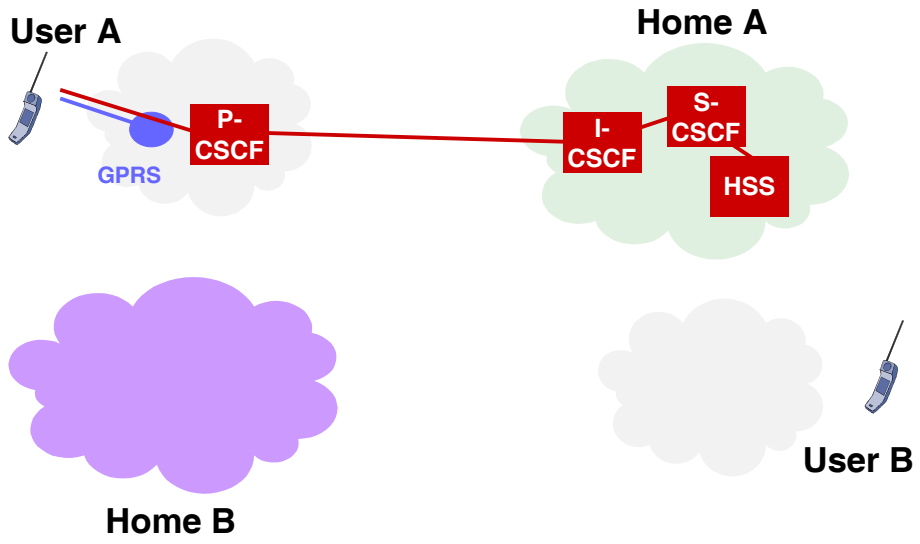


# SIP Components in 3GPP




**A mixture of SIP and MEGACO…**

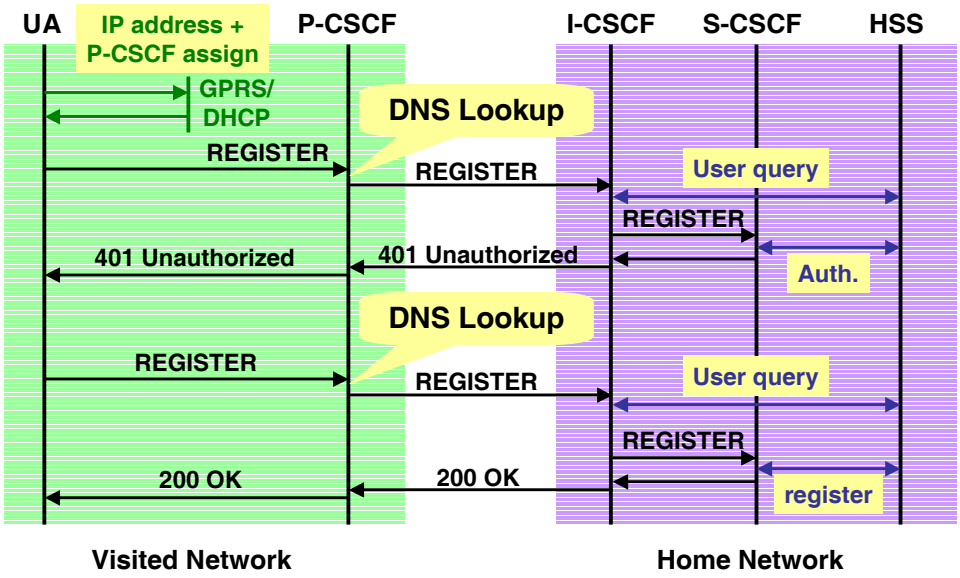
# 3G Roaming Scenario: Registration



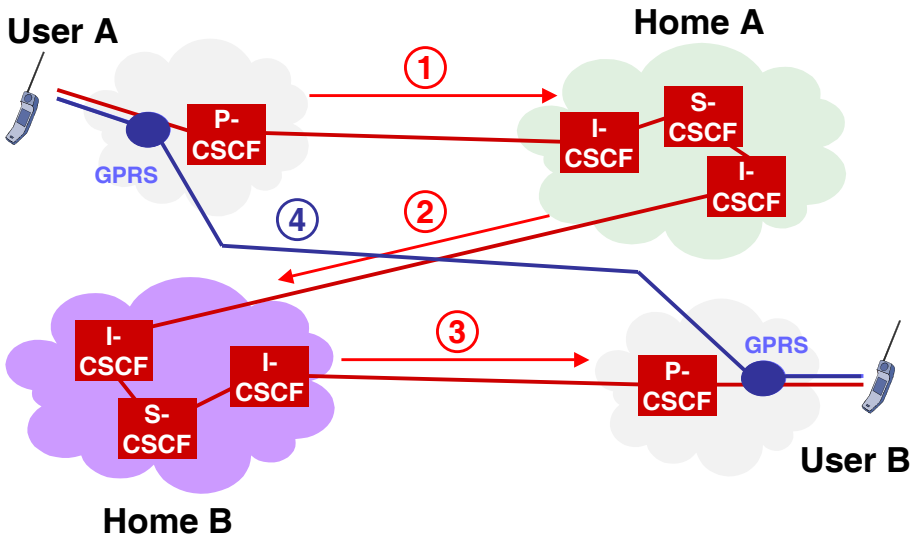





# SIP Registration of a Mobile Node



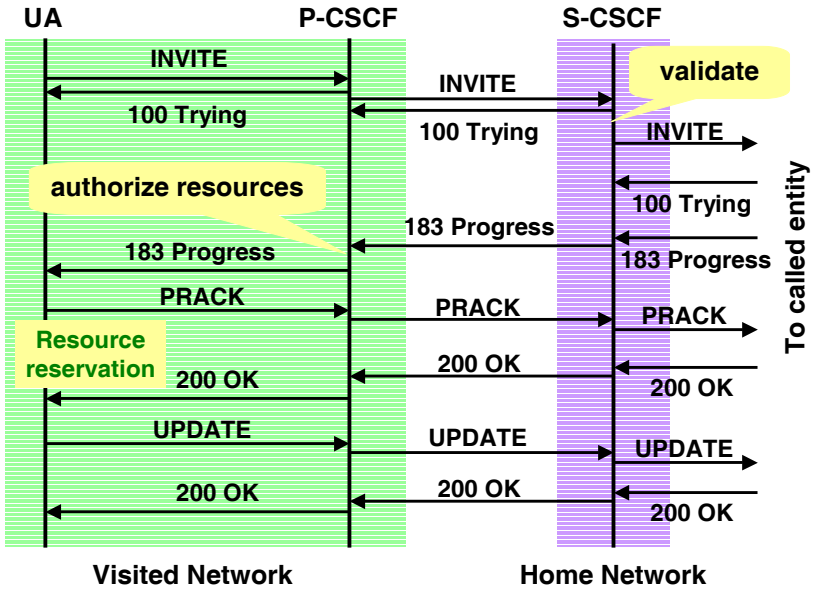
# 3G Roaming Scenario: Simple Call



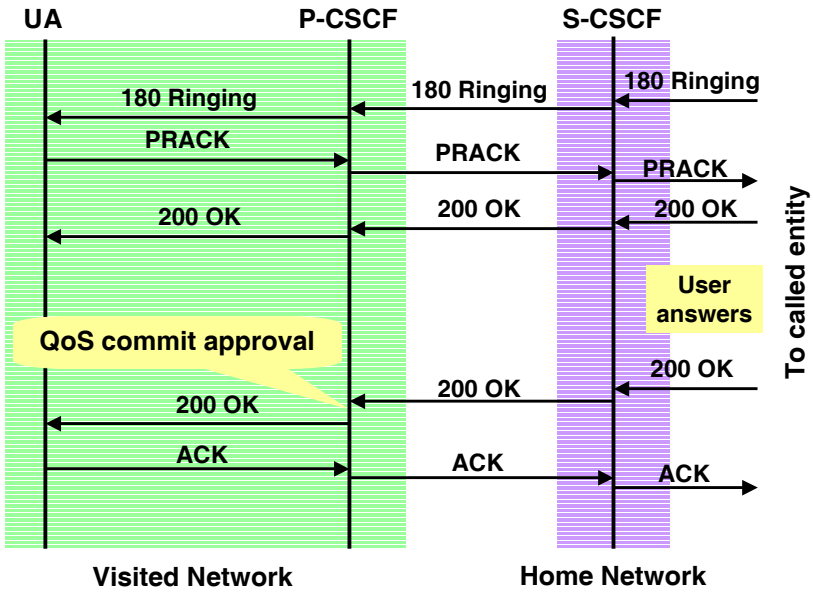





# Simple SIP Call: Caller Side (1)



## Simple SIP Call: Caller Side (2)








## Some further 3G Signaling

- Called side: pretty much the same
- Call termination by either endpoint: straightforward
  - Simple two-way handshake: BYE – 200 OK
  - Release associated GRPS resources
  - Accounting in S-CSCF/HSS: e.g. generate Call Detail Record (CDR)
- Call termination by "network"
  - S-CSCF generates SIP BYE requests to called and calling UA
- Terminal de-registration by user
  - Simple two-way SIP handshake: REGISTER – 200 OK
- Terminal de-registration by "network"
  - Uses SIP NOTIFY mechanism to inform the UA

## SIP Features used in 3G

- SIP Base Spec (RFC 3261)
- Reliable Provisional Response (PRACK) (RFC 3262)
- Session Description Protocol (RFC 2327)
- SDP Offer/Answer Scheme (RFC 3264)
- SUBSCRIBE / NOTIFY method (RFC 3265)
- SDP extensions for IPv6 (RFC 3266)
- UPDATE method (RFC 3311)

- Resource reservation extensions (“manyfolks”)
- Authentication and privacy

… among others … + … a number of extensions …





## Service Routing Extensions in 3G

- P-CSCF and S-CSCF perform central functions for UAs
  - Calls need to pass (at least) through both of them
  - to ensure all services are available
- Configure P-CSCF as outbound proxy
  - Address obtained during link layer initialization
- Record proxy chain to be traversed during registration
  - SIP extension for recording proxies to be traversed en-route
  - Path: header
  - Used together with loose source routing from S-CSCF to UA
- Learn about S-CSCF in REGISTER response
  - SIP extension for service routing
  - Service-Route: header
  - Used together with loose source routing (as defined in RFC 3261)

## Some SIP Security in 3G

- Basic Approach: closed network infrastructure
  - Transitive trust (and secure communications) within the network
- SIP Digest Authentication for registration
- Stateful network infrastructure
  - Only registered users can place calls
  - Ingress nodes (P-CSCFs) validate UA requests
  - S-CSCF additionally check validity per request
- P-Asserted-Identity: header
  - May contain network-generated (anonymized) identifier
- Privacy: header to select CLIP/CLIR
  - User id may be removed per user request when leaving the network

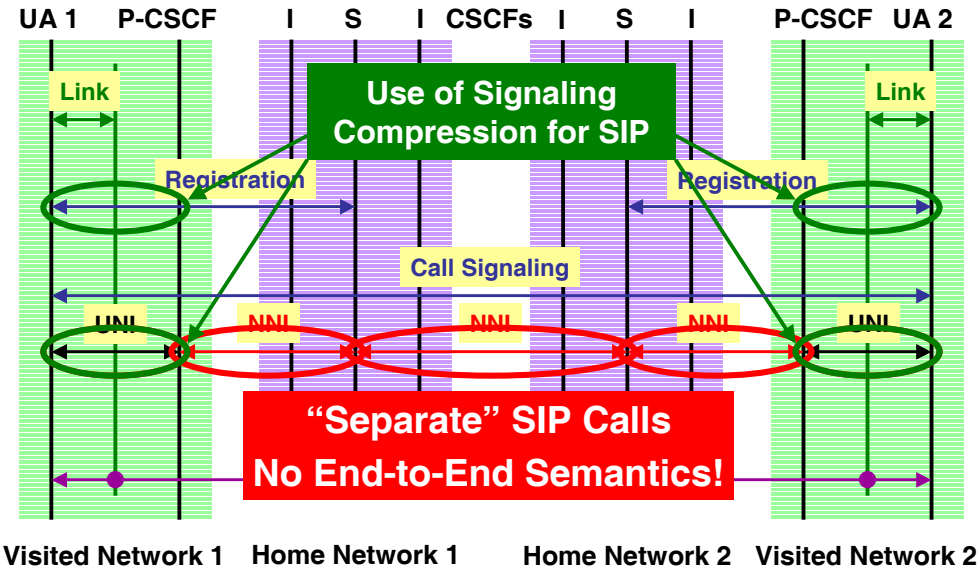




## Further 3G Extension Headers

- P-Associated-URI:
  - URIs allocated by service provider returned in REGISTER response
- P-Called-Party-ID:
  - To indicate original request URI to the called party
- P-Visited-Network-ID:
  - Conveys the identifier of the currently visited network
- P-Access-Network-Info:
  - Reports information about the currently used access radio link
- P-Charging-Function-Addresses:
  - Indicates where to report charging information (e.g. CDRs) to
- P-Charging-Vector:
  - Enable correlation of charging information across the network

# Summary: Signaling and Media Flows in 3G








# Service Creation in 3GPP

- Standardized functional building blocks in the network
  - services to be created on top of those
  - Partially standardized services
- "APIs" for service providers
- SIP for interfacing to Application Servers
- Service creation in the network
  - "closed" environment
  - potential for limited outside access
- But: No real end-to-end signaling
  - Header fields may be removed by P-CSCF
  - New methods may not be passed through
- No / limited end user or third party innovation possible

## Conclusion: SIP and 3G

**3G is partially built the *old* way – the telephony way…**

- **Breaks the (S)IP end-to-end model**
  - **built-in limitation of innovation**
- **Enforces net-centric service creation**
  - **built-in protection against newcomers**
- **Closed architectural model**
  - **built-in gateways even to external SIP devices**

**(obvious economic motivations for the above)**

- **Many boxes, many lines, many interfaces**
  - **High degree of complexity? (! KISS)**





## Further Information

**http://www.ietf.org/html.charters/sip-charter.html**
**http://www.ietf.org/html.charters/sipping-charter.html**
**http://www.ietf.org/html.charters/simple-charter.html**
**http://www.ietf.org/html.charters/xcon-charter.html**

**http://www.softarmor.com/sipwg/**
**http://www.softarmor.com/sippingwg/**
**http://www.softarmor.com/simple/**

**http://www.cs.columbia.edu/sip/**

**http://www.cs.columbia.edu/sip/sipit/**