

# Session Initiation Protocol (SIP) Overview

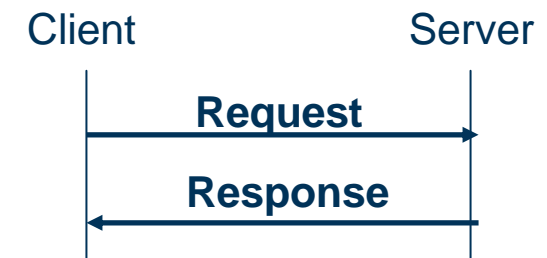
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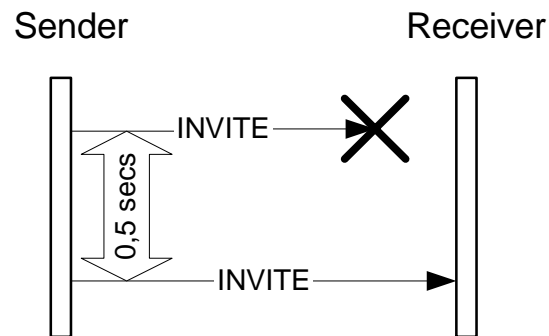
# SIP general overview

- SIP is a end-to-end, client-server, extensible, text based protocol.
- The design base was HTTP and SMTP
- SIP was originally used to establish, modify and terminate multimedia sessions in the internet.
- SIP has evolved to be able to set-up a broad range of sessions:
  - Multimedia (e.g., voice, video, etc)
  - Gaming
  - Presence and Instant Messaging
- SIP messages are either requests or responses.
- SIP messages carries zero or more “bodies”.
- SDP is the common body for session initiation.
- SIP runs on any transport protocol (UDP, TCP, TLS, SCTP)
  - The spec mandates UDP and TCP. Other transport protocols are optional

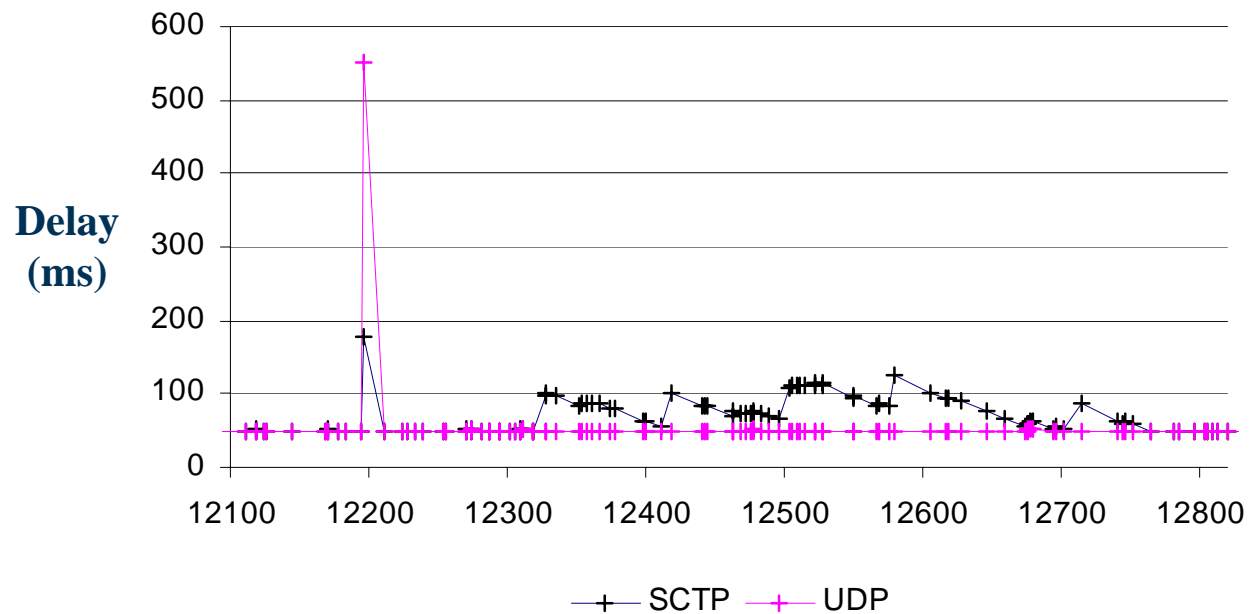


# UDP: User Datagram Protocol

- Pros
  - No connection establishment time
- Cons
  - No congestion control
  - More state information to be stored at the application layer
  - No transport layer fragmentation

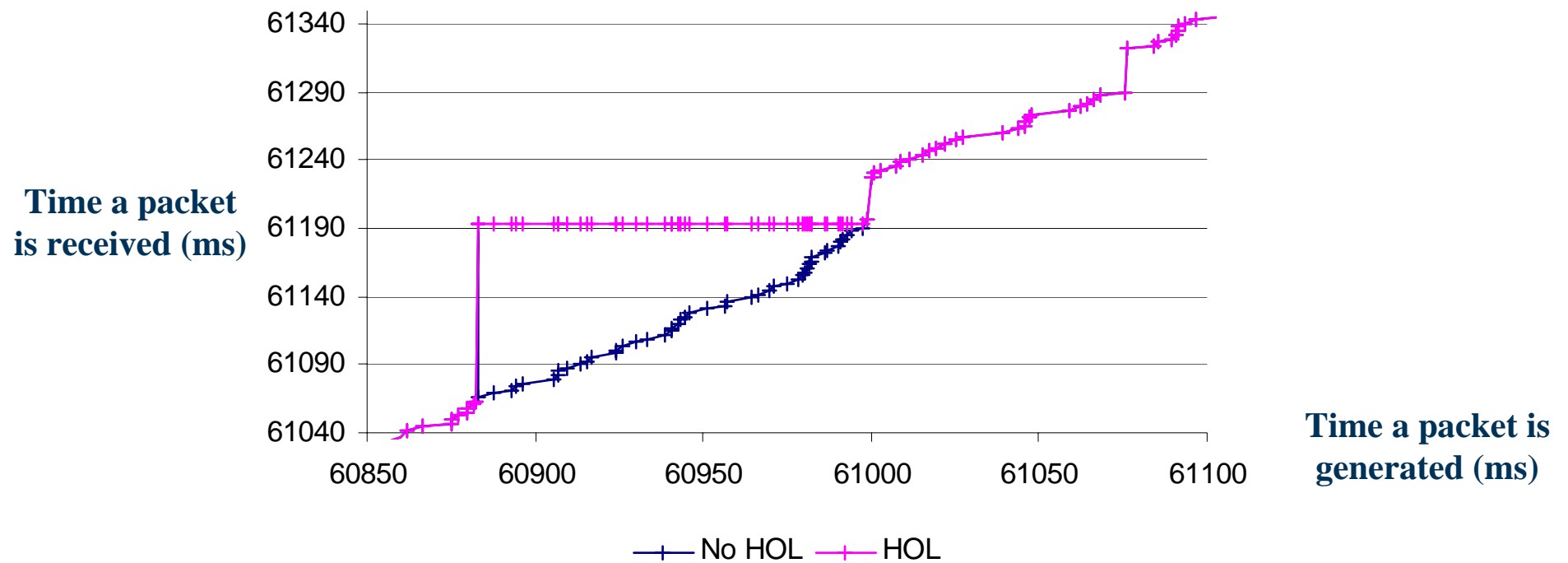


# Timeouts vs. fast retransmit



**Time a packet is generated (ms)**

# HOL blocking

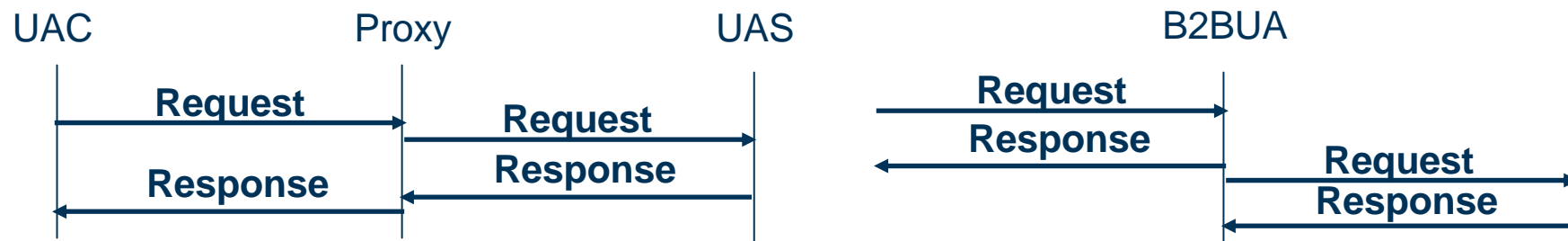


# Overview of SIP functionality

- SIP provides the following functionality:
  - User location (not geographical location)
  - User availability
  - User capabilities
  - Session set-up
  - Session management
- SIP does not provide services
  - But it enables the system to provide services
  - It has been demonstrated that it is easy to provide services with SIP

# SIP logical entities

- User Agent (UA): An endpoint
  - User Agent Client (UAC): sends requests, receives responses
  - User Agent Server (UAS): receives requests, sends responses
- Proxy server: A network host that proxies requests and responses, i.e., acts as a UAC and as a UAS.
- Redirect server: a UAS that redirects request to other servers.
- Back-to-back User Agent: a UAS linked to a UAC
  - Acts as a UAS and as a UAC linked by some application logic
- Registrar: A special UAS that accepts only registrations



# Stateless and stateful SIP proxies

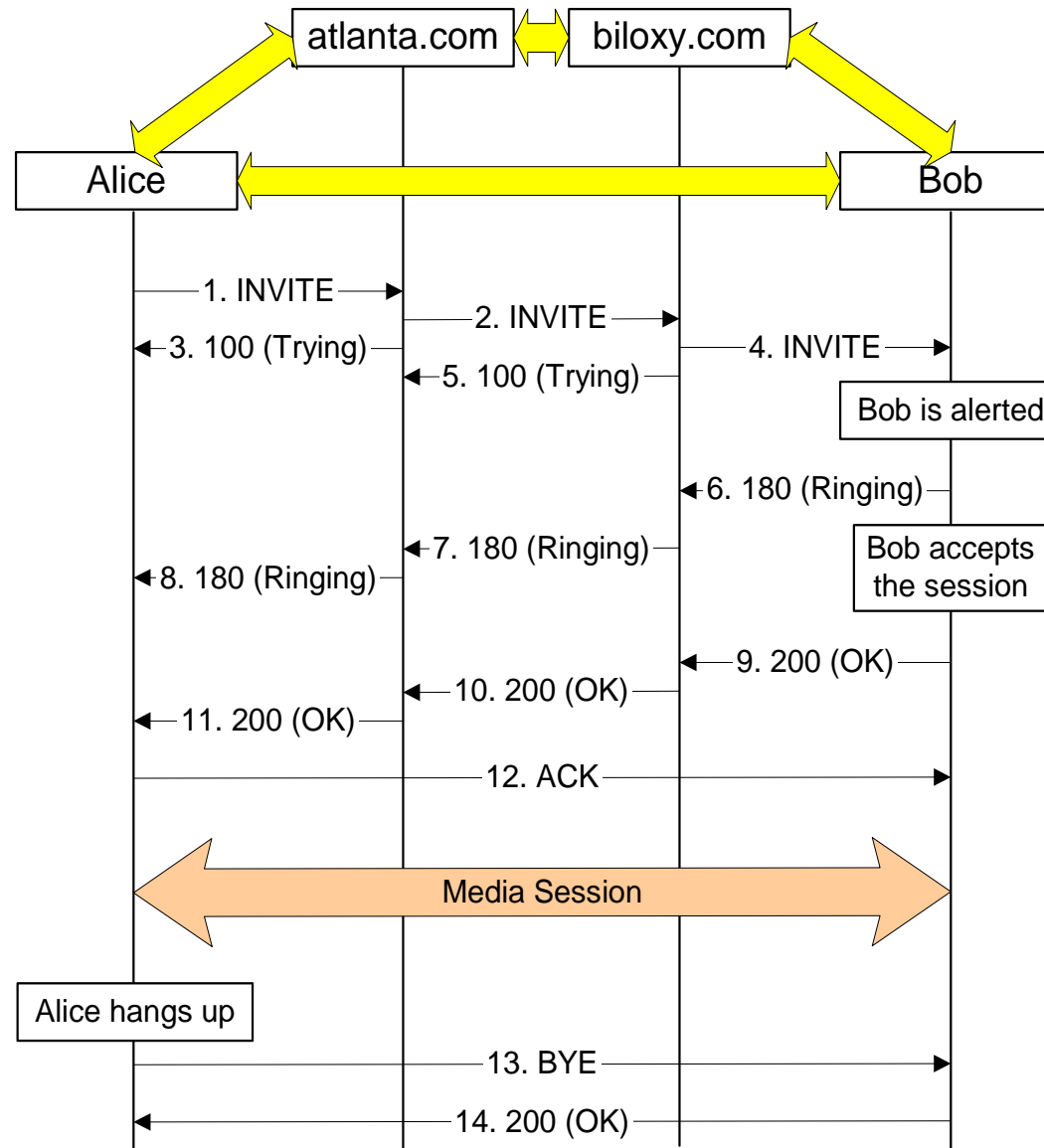
There are several types of SIP proxies, depending on the state they keep:

**1.Stateless proxy:** a proxy that does not keep any state when forwarding requests and responses.

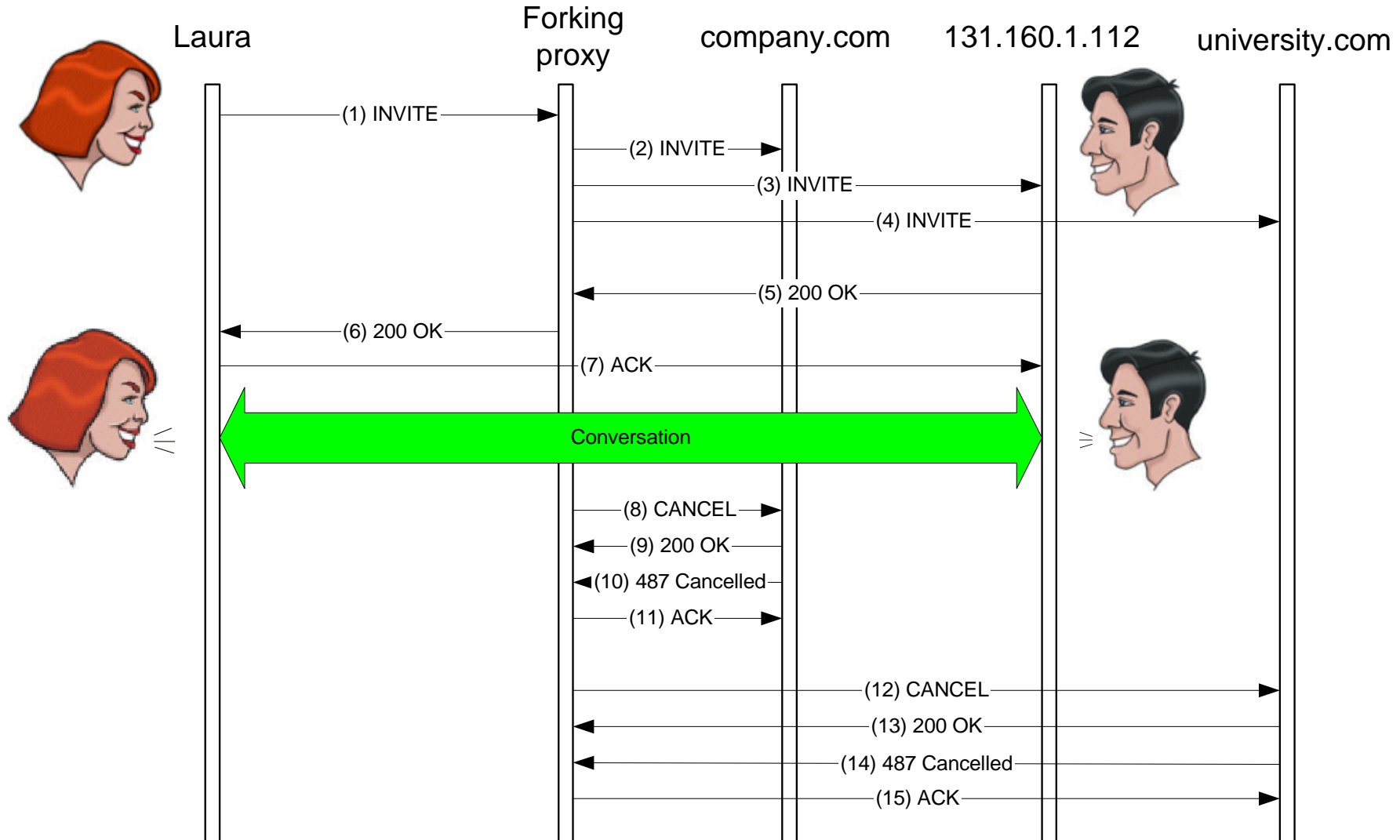
**2.Transaction stateful proxy, or stateful proxy:** a proxy that stores state during the duration of the transaction.

**3.Call stateful proxy:** a proxy that stores all the state pertaining to a session (e.g., from INVITE to BYE). A call stateful proxy is always a transaction stateful proxy, but not the other way round.

# The SIP trapezoid



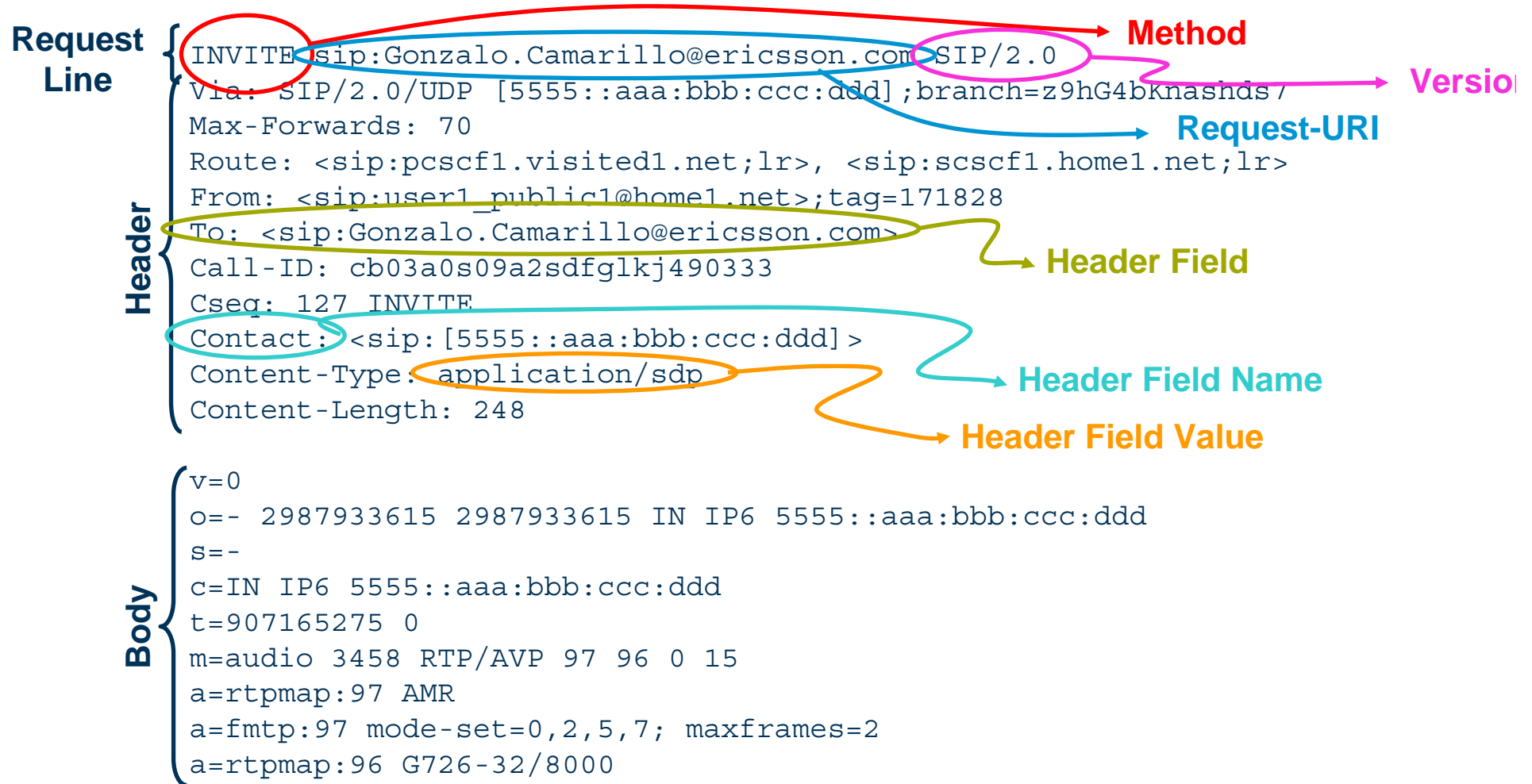
# Forking Proxy Example



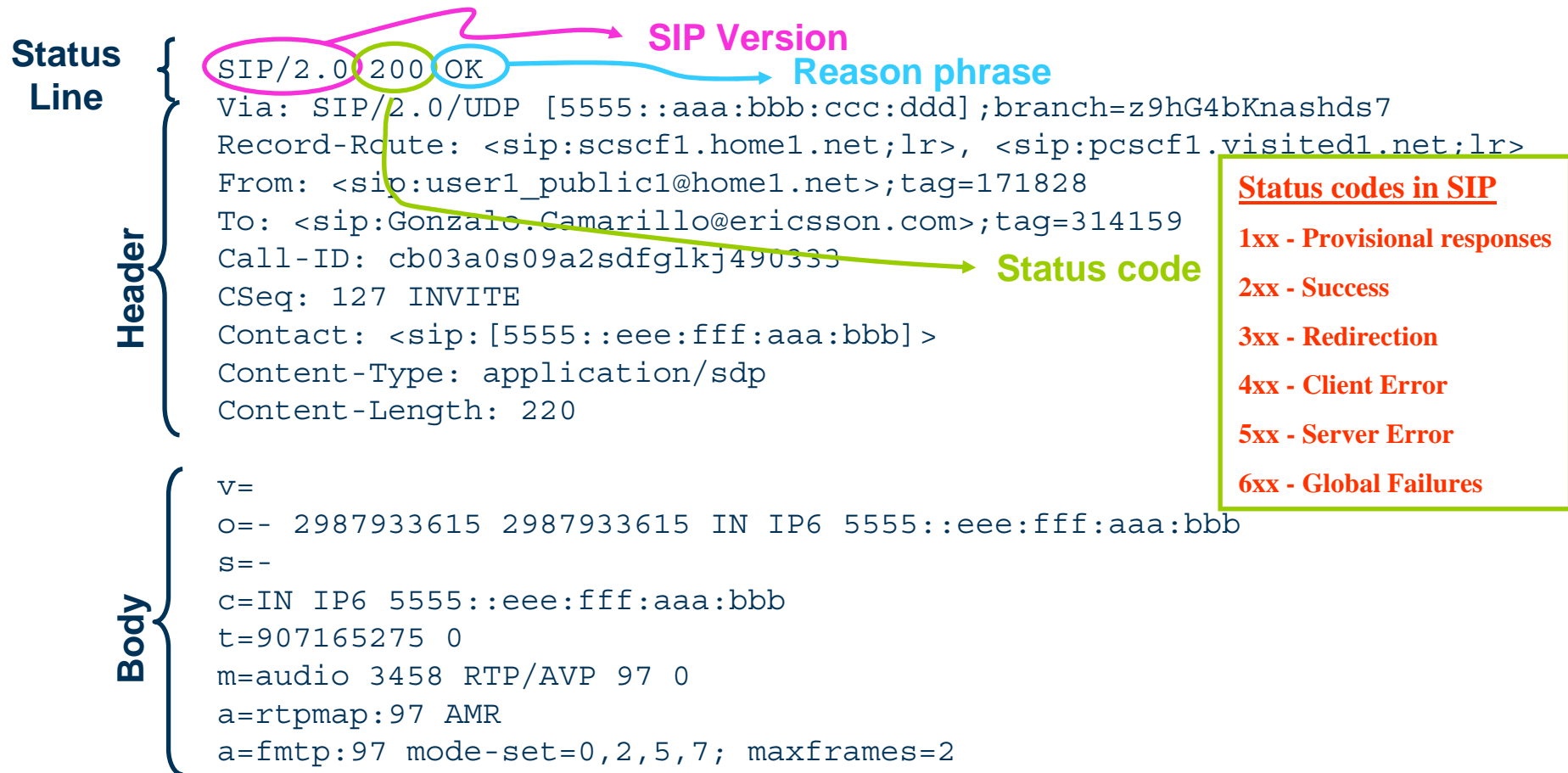
# SIP addresses

- SIP uses Uniform Resource Identifiers (URIs). At least, SIP URIs and SIPS URIs are supported, although others (such as TEL URL) are commonly supported.
  - sip:Gonzalo.Camarillo@ericsson.com
  - sips:Gonzalo.Camarillo@ericsson.com
  - tel:+358-9-299-3371
  - sip:proxy.atlanta.com:5060
  - sip:another-proxy.biloxi.com;transport=UDP
- SIP and SIPS URIs must include a host name, and may include username, may include port numbers, may include parameters
- Address space is unlimited
- Non SIP/TEL URIs are also valid under certain circumstances: HTTP, IM, PRES, MAILTO...

# An example of a SIP request

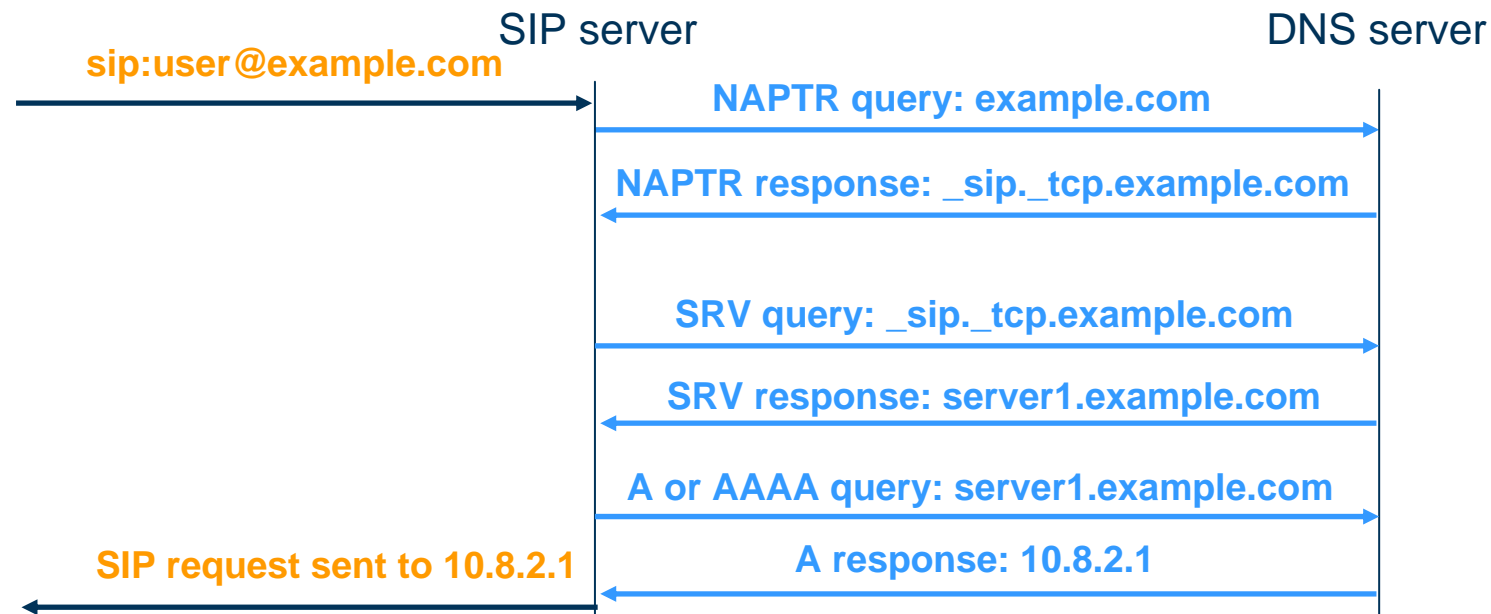


# An example of a SIP response

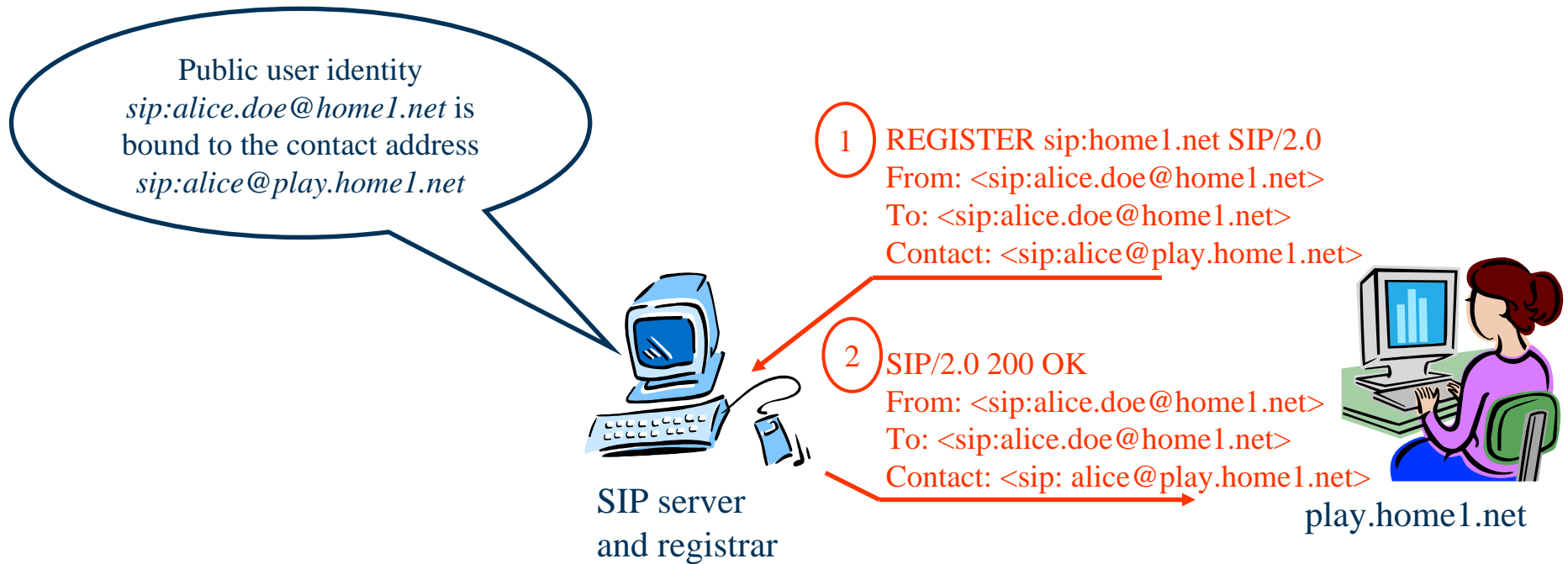


# SIP routing and Domain Name System

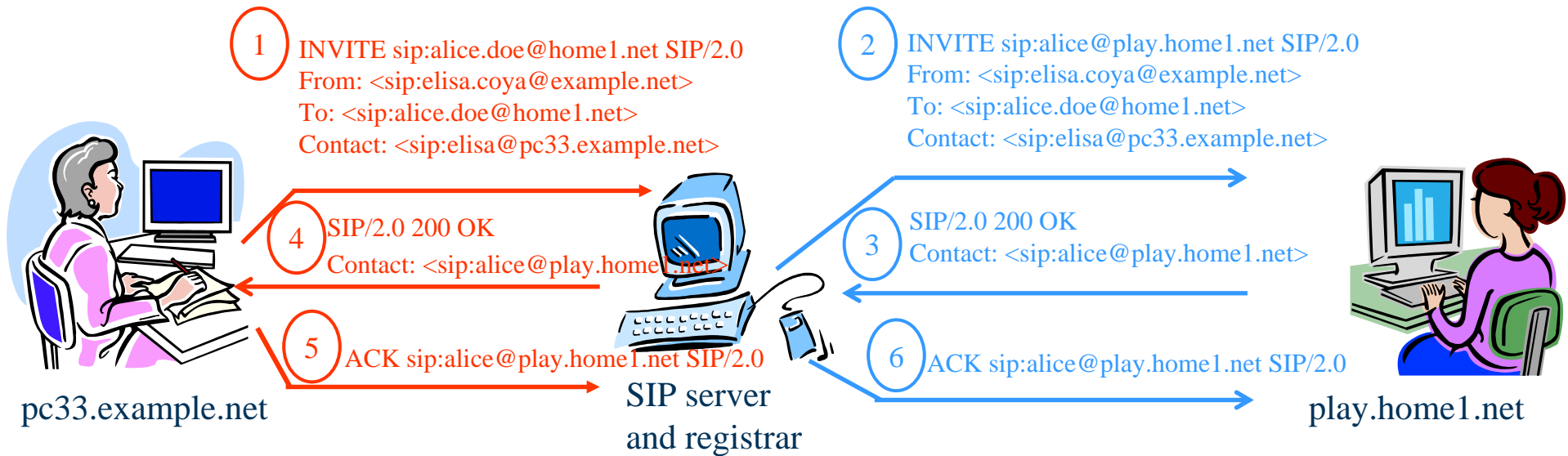
- SIP clients use DNS to route requests and find the next hop to route the request
  - By looking into a NAPTR (Naming Authority Pointer) record in DNS
  - By looking into a SRV (Services) record in DNS
  - By looking into A (IPv4) or AAAA (IPv6) records in DNS



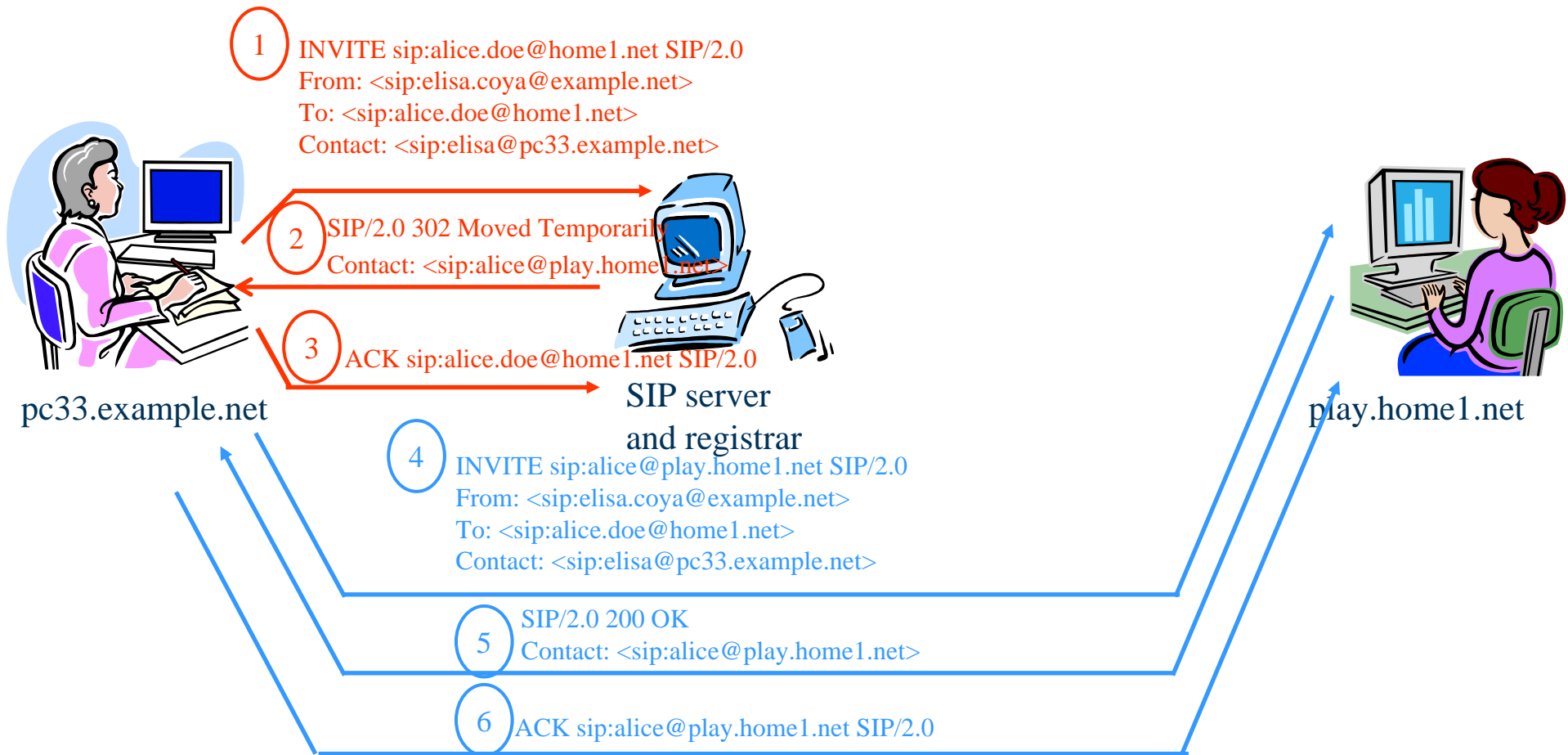
# SIP registration



# Routing: SIP server in proxy mode



# Routing: SIP server in redirect mode



# Summary of SIP methods

- INVITE: create a session
- BYE: terminates a session
- ACK: acknowledges a final response for an INVITE request
- CANCEL: cancels an INVITE request
- REGISTER: binds a public SIP URI to a Contact address
- OPTIONS: queries a server for capabilities
- SUBSCRIBE: installs a subscription for a resource
- NOTIFY: informs about changes in the state of the resource
- MESSAGE: delivers an Instant Message
- REFER: used for call transfer, call diversion, etc.
- PRACK: acknowledges a provisional response for an INVITE request
- UPDATE: changes the media description (e.g. SDP) in an existing session
- INFO: used to transport mid-session information
- PUBLISH: publication of presence information

# Application areas of SIP

- SIP was originally developed to establish multimedia sessions on the internet (audio, video)
  - Mainly multicast sessions
  - Also unicast sessions
- But has evolved to support other aspects of communications:
  - Voice over IP calls between SIP terminals
  - Voice over IP calls between Gateways (SIP-T)
  - Gaming sessions
  - Instant messaging
  - Presence
  - Multimedia conferences
  - Machine-to-machine communication (e.g., vending machine notifications)

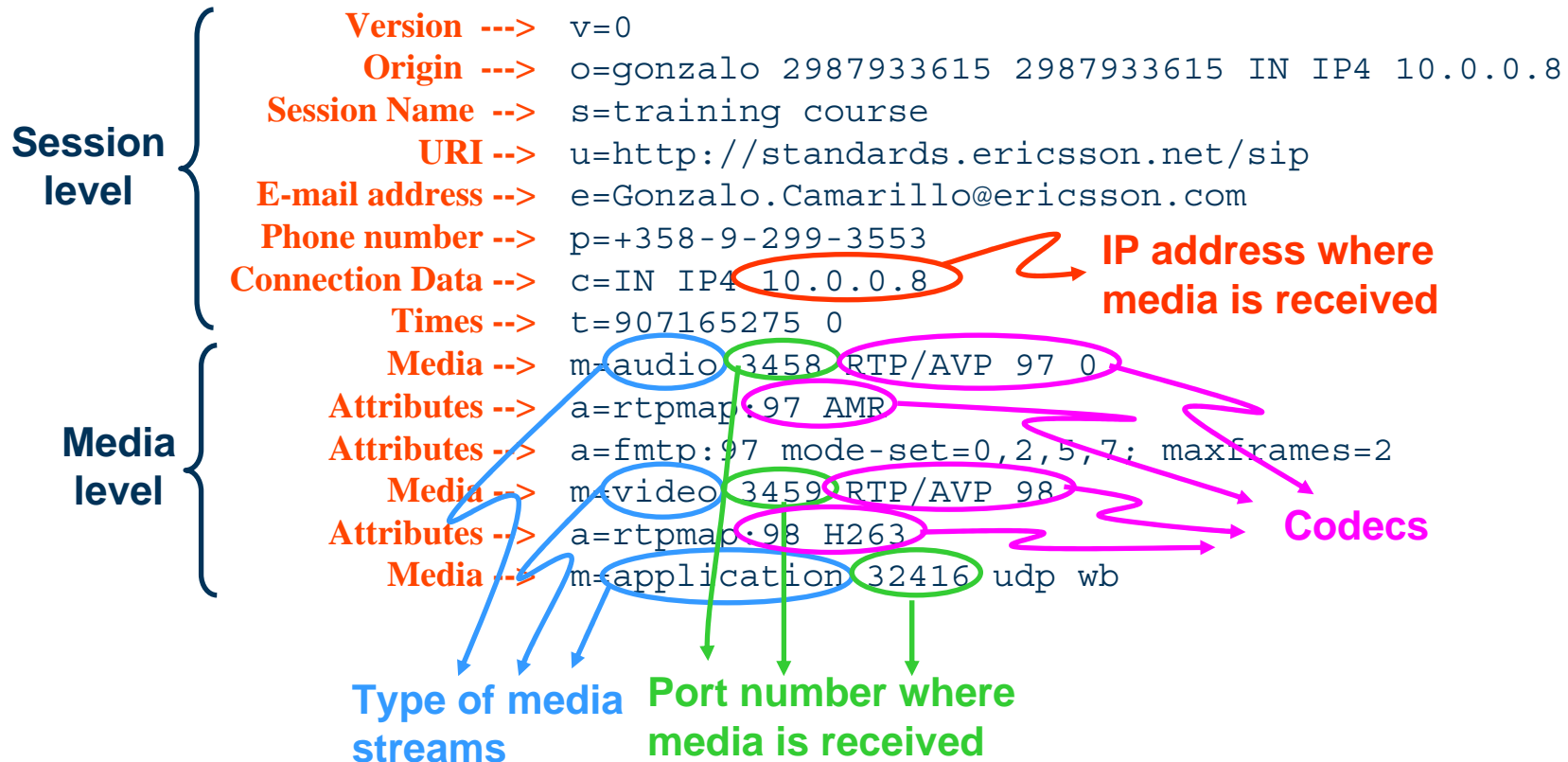
# Modularity of SIP

- SIP is a modular extensible protocol. Depending on the desired application, some functions may or may not be present in an implementation.
  - Because of extensions
  - Or because of options in the core protocol.
  - Unlike ISUP, there are not different SIP flavours
  - But there are different functionalities, security mechanism, methods, headers, options, transport protocols, etc., that may or may not be implemented.
  - Generally, SIP contains mechanisms to discover what is supported by a proxy or remote end.
    - Require, Supported, Proxy-Require, Allow headers
    - Contact header in registration

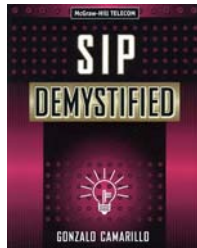
# Session Description Protocol (SDP)

- SDP is a session description protocol for multimedia sessions
- SDP is used to describe the set of media streams, codecs, and other media related parameters supported by either party.
- All SIP implementations **MUST** support SDP, although they can support other bodies
- Used by other protocols than SIP: RTSP, SAP, etc.
- SDP was initially developed to support multicast sessions in the Internet. Gradually tailored for SIP purposes.

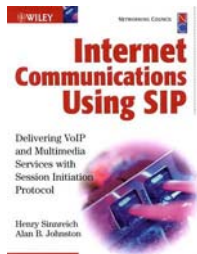
# SDP example



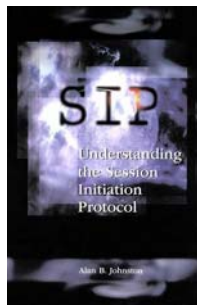
# Books



- G. Camarillo: "SIP Demystified". McGraw-Hill Professional, 2001.



- H. Synnreich, A. B. Johnston: "Internet communications using SIP". John Wiley and Sons, 2001.



- A. B. Johnston: "SIP: Understanding the Session Initiation Protocol", Artech House Telecommunications Library, 2001 (2<sup>nd</sup> edition, 2004).

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