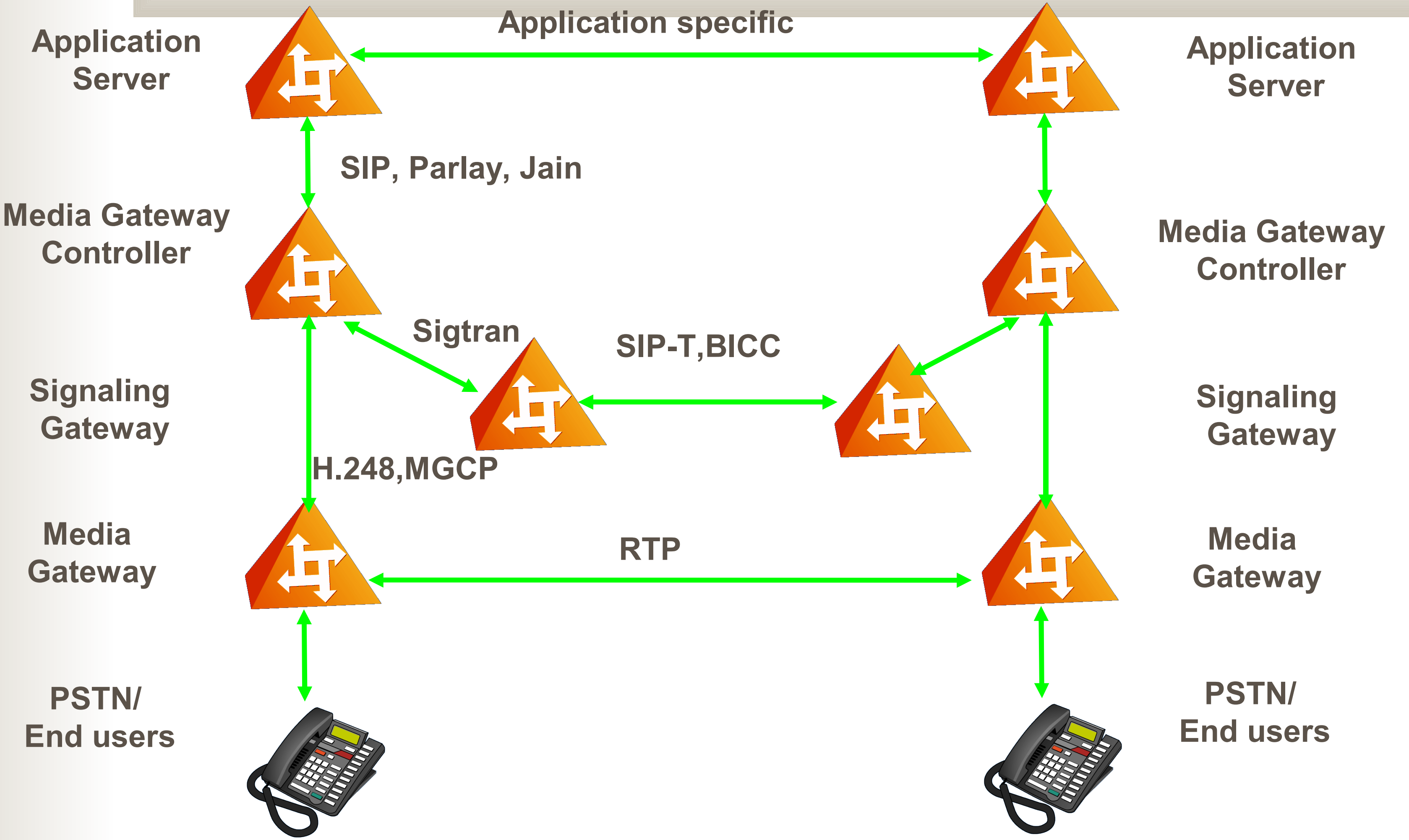
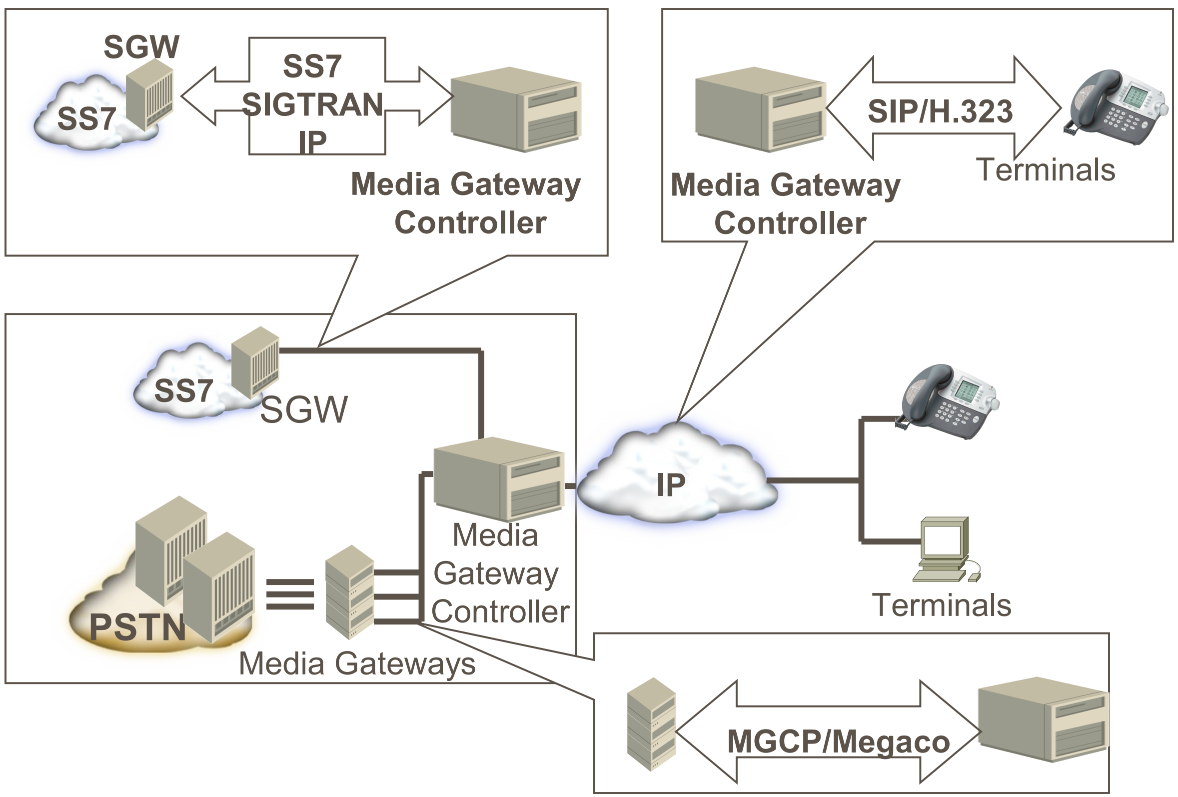
Softswitch Architecture – Interdomain protocols



NGN Protocols



- **SIP : SESSION INITIATION PROTOCOL**

WHAT IS A PROTOCOL

- A protocol is a set of rules to be used for communication between two entities. Elements of a protocol are :

2. Syntax: Refers to “How” part. The data format e.g.TCP/IP header
3. Semantics: Refers to “What” part. The control information communicated.
4. Timing: Refers to “ When” part. Synchronous or asynchronous.

What is a Session?

- It is a temporary communication relationship among a group of objects in the service stratum that are assigned to collectively fulfill a task for a period of time
- A state of a session may change during its lifetime
- Session based communications may be one-to-one, one-to-many, many-to-one or many-to-many

What is SIP ?

- The Session Initiation Protocol (SIP) is an application layer Signaling Control Protocol used to establish, maintain and terminate multimedia sessions.
- Multimedia sessions include Internet Telephony, conferences and other similar applications involving such media as audio, video and data.
- SIP is a protocol from IETF and is defined in RFC 2543 and RFC 3261

Why SIP?

SIP isn't simply 'voice',
it supports services that provide
'integrated personal communications environments'
that include 'voice'.

The last service of the voice world, or the first service of the interactive multimedia world?

SIP Characteristics

- User to User protocol
- Establishes, modifies, and tears-down sessions
  - Relies on other protocols for transport, QoS, Accounting
  - Allows multi-party sessions; multi-casting
- SIP itself is independent of session type
  - Supports multi-service sessions
- Provides authentication and privacy
  - Signaling only, not bearer data
- Supports personal mobility
- SIP itself is a text based signaling protocol
  - Heavily influenced by HTTP

SIP Characteristics

- SIP is dependent upon other protocols like RSVP for QoS, RTP for media transport, Real Time Streaming Protocol (RTSP), Session Description Protocol (SDP) for negotiation of user capabilities and other parameters etc.

SIP Characteristics

- SIP supports all facets of establishing and terminating multimedia communications:
  - User Location: determination of the end system to be used for communication
  - User Capabilities: determination of the media and media parameters to be used
  - User availability: determination of the willingness of the called party to engage in communication
  - Call setup: “ringing”, establishment of call parameters at both called and calling party
  - Changing the session parameters during the call

SIP ARCHITECTURE

SIP Components

- SIP basically has two components
  - User Agent (End system/terminal)
  - Network Servers (Proxy, Redirect, Registrar)

SIP Components
User Agents

- User agents are end-system applications that contain both a user agent client (UAC) and a user agent server (UAS), otherwise known as *client* and *server* respectively
  - Client – Initiates SIP requests and acts as the user's calling agent
  - Server- Receives requests and returns responses on behalf of users

SIP Components
Network Servers

- Three types of network servers can exist in SIP network
  - Proxy Server – Acts on behalf of other clients and contains both client and server functions. A proxy server interprets and can rewrite request headers before passing them on to other servers. Rewriting the headers identifies the proxy as the initiator of the request and ensures that replies follow the same path back to the proxy instead of client

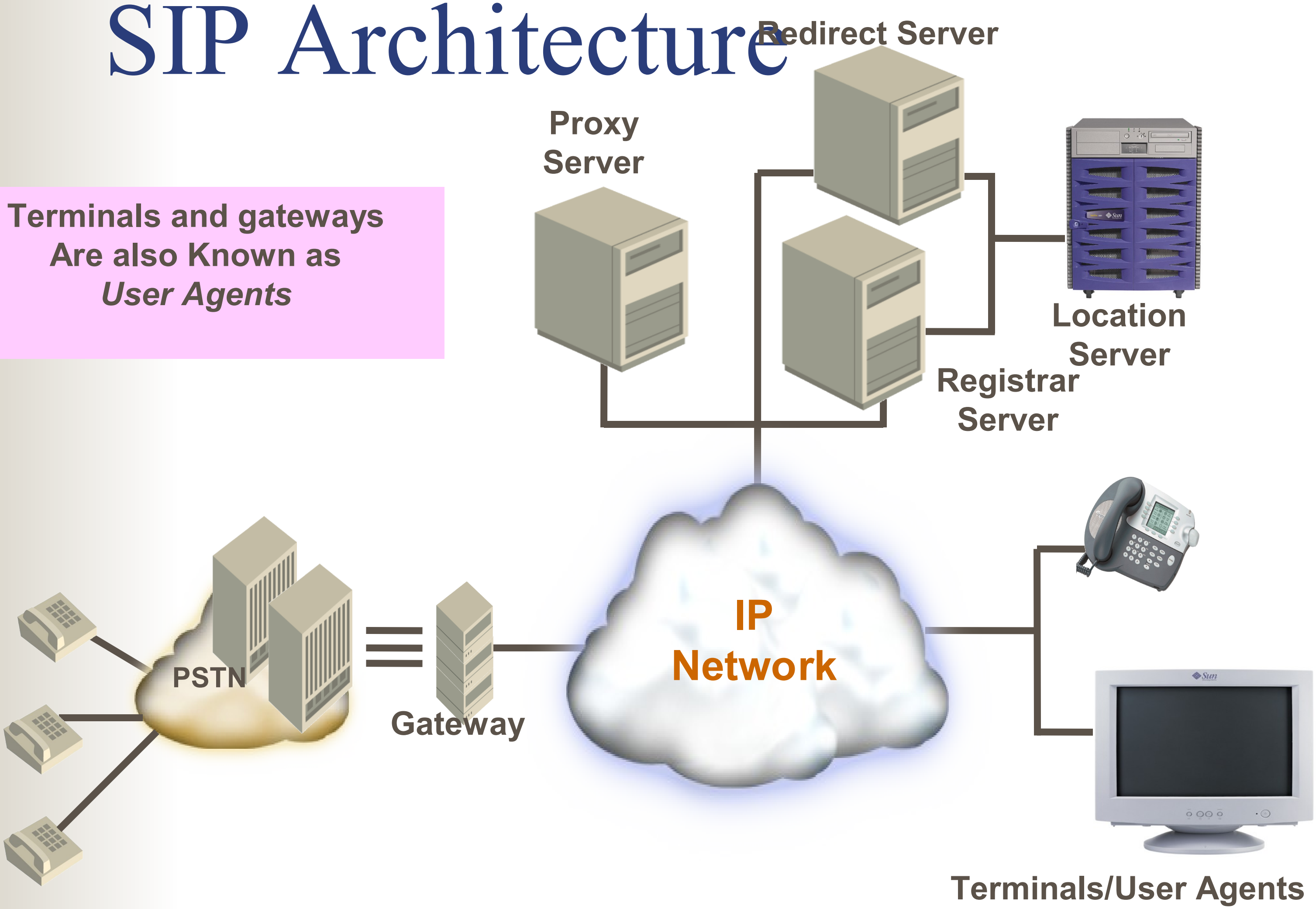
SIP Components
Network Servers

- Redirect Server – Accepts SIP requests and sends a redirect response back to the client containing the address of the next server. Redirect servers do not accept calls, nor do they process or forward the SIP requests.

- Registrar Servers – A registrar server is a server that accepts REGISTER requests and maintains the availability details of various servers and clients. A registrar is typically co-located with a proxy or redirect server and may sometime offer location services also.

SIP Architecture

Terminals and gateways
Are also Known as
User Agents



SIP Addressing

- The SIP Uniform Resource Identifier (URI), sometimes called the Uniform Resource Locator (URL), identifies participants in a SIP session.
- SIP:user@host
  - Similar to HTTP URI
  - Can designate an individual,……
    - Sip:dgmalttc@bsnl.co.in

PSTN Addressing Conventions

SIP:2145551212@gateway

Tel:214-555-1212

Can initiate a session with PSTN

SIP Messages

- SIP messages are typically of type *requests* and *responses*. Requests flow from client to server and a response from server to a client
- SIP messages are send either in TCP or UDP.
- If the protocol type is not listed in URI the client must first attempt to connect using UDP and then TCP
- Being text based protocol, headers are largely self describing

SIP Messages

- The SIP request starts with a command line. The command line indicates the type of request that is being sent, a field request URI giving the detail of destination and the version of SIP protocol
- Several headers follow the command line
  - *To* and *From* (Calling and called party identity)
  - *Call-Id* is a unique token that identifies session
  - *Via* defines the path that the request has taken
  - *Cseq* defines the sequence number of the request

SIP Messages

- The SIP message optionally contains a message body. The content and structure of message body is variable
  - The content type and content-length headers specify the form and size of the message body
  - The message body will typically contain a Session Description Protocol (SDP) messages that contain information about the session, such as media coding format and IP address and port number

SIP Message Format

Command Line

```
INVITE sip:vonod@mumbai.tcs.co.in SIP/2.0
```

Headers

```
Via :SIP/2.0/UDP anilworkstation.com
From:Anil<sip:anil@delhi.tcs.co.in>
To: Vinod Bhat <sip:vinod@mumbai.tcs.co.in>
Call-ID: 123456789 @anilworkstation.com
Cseq: 1 INVITE
Content -type : application/sdp
Content-length : xxxx
```

Message Body (Variable)

```
V=0
O= anil 28960783 IN IP4 157.277.12.184
S= Urgent Phone call from Anil
T=3149329600  3149329755
C=IN IP4 anilworkstation.com
M=audio  5004  RTP/AVP o 3.5
a=rtpmap:0 PCMU/8000
A=rtpmap:3 GSM/8000
```

- **v**—Tells the SDP version
- **o**—Lists the organization of the calling party
- **s**—Describes the SDP message(subject)
- **c**—Lists the IP address of the originator
- **t**—Tells the timer value
- **m**—Describes the media that the originator expects
- **a**—Gives the media attributes

SIP Message Format

- Command Line
  - Command/Request line format:
    - Request Line = Request method@URI SIP/Version
- Headers
  - There are 44 SIP headers listed in RFC 2543. These headers can be divided into four different groups
    - General Headers
    - Request Headers
    - Response headers
    - Entity Headers

SIP Message Request Methods

- The following six methods are supported by SIP for generating requests
  - INVITE
  - ACK
  - OPTIONS
  - BYE
  - CANCEL
  - REGISTER

A new method INFO is under finalisation state.

SIP Message Responses

- Six types of response codes are defined in SIP
  - 1xx: Informational – request received, continuing to process the request
  - 2xx: Success- the action was successfully received, understood and accepted
  - 3xx: Redirection - further action needs to be taken in order to complete the request
  - 4xx: Client error- the request contains bad syntax or cannot be fulfilled at this server
  - 5xx: Server error – the server failed to fulfill an apparently valid request
  - 6xx: Global failure – the request cannot be fulfilled at any server

SIP Messages
Response Codes

- Some Important Response codes are:
  - 1xx Provisional
    - 100 – continue
    - 180 – ringing
  - 2xx Success
    - 200 – OK
  - 3xx Redirect
    - 301 moved permanently
    - 302 moved temporarily
  - 4xx Client error
    - 400 bad request
    - 401 unauthorized

SIP Messages
Response Codes

- 5xx: Server error
  - 500 Server internal error
  - 502 bad gateway
- 6xx: Global failure
  - 600 busy
  - 604 does not exist

SIP Message Body

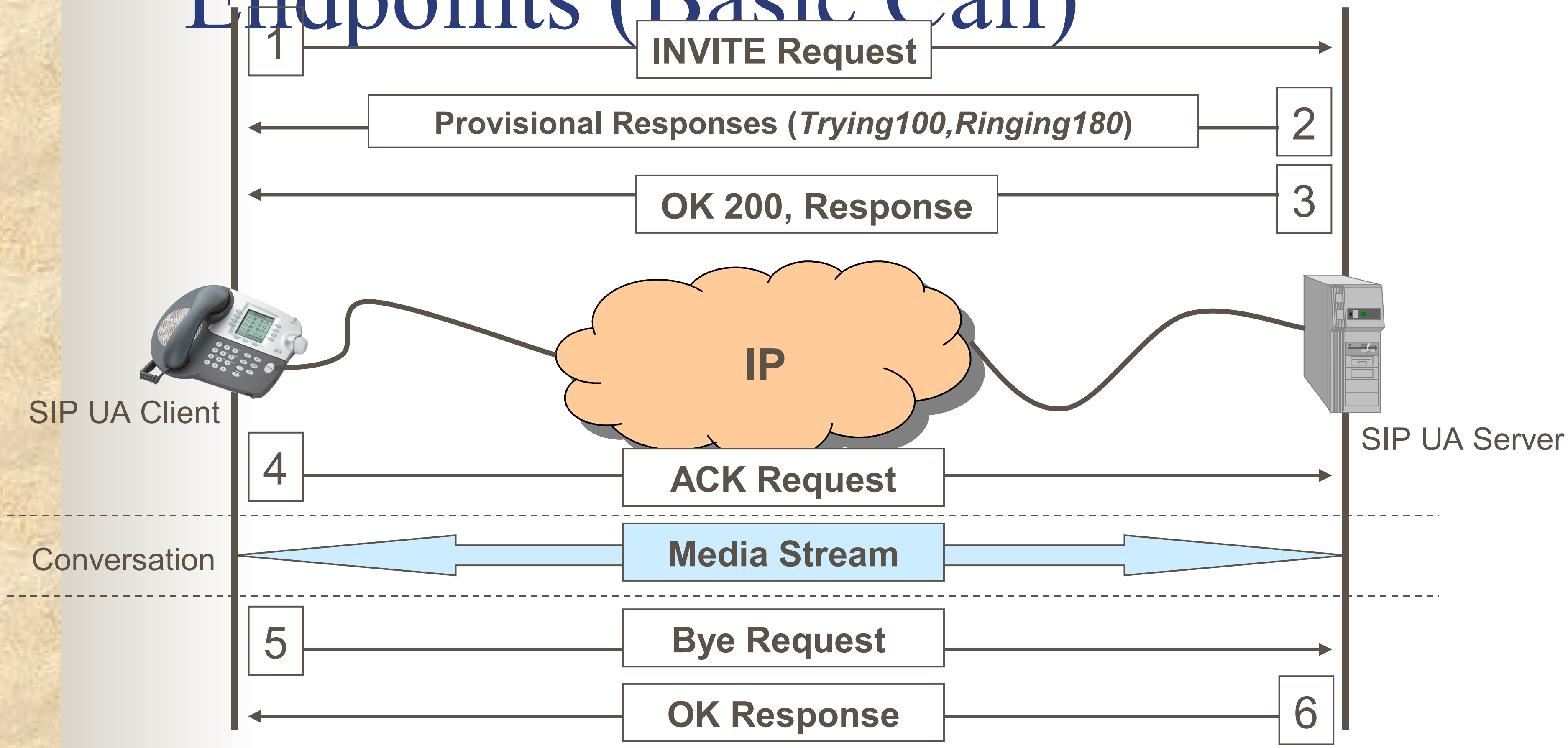
SDP example for Internet Telephony

```
v=0
o=g.bell  877283459  877283519  IN  IP4  132.152.1.19
c=IN  IP4 132.151.1.19
b=CT:64
t=3086272736
m=audio  3456  RTP/AVP   96
a=rtpmap : 96   VDVI/8000/1
m=video    3458    RTP/AVP    31
m=application   32416    udp   wb
a=orient:portrait
```

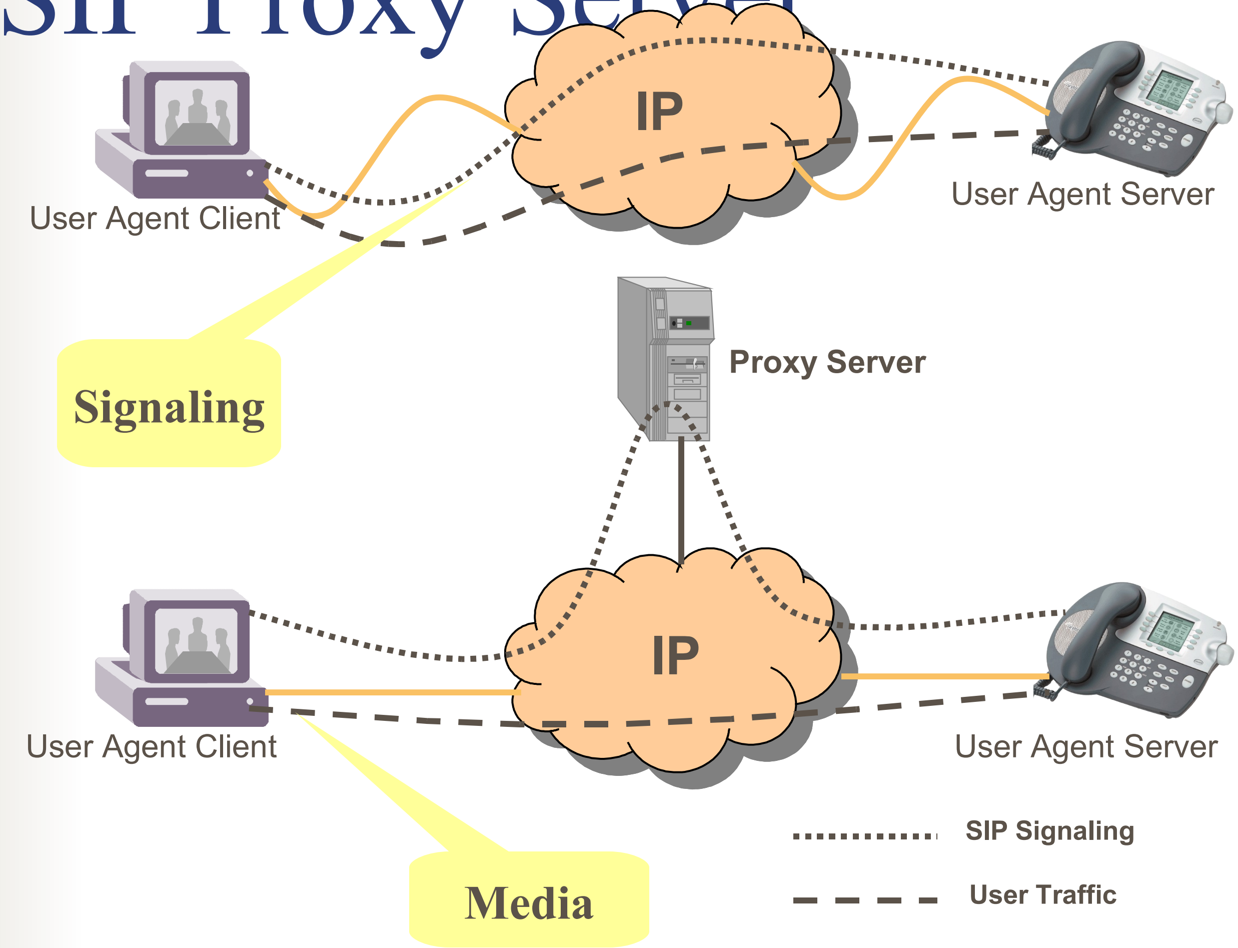
Locating a server

- The client must determine the IP address and port no of the server for which the request is destined
- If the port no is not listed in the SIP URL, the default port is 5060
- If the protocol is not listed the client must first attempt using UDP and then TCP
- The client queries the Domain Name System (DNS) for the host IP address

Direct Communication Between Endpoints (Basic Call)



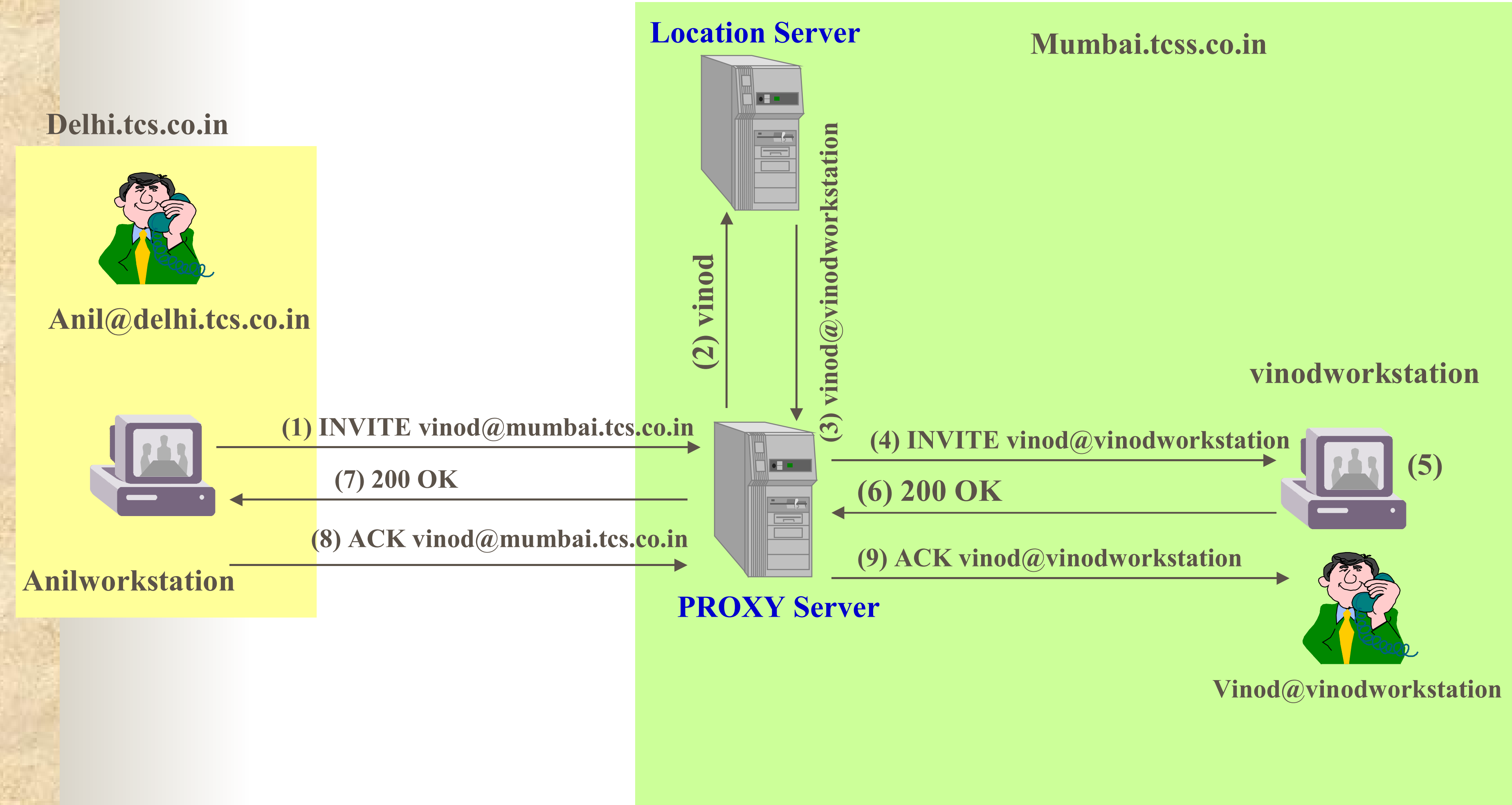
SIP Proxy Server



SIP Proxy Server

- The proxy server accepts the INVITE request from the client
- The proxy server determines the location by using the supplied address and location services
- An INVITE request is issued to the address of the location returned
- The called party user agent alerts the user and returns a success indication to the requesting proxy server
- An OK(200) response is sent from the proxy server to the calling party
- The calling party confirms receipt by issuing ACK request, which is forwarded by the proxy or sent directly to the called party

SIP Proxy Server



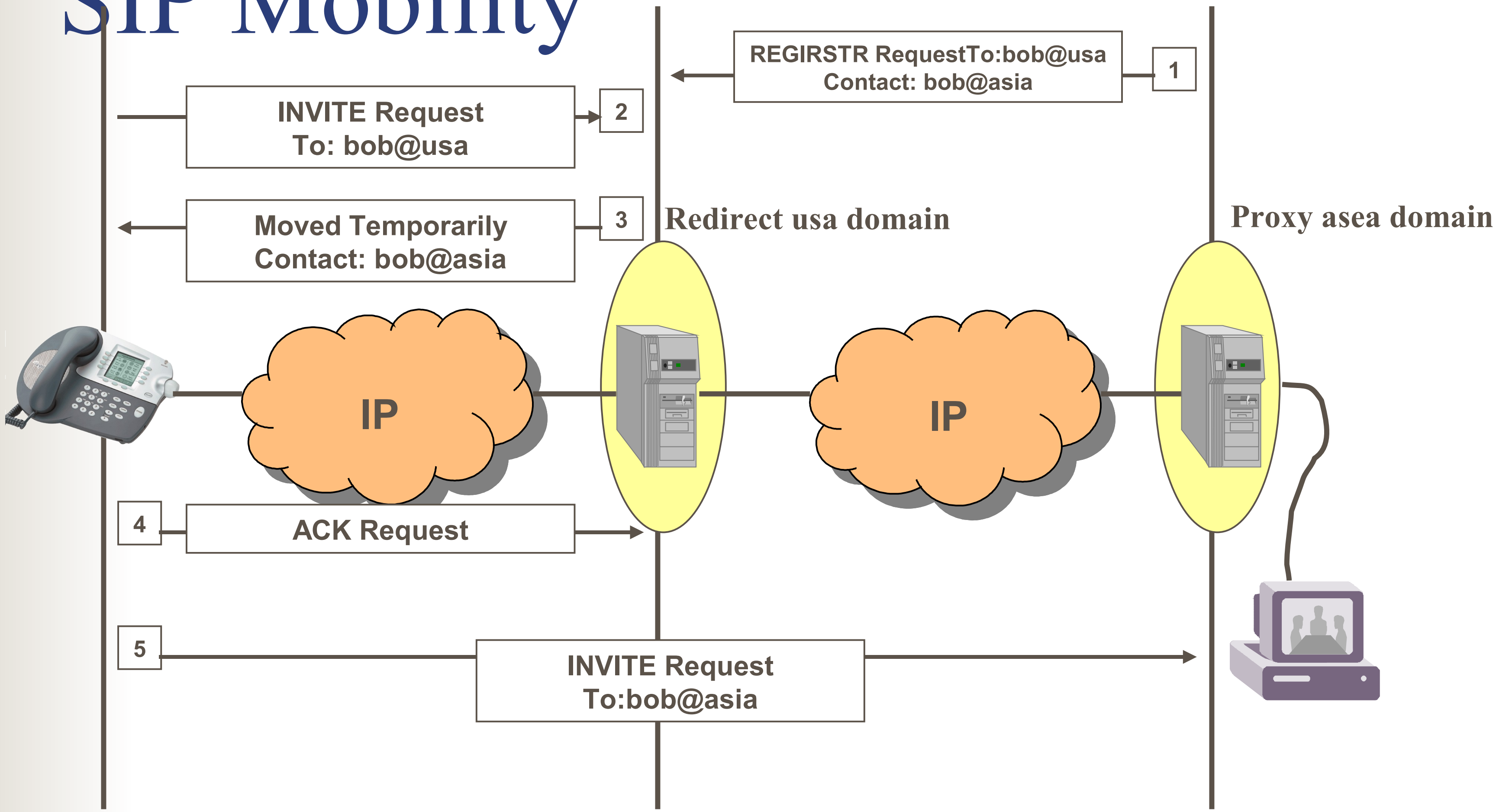
SIP Mobility

- SIP provides full mobility to the user. User can move from one service provider to another service provider and can retain the same address. The caller may always use the same address, phone number or URL, but will be redirected transparently to the network, location or device of choice of the called party. User can move temporarily or permanently to another network.
- SIP provides local number portability LNP

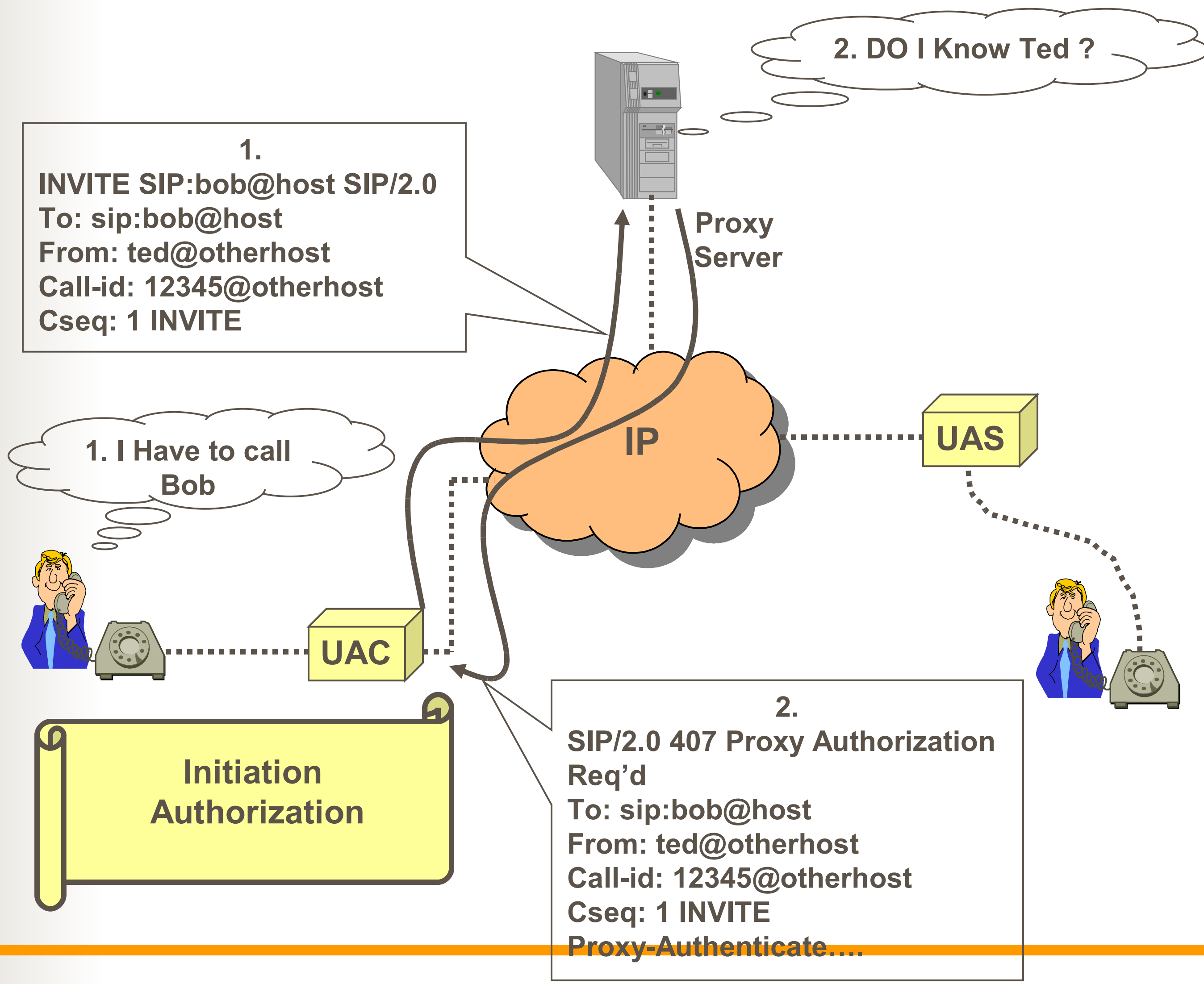
SIP Mobility

- Mobility in an IP environment is classified as :
  - Personal mobility – different terminals, same personal identity (address)
  - Terminal mobility – the ability to maintain communications when moving a single end system from one subnet to another
  - Service mobility – keep same services while mobile

SIP Mobility

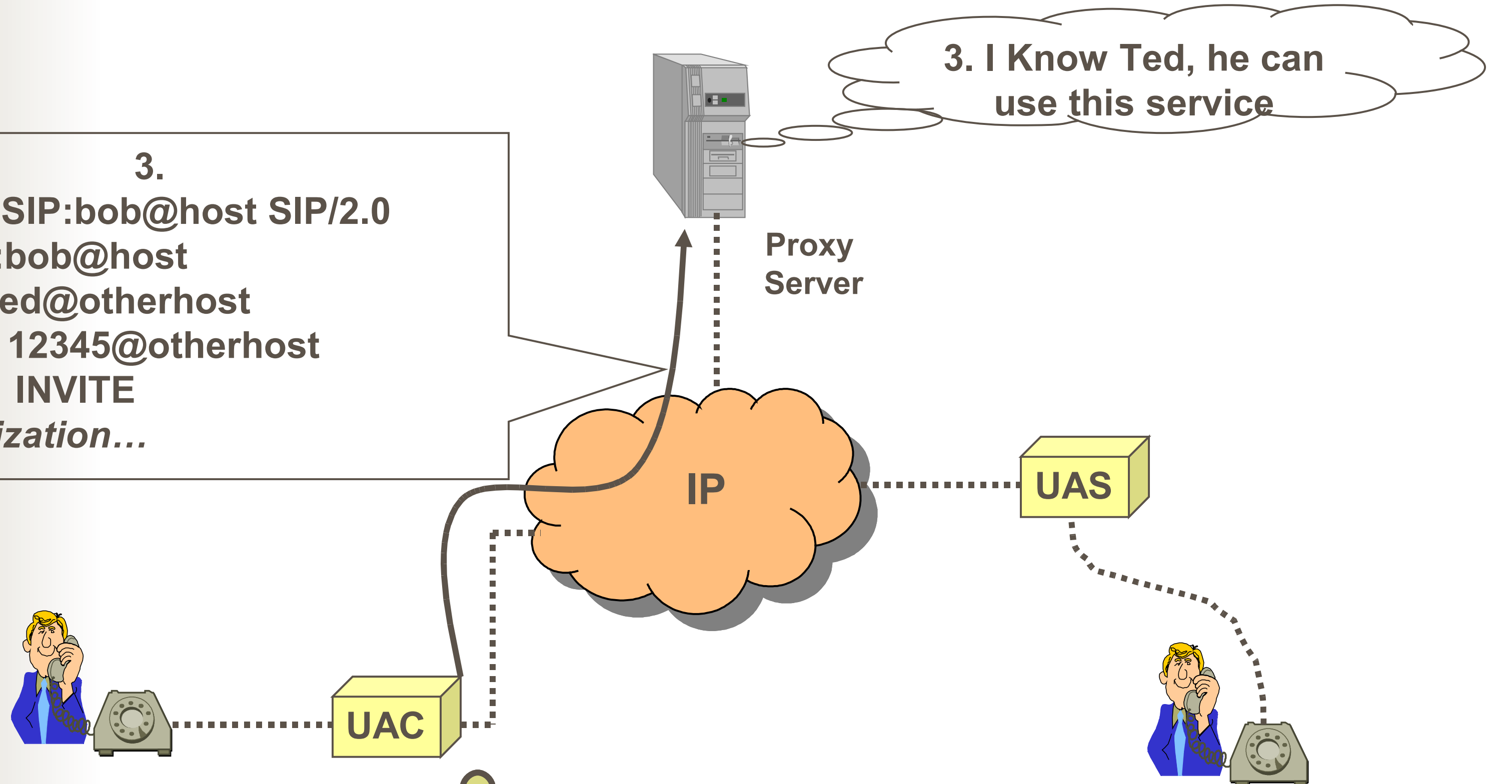


Call Setup Through Proxy Server



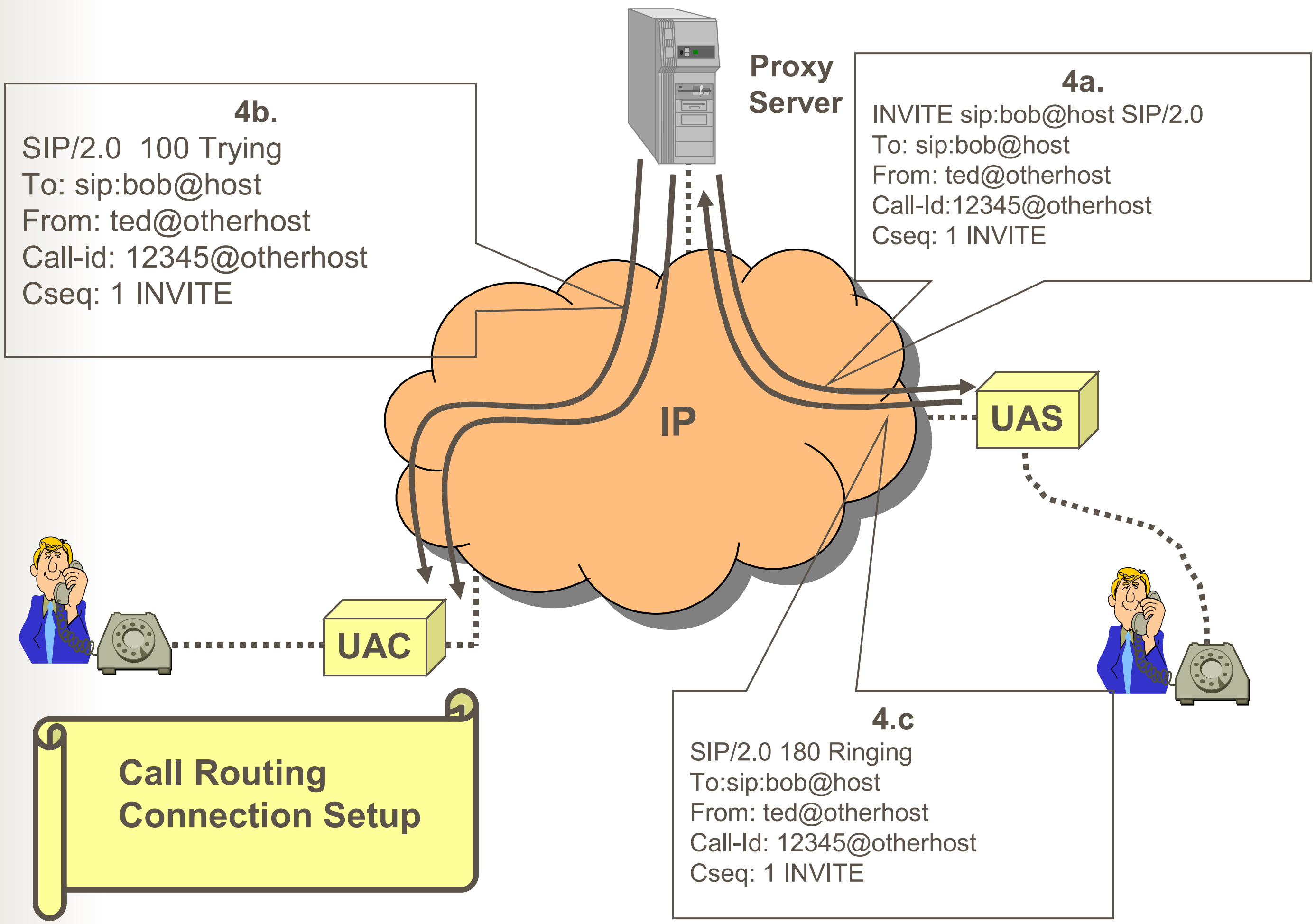
Call Setup Through Proxy Server

3.
INVITE SIP:bob@host SIP/2.0
To: sip:bob@host
From: ted@otherhost
Call-id: 12345@otherhost
Cseq: 1 INVITE
Authorization…



Authorization
Digit Collection
Digit Translation

Call Setup Through Proxy Server



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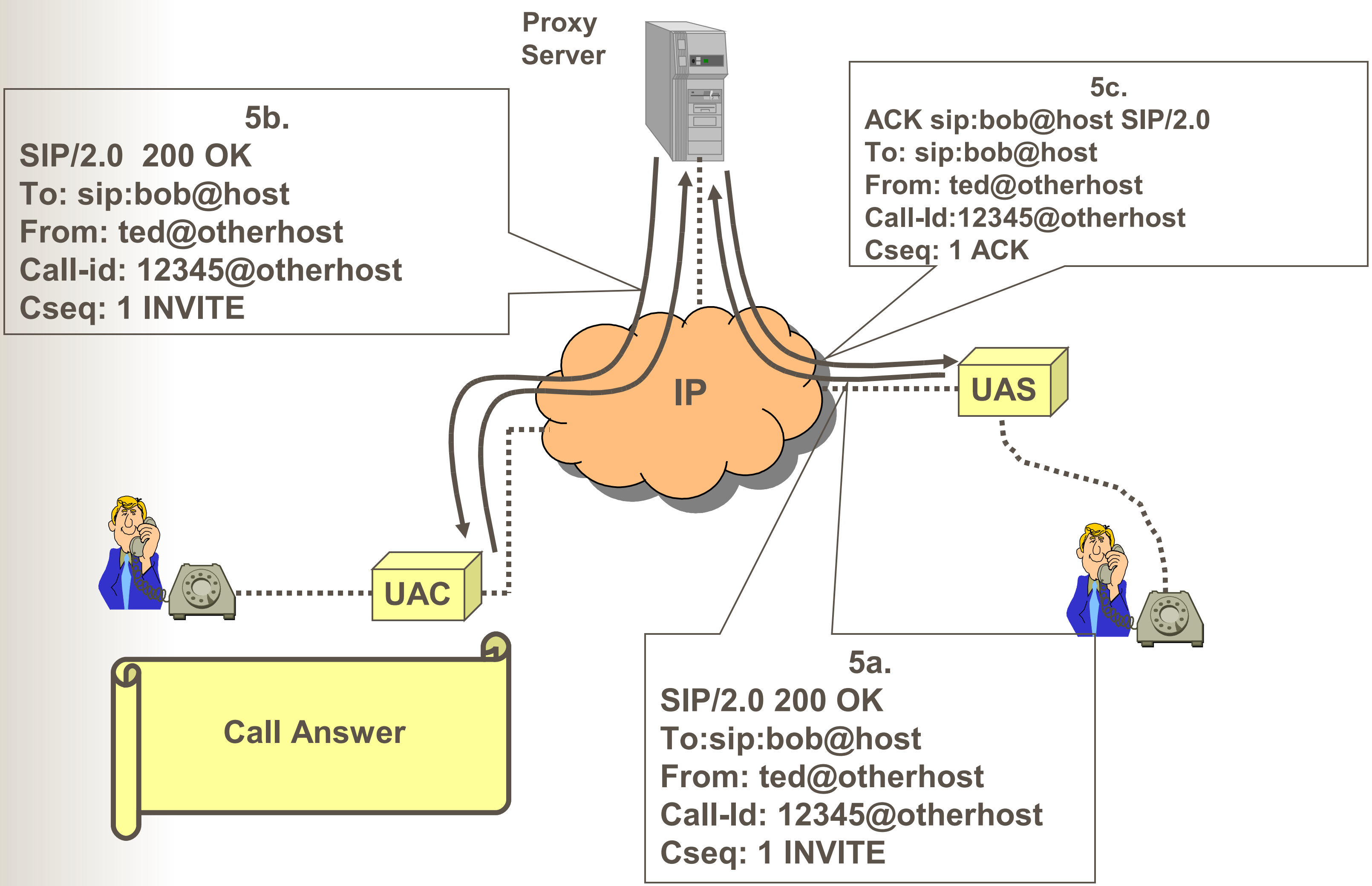
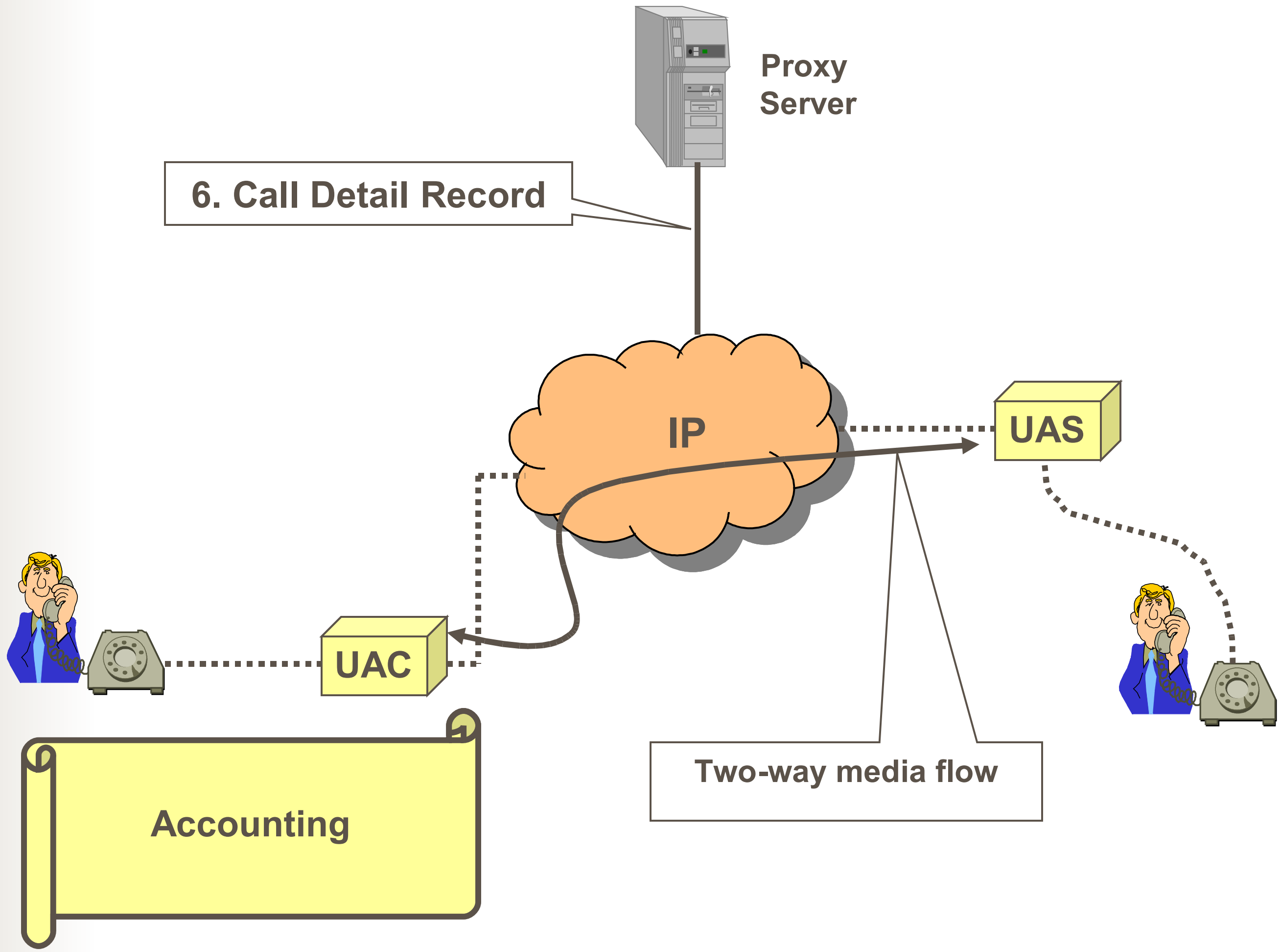
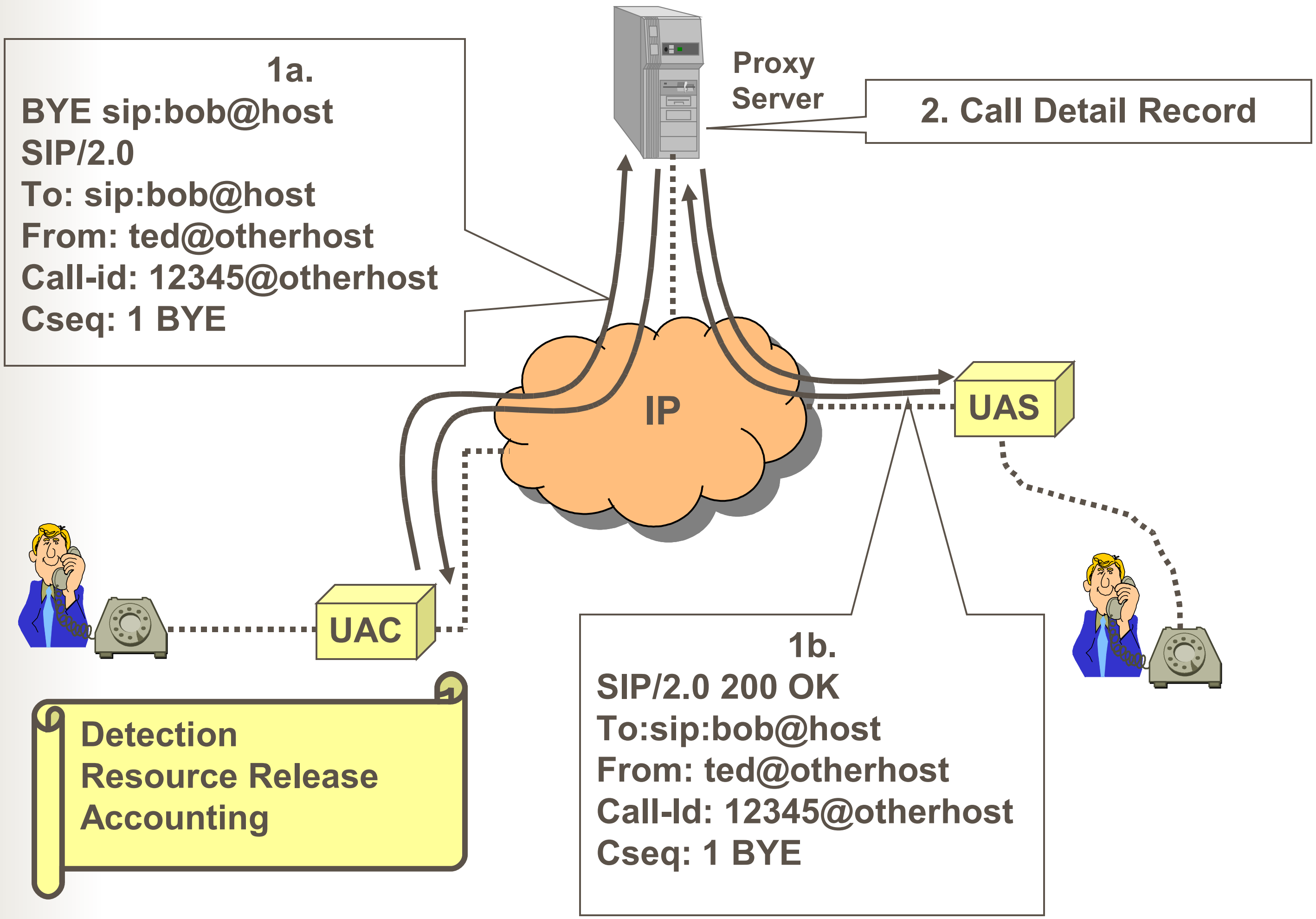


Call Setup Through Proxy Server



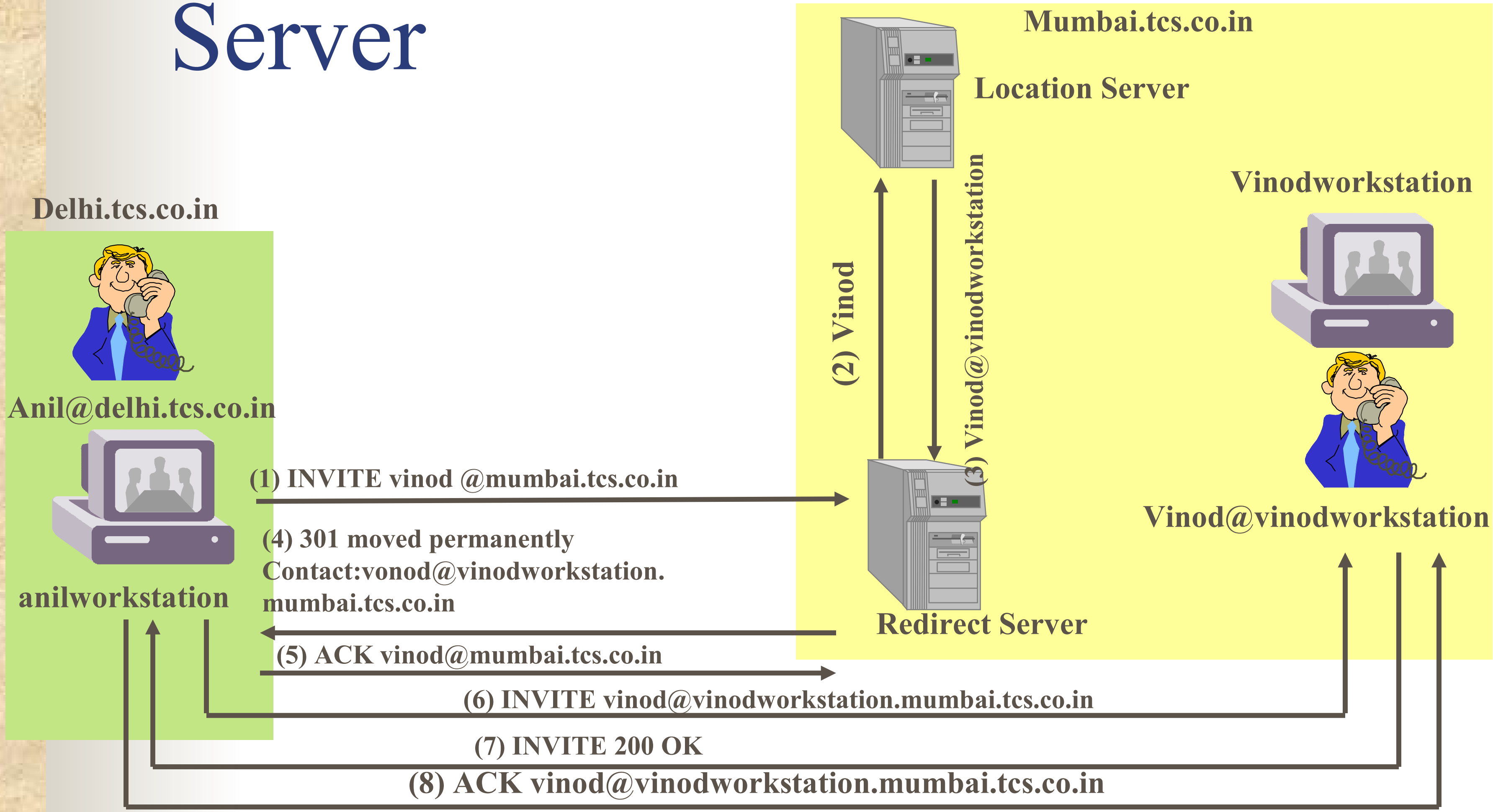
Call Release



Call Setup Through Redirect Server

- Redirection is an alternative to relaying. Redirection makes the caller's machine do some additional work, so it makes life easier for the proxy server.
- In redirection the request of called party is not relayed to other proxy servers, instead caller is provided with the address of the called party and the caller now can directly send the signal to the called party

Call Setup Through Redirect Server



TDM call with SS7 signaling

SIMPLE, Tested and Easy

Multimedia session with SIP signaling

Multiple Services, Multiple protocols and Multiple Media

Problems with SIP

- Processing text messages puts a higher load on gateways. The router must translate that text into a language that the router can understand.
- SIP is a fairly new protocol, so fewer people understand it than the older protocols.
- SIP features are still being developed, and many vendors have proprietary implementations of the protocol.

Thanks