SIP Tutorial

Presentation Of

Session Initiation Protocol

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Tutorial Overview

- SIP history and architecture
- SIP functionality
- SIP and Voice over IP
- Service creation over SIP
- SIP Security
- Present and Future of SIP

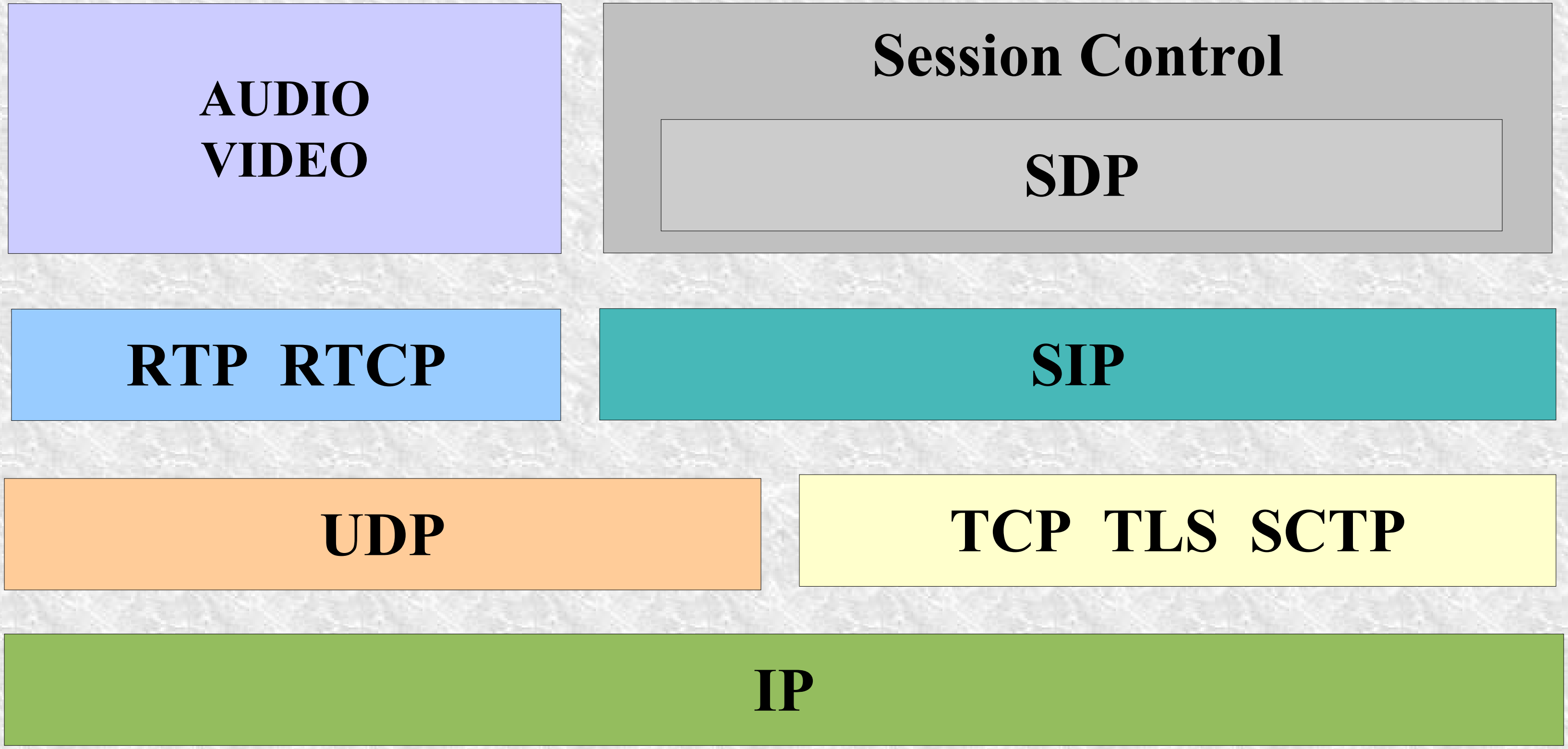
SIP History

- 1980s – first packet multimedia experiments
- 1992 – first IETF audiocast
- 1996 – first SIP related IETF drafts
  - Session Invitation Protocol
  - Simple Conference Invitation Protocol
  - MMUSIC IETF WG
- 1999 – RFC 2543
- 2002 – RFC 3261
- Today - over 30 IETF RFCs related to SIP, many Internet Drafts and Working Groups

SIP Purpose

- Application-layer signaling protocol
- Easy to understand
- Creation, modification and termination of multimedia communication sessions
- Negotiation of session's parameters
- Re-negotiation during communication session
- User mobility
- Ability to allow supplementary services
- Extensibility

SIP And VoIP Architecture



VoIP Protocols

- **transport protocols**
  - TCP – Transmission Control Protocol
  - UDP – User Datagram Protocol
  - SCTP – Stream Control Transmission Protocol
  - TLS – Transport Layer Security Protocol
- **media transport and control protocols**
  - RTP – Real-time Transport Protocol (RFC1889)
  - RTCP – Real-time Control Protocol (RFC3605)
  - SRTP – Secure Real-time Transport Protocol (RFC3711)
- **signaling protocol**
  - SIP – Session Initiation Protocol (RFC3261)
- **session negotiation**
  - SDP – Session Description Protocol (RFC2327)

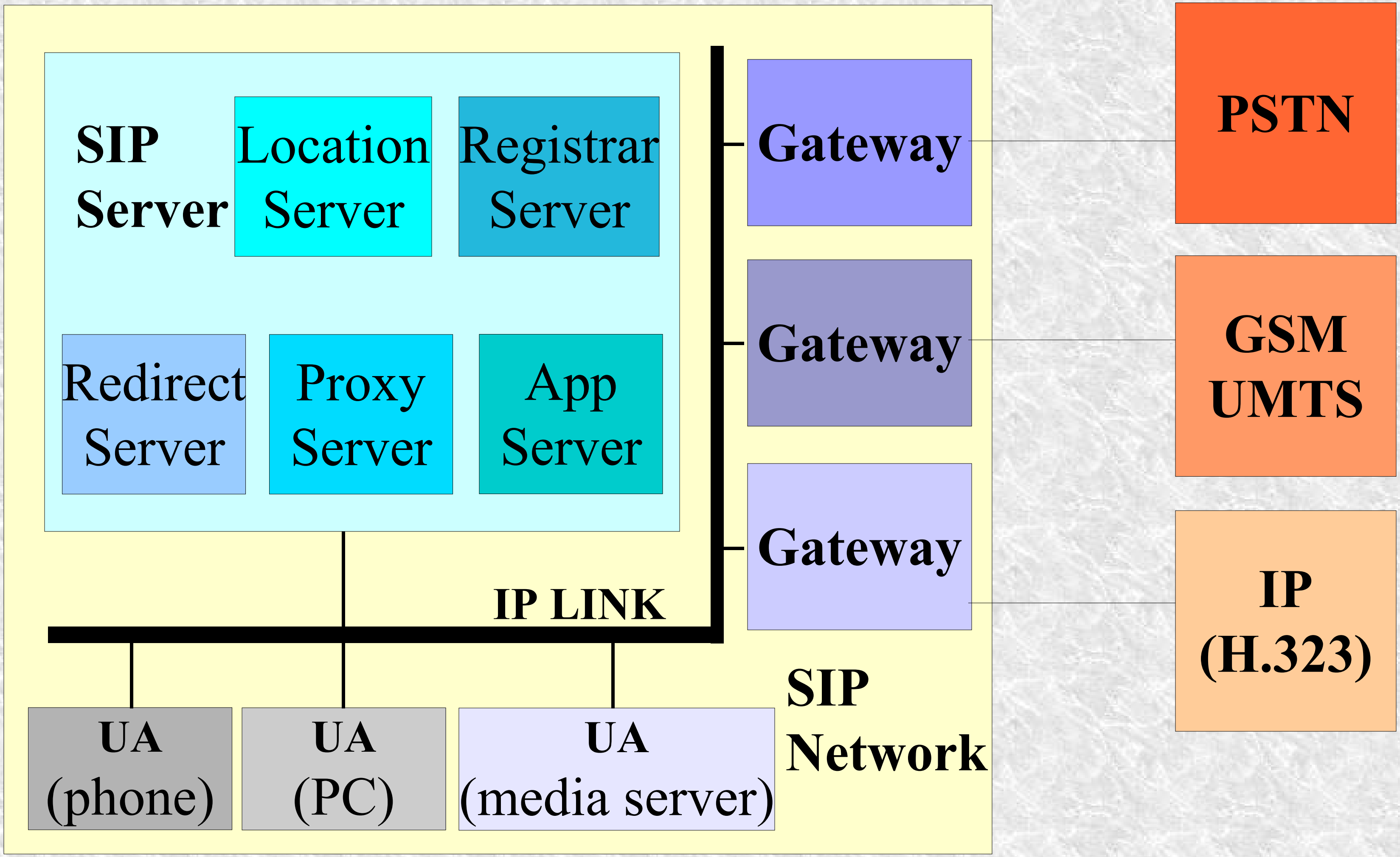
SIP Design

- Text based protocol in a format similar to HTTP
- Client-server communication
- Transaction oriented: request-response sequences
- Independent of transport layer protocol
- Request meaning is specified by method type
- Session capability negotiation
- Allow alpha-numeric addresses in URI format (email-like address) as well as E.164 numbers
- Use of domain names to locate servers
- PSTN number translation via ENUM

SIP Entities

- User Agent (UA)
  - User Agent Client (UAC) – initiates a SIP request
  - User Agent Server (UAS) – handles and eventually sends a response to a request
- Proxy server – routing of SIP requests
- Registrar server – registration of user's contact addresses
- Location server – providing of user location details
- Redirect server – return callee's addresses to caller
- Application server – providing advanced services for users

SIP Deployment Architecture



SIP Request Syntax

Start line

INVITE sip:user@sipserver.com SIP/2.0

Message headers

Via: SIP/2.0/UDP 10.10.10.10:5060
From: "Me" <sip:me@sipserver.org>;tag=a0
To: "User" <sip:user@sipserver.org>
Call-ID: d@10.10.10.10
CSeq: 1 INVITE
Contact: <sip:10.10.10.10:5060>
User-Agent: SIPTelephone
Content-Type: application/sdp
Content-Length: 251

Message body

v=0
o=audio1 0 0 IN IP4 10.10.10.10
s=session
c=IN IP4 10.10.10.10
m=audio 54742 RTP/AVP 4 3
a=rtpmap:4 G723/8000
a=rtpmap:3 GSM/8000

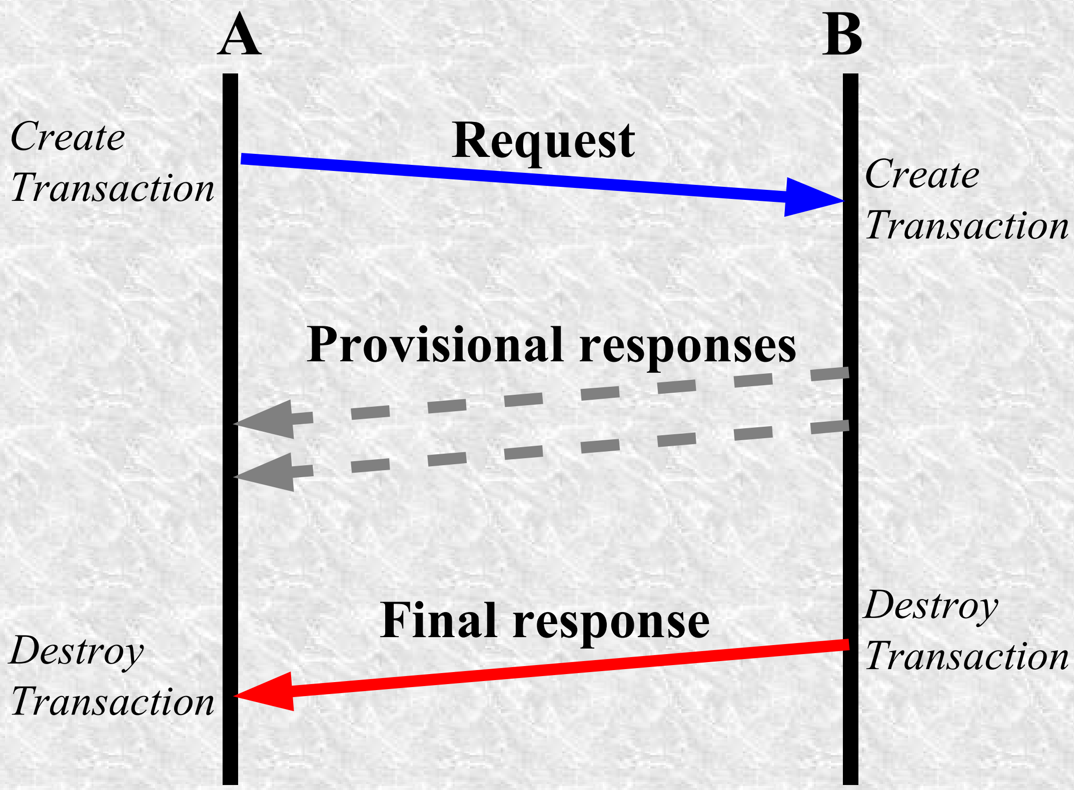
SIP Reply Syntax

Start line	**SIP/2.0 200 OK**
Message headers	**Via: SIP/2.0/UDP 10.10.10.10:5060** **From: "Me" <sip:me@sipserver.org>;tag=a0** **To: "User" <sip:user@sipserver.org>;tag=b0** **Call-ID: d@10.10.10.10** **CSeq: 1 INVITE** **Contact: <sip:10.10.10.20:5060>** **User-Agent: SIPSoftPhone** **Content-Type: application/sdp** **Content-Length: 123**
Message body	**v=0** **o=audio2 0 0 IN IP4 10.10.10.20** **s=session** **c=IN IP4 10.10.10.20** **m=audio 62043 RTP/AVP 0 4**

SIP Transaction

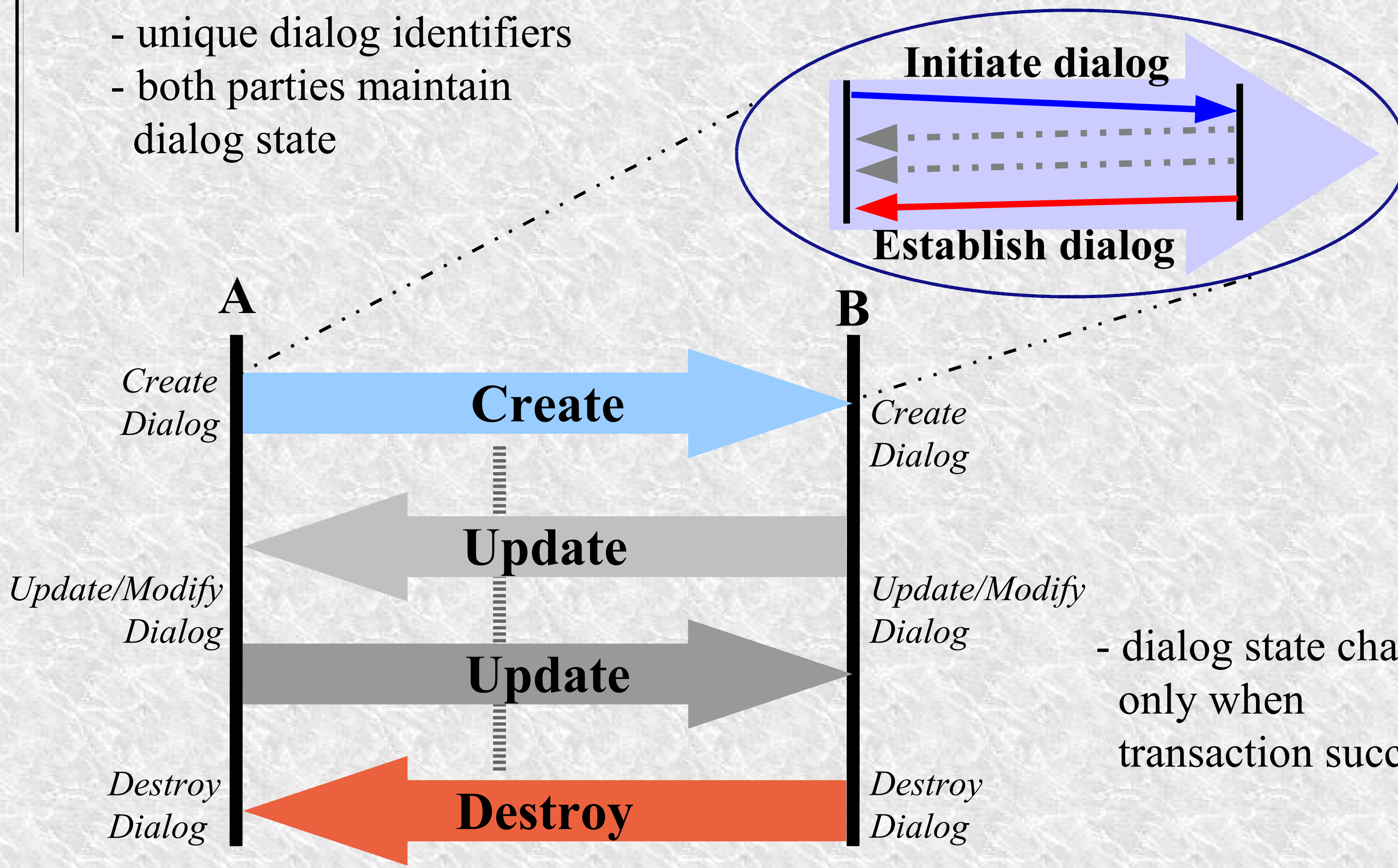
Request methods:
- INVITE
- CANCEL
- ACK
- BYE
- REGISTER
- ...

Response classes:
- provisional
  - Status code 100-199
- final
  - Successful (200-299)
  - Redirection (300-399)
  - Request failure (400-499)
  - Server failure (500-599)
  - Global failures (600-699)



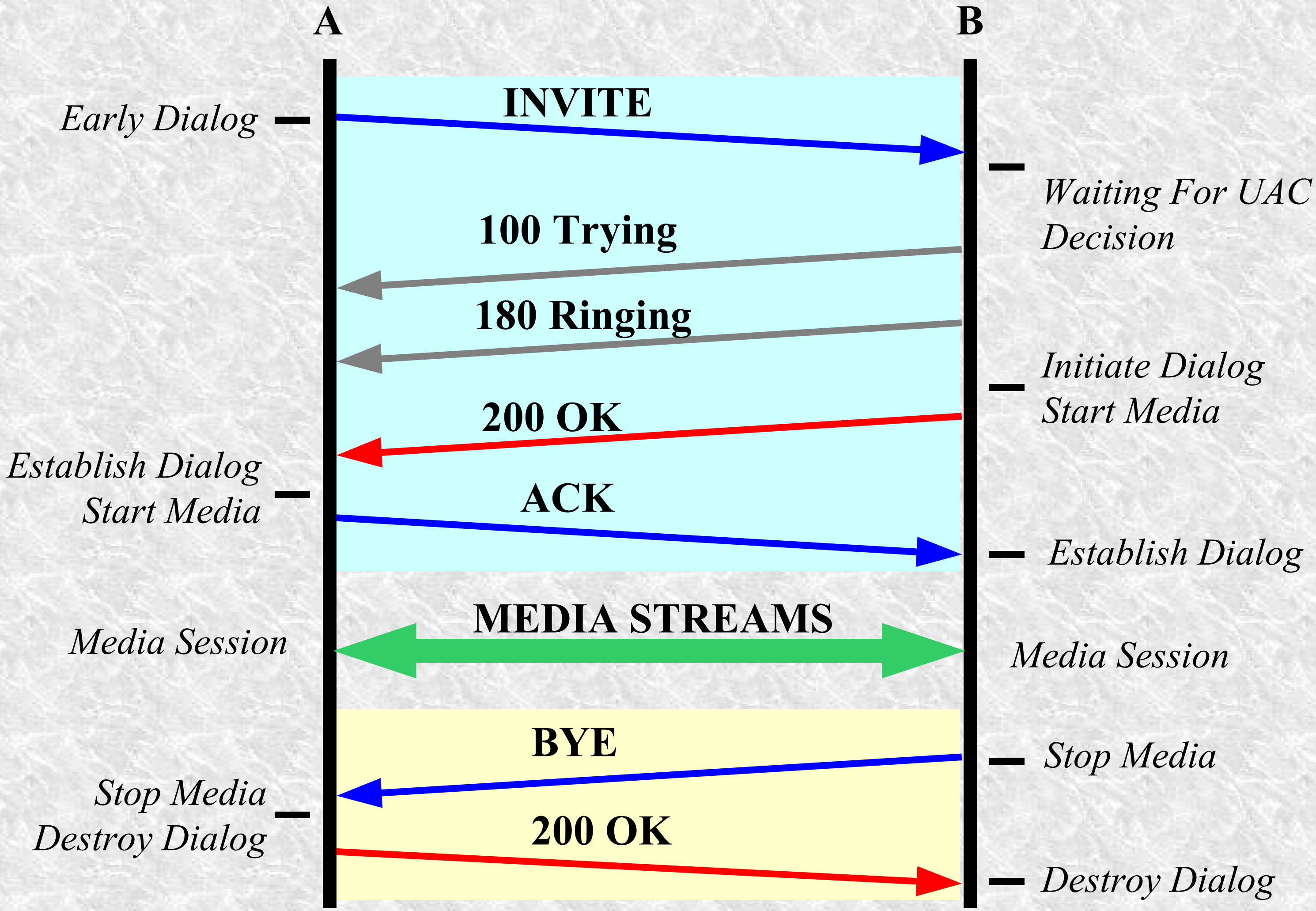
SIP Dialog

- unique dialog identifiers
- both parties maintain dialog state



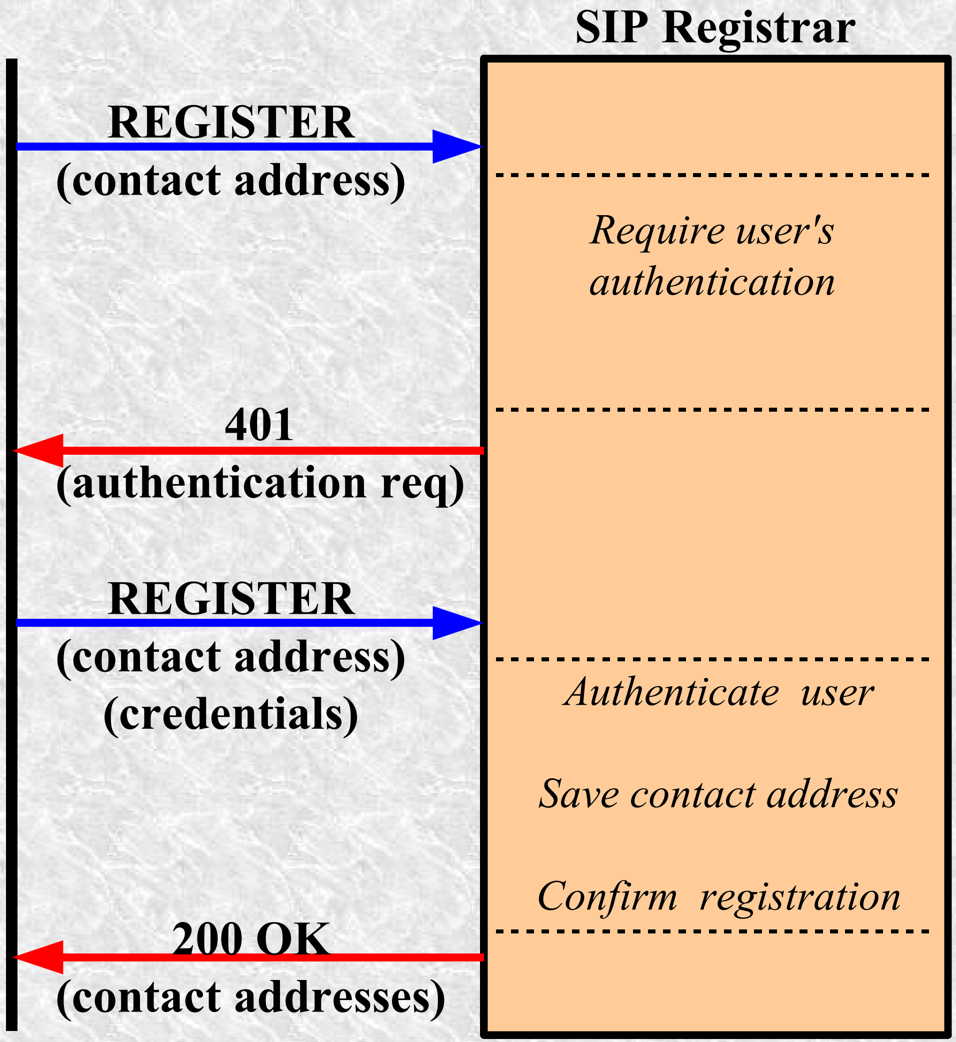
- dialog state changed only when transaction succeed

SIP Media Session

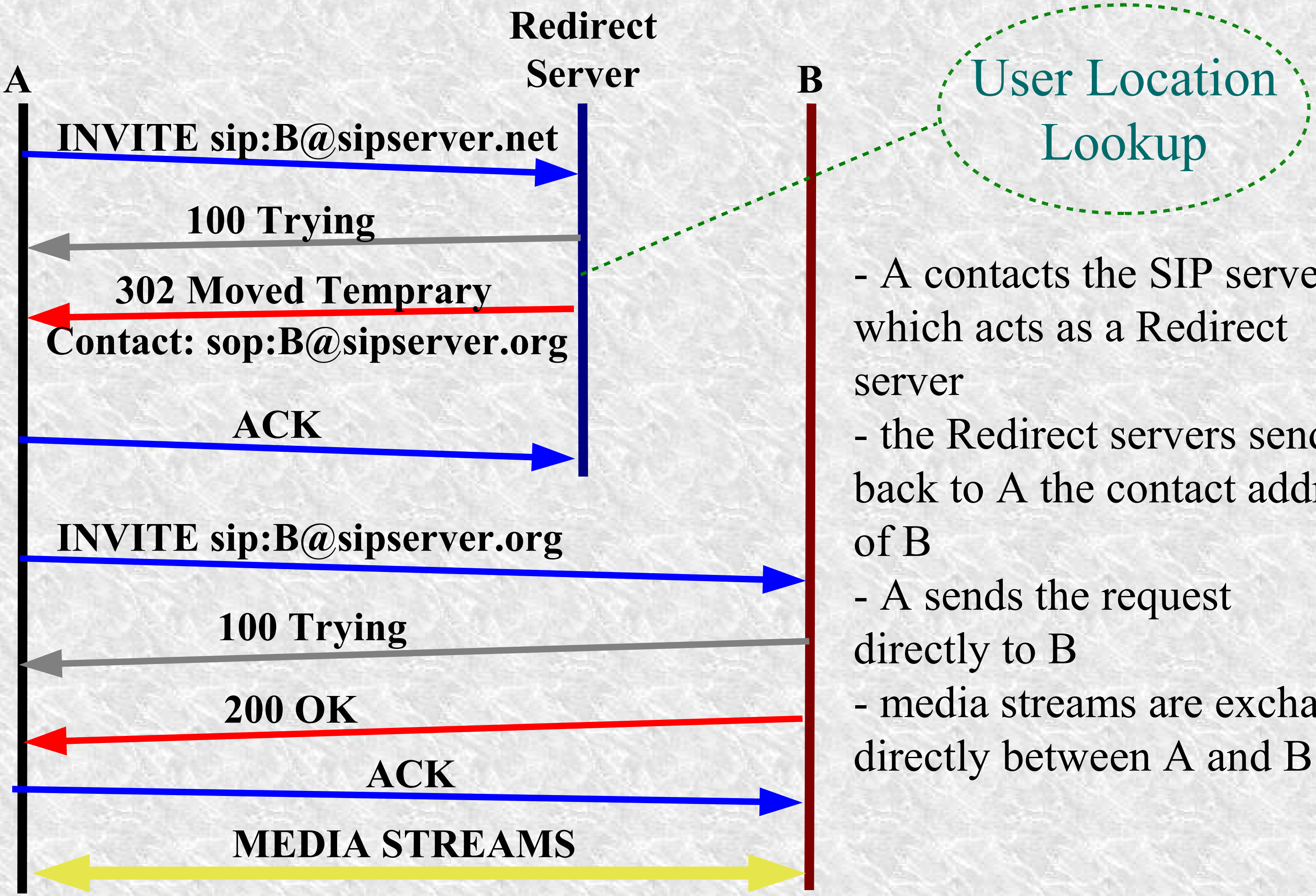


SIP Registration

- registration via REGISTER request
- third party registration support
- multiple contact addresses
- refreshing contact address
- user authentication is recommended
- Registrar server works together with Location server



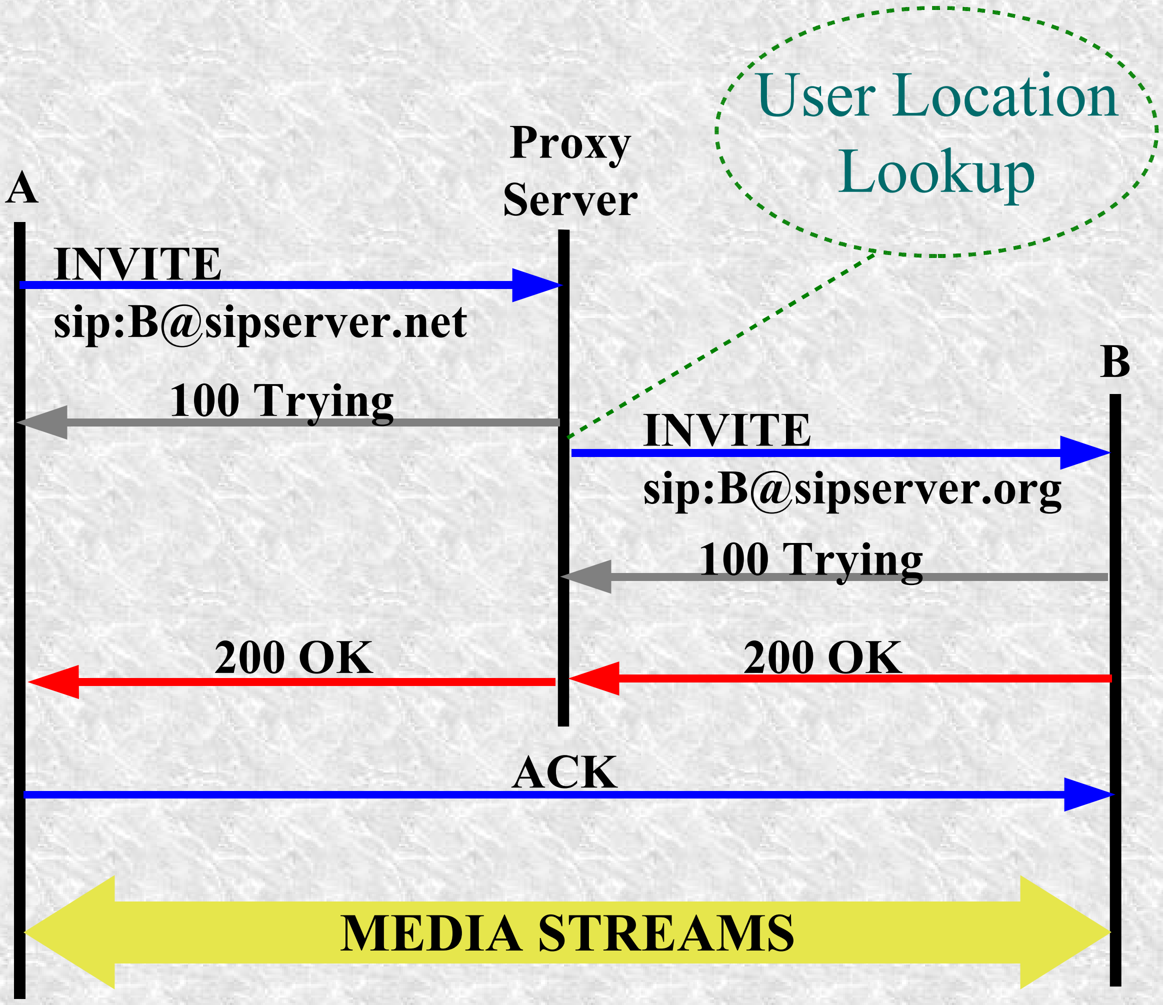
SIP Redirection



- A contacts the SIP server which acts as a Redirect server
- the Redirect servers sends back to A the contact address of B
- A sends the request directly to B
- media streams are exchanged directly between A and B

SIP Proxying

- A contacts the SIP server which acts as a Proxy server
- the Proxy servers sends the INVITE to the contact address of B
- A sends the ACK request directly to B
- media streams are exchanged directly between A and B



SIP And NAT Traversal

- important issue in the early stage of VoIP
- SIP clients cannot go alone through symmetric NATs (very common: Linux/iptables)
- client-side solution
  - STUN - Simple Traversal of User Datagram Protocol (UDP) Through Network Address Translators (NATs) (RFC3489)
- server-side solution
  - TURN – Traversal Using Relay NAT (IETF draft)
- combined solution
  - ICE – Interactive Connectivity Establishment

SIP Security

- ensure privacy, service protection, proper accounting and billing
- HTTP digest authentication schema
  - challenge-response architecture
  - basic authentication deprecated
- TLS for SIP entity identification and traffic encryption
- IPSeq secure channels between SIP servers
- S/MIME extensions for end-to-end encryption

Services on top of SIP

- voice conversations
  - advanced call features: call redirect, call forwarding, call baring, black/white lists
  - easy to manage and use caller's preferences and callee's capabilities
  - parallel and serial search of users
- audio conferences, video and instant messaging sessions, gaming
- presence and service location
- system provisioning
- extensible and programmable environment

Present And Future Of SIP

- Deployed all over the world
  - Europe: Deutsche Telekom, BT, Tiscali, Arcor, SipGate, Telio, Voztelecom ...
  - USA: Earthlink, AOL, FreeWorldDialup, SipPhone, Vonage ...
- Replacement for H.323 and adopted as signaling protocol in 3GPP
- Continuous extension development within IETF
- Widest used protocol by newest ITSP
- Devices and applications from most famous providers: CISCO, Avaya, Microsoft and very good representation in Open Source world

References

- SIP: RFC3261 - http://www.ietf.org/rfc/rfc3261.txt
- SIP: RFC3263 - http://www.ietf.org/rfc/rfc3263.txt
- SDP: RFC2337 - http://www.ietf.org/rfc/rfc2327.txt
- RTP: RFC1889 - http://www.ietf.org/rfc/rfc1889.txt
- RTCP: RFC3605 - http://www.ietf.org/rfc/rfc3605.txt
- STUN: RFC3489 - http://www.ietf.org/rfc/rfc3489.txt
- IETF - http://www.ietf.org
- 3GPP - http://www.3gpp.org

- Further details:
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