



Session Initiation Protocol (SIP)



Introduction

- A powerful alternative to H.323
- More flexible, simpler
- Easier to implement
 - Advanced features
- Better suited to the support of intelligent user devices
- A part of IETF multimedia data and control architecture
 - SDP, RTSP (Real-Time Streaming Protocol), SAP (Session Announcement Protocol)

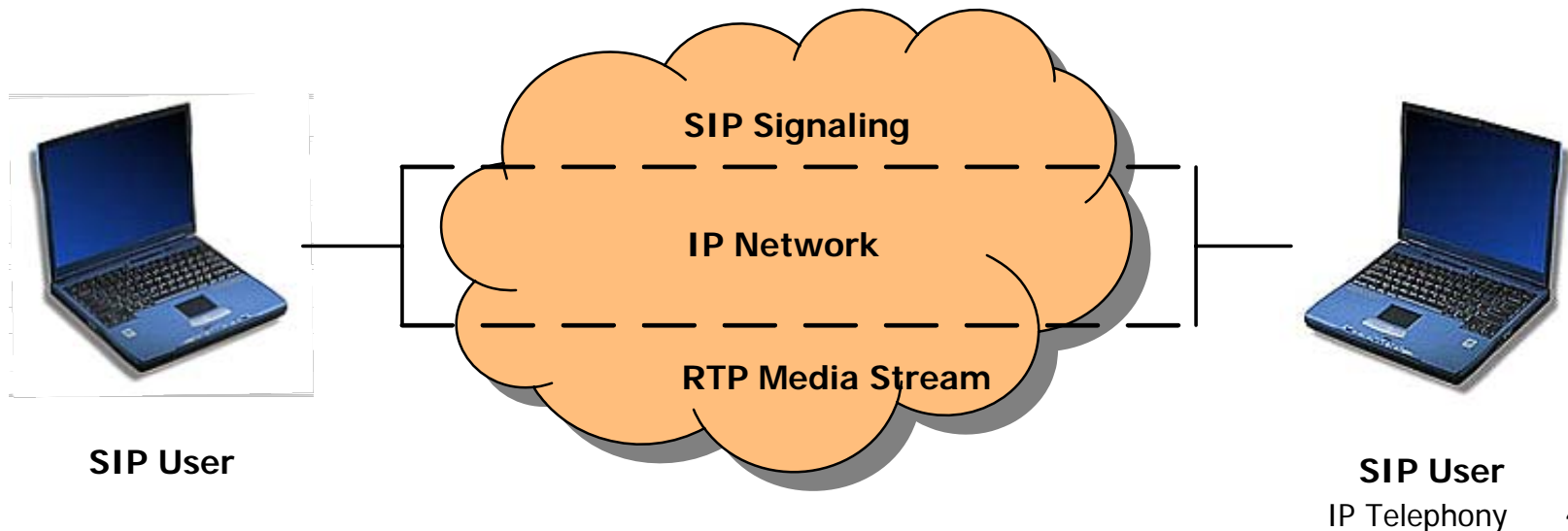


Popularity of SIP

- Originally Developed in the MMUSIC (Multiparty Multimedia Session Control)
 - A separate SIP working group
 - RFC 2543
 - Many developers
 - The latest version: RFC 3261 (June 2002)
- SIP + MGCP/MEGACO
 - The VoIP signaling in the future
- “bake-offs” or SIP Interoperability Tests
 - The development of SIP and its implementation by system developers has involved a number of events.
 - Various vendors come together and test their products against each other
 - to ensure that they have implemented the specification correctly
 - to ensure compatibility with other implementations

SIP Architecture

- A signaling protocol
 - The setup, modification, and tear-down of multimedia sessions
- SIP + SDP
 - Describe the session characteristics to potential session participants
- Separate signaling and media streams
 - Signaling may pass via one or more proxy or redirect servers
 - Media stream takes a more direct path.





SIP Network Entities [1/4]

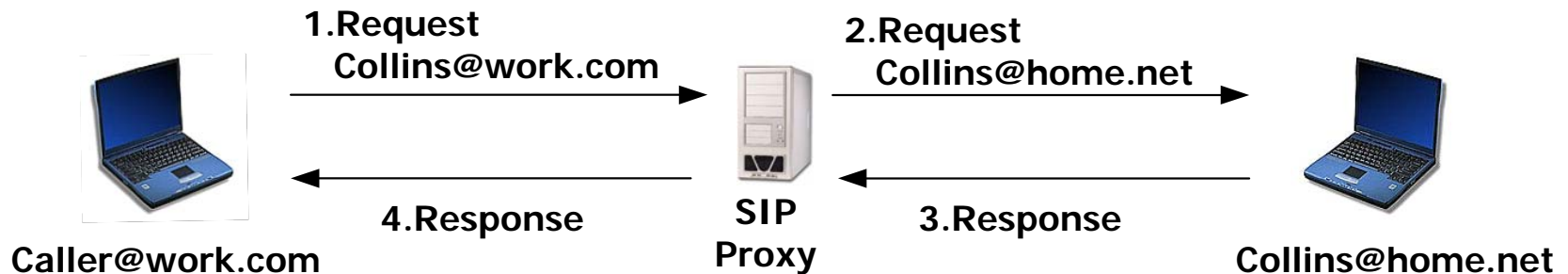
- Clients
 - User agent clients
 - Application programs sending SIP requests
- Servers
 - Responds to clients' requests
- Clients and servers may be in the same platform.
 - Proxy acts as both clients and servers

SIP Network Entities [2/4]

- Four types of servers

- Proxy servers

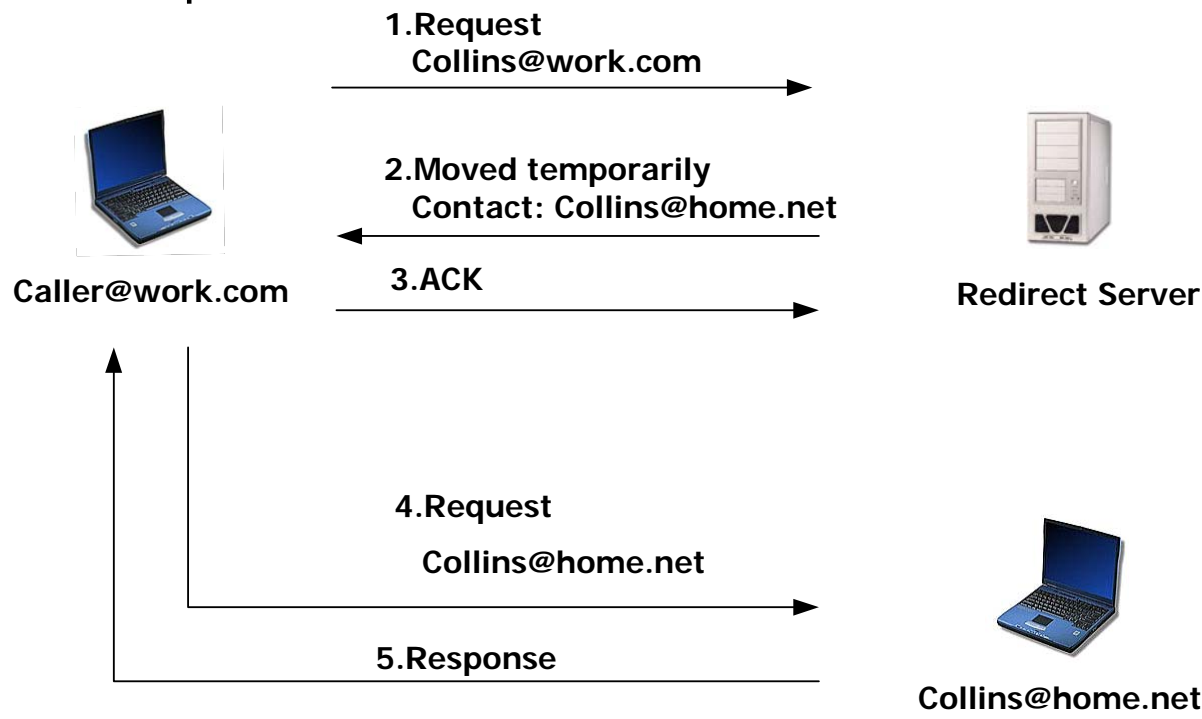
- Act in a similar way to a proxy server used for web access
 - Handle requests or forward requests to other servers after some translation
 - Can be used for call forwarding, time-of-day routing, or follow-me services



SIP Network Entities [3/4]

■ Redirect servers

- Accept SIP requests
- Map the destination address to zero or more new addresses
- Return the new address(es) to the originator of the request



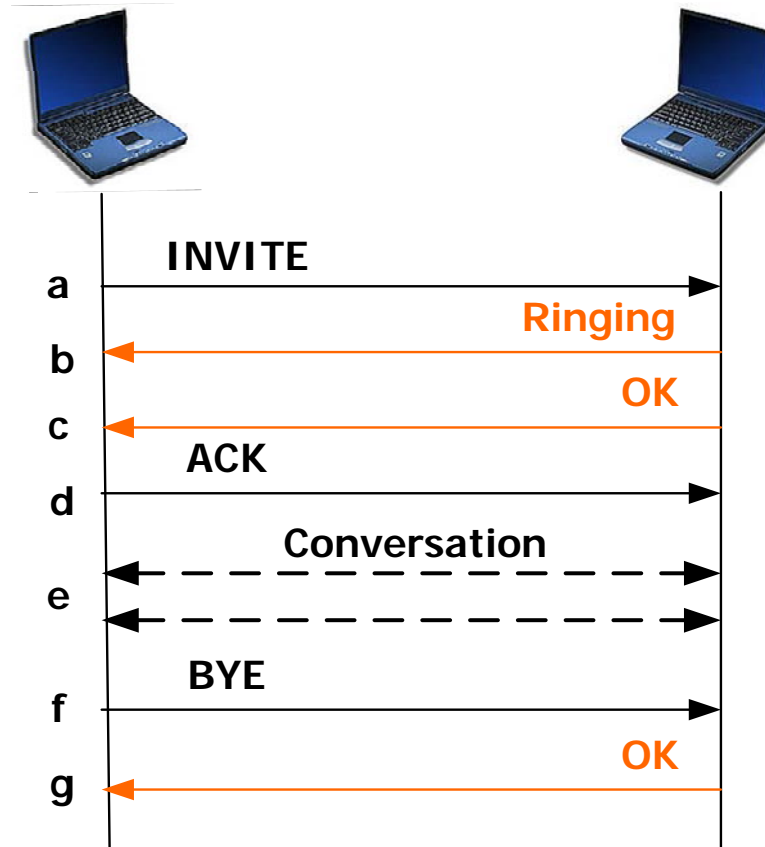


SIP Network Entities [4/4]

- A user agent server
 - Accepts SIP requests and contacts the user
 - The user responds → an SIP response
 - A SIP device
 - E.g., a SIP-enabled telephone
- A registrar (location server)
 - Accepts SIP REGISTER requests
 - Indicating that the user is at a particular address
 - Personal mobility
 - Typically combined with a proxy or redirect server

SIP Call Establishment

- SIP call establishment is simple.
- A number of interim responses may be made to the INVITE prior to the called party accepting the call.

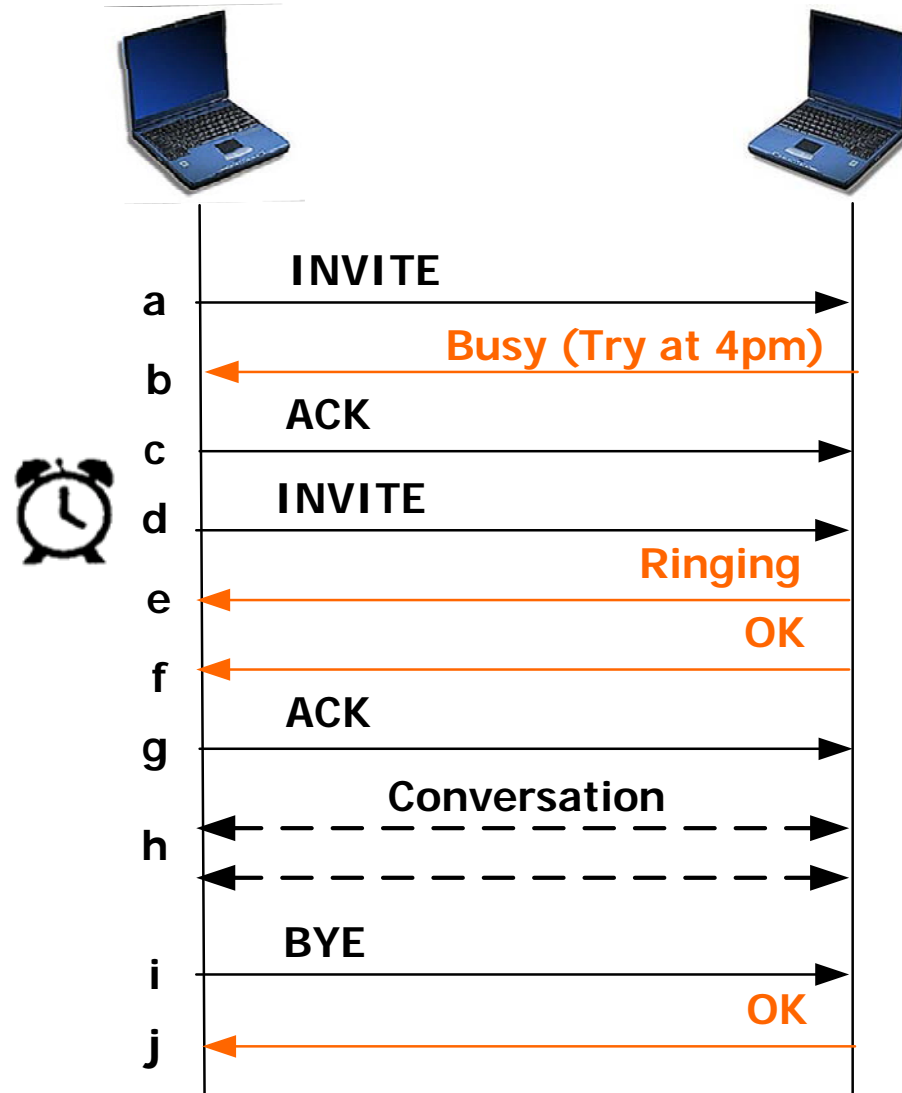




SIP Advantages

- Attempt to keep the signaling as simple as possible
- Offer a great deal of flexibility
 - Does not care what type of media is to be exchanged during a session or the type of transport to be used for the media
- Various pieces of information can be included within the messages
 - Easily include non-standard information
 - Text-based encoding
 - Enable the users to make intelligent decisions
 - The control of the intelligent features is placed in the hands of the customer, not the network operator.
 - E.g., SUBJECT header

Call Completion to Busy Subscriber Service



- Text-based
 - Similar to HTTP
 - Disadvantage – more bandwidth consumption
- SIP messages
 - message = start-line
 - *message-header CRLF
 - [message-body]
 - start-line = request-line | status-line
- Request-line specifies the type of request
- The response line indicates the success or failure of a given request.



■ Message headers

- Additional information of the request or response
- E.g.,
 - The originator and recipient
 - Retry-after header
 - Subject header

■ Message body

- Describe the type of session
- The most common structure for the message body is SDP (Session Description Protocol).
- Could include an ISDN User Part message
- Examined only at the two ends



SIP Requests [1/2]

- **Method** SP **Request-URI** SP **SIP-version** CRLF
- Request-URI
 - The SIP address of the destination
- Methods
 - INVITE, ACK, OPTIONS, BYE, CANCEL, REGISTER
 - INVITE
 - Initiate a session
 - Information of the calling and called parties
 - The type of media
 - ~IAM (initial address message) of ISUP
 - ACK only when receiving the final response



SIP Requests [2/2]

- BYE
 - Terminate a session
 - Can be issued by either the calling or called party
- OPTIONS
 - Query a server as to its capabilities
 - To support a particular type of media
- CANCEL
 - Terminate a pending request
 - Pending Request: an INVITE did not receive a final response
- REGISTER
 - Log in and register the address with a SIP server
 - “all SIP servers” – multicast address (224.0.1.175)
 - Can register with multiple servers
 - Can have several registrations with one server

“One Number” Service

User at Address 2



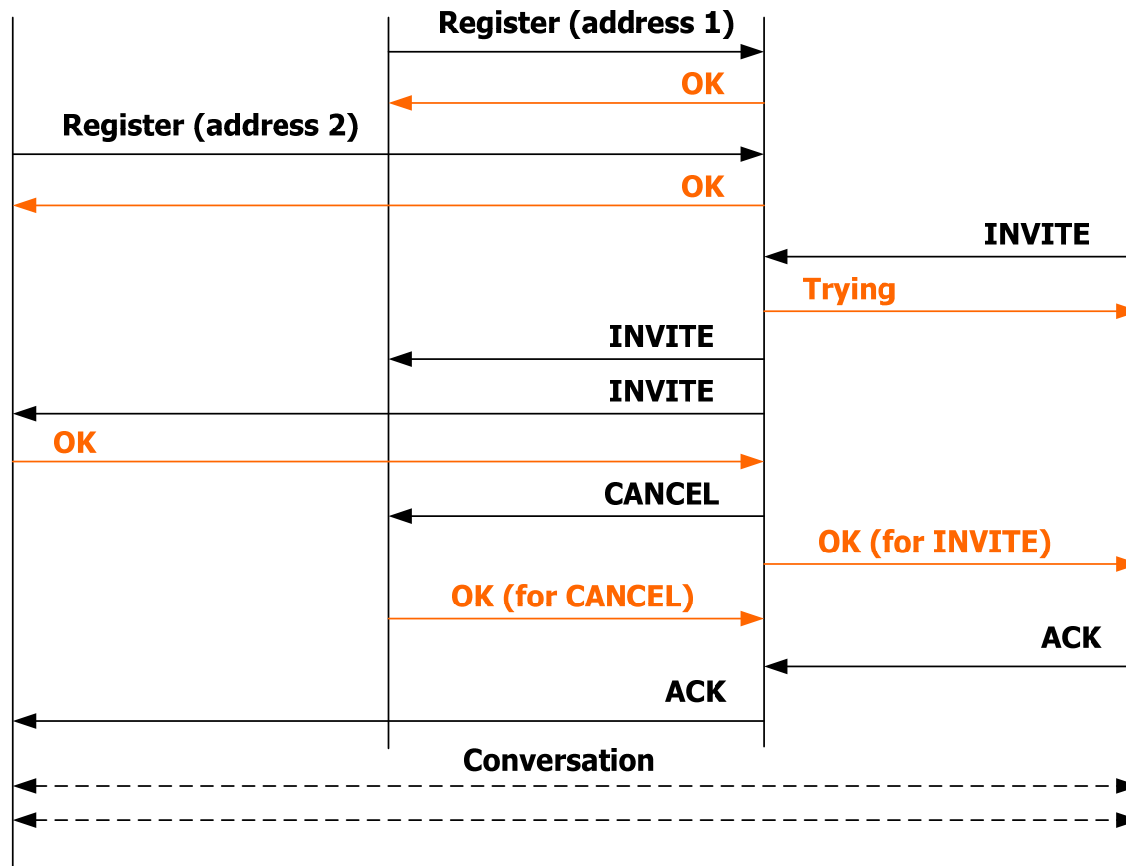
User at Address 1



Registrar/Proxy



Caller





SIP INFO Method

- Specified in RFC 2976
 - For transferring information during an ongoing session
- The transfer of DTMF digits
- The transfer of account balance information
 - Pre-paid service
- The transfer of mid-call signaling information



SIP Responses

- SIP Version SP Status Code SP Reason-Phrase CRLF
- Reason-Phrase
 - A textual description of the outcome
 - Could be presented to the user
- Status code
 - A three-digit number
 - 1XX Informational
 - 2XX Success (only 200 is defined: the request has been understood and has been performed)
 - 3XX Redirection (302: the called party is not available at the address used in the request, and the request should be reissued to a new address included with the response)
 - 4XX Request Failure (401: the client is not authorized to make the request)
 - 5XX Server Failure (505: the server does not support the SIP version specified in the request)
 - 6XX Global Failure (600: busy)
 - All responses, except for 1XX, are considered final
 - Should be ACKed if the original message happened to be an INVITE



SIP Addressing

- SIP URIs (Uniform Resource Indicators)
 - user@host
 - sip:collins@home.net
 - sip:3344556789@telco.net



Message Headers

- Provide further information about the message
- E.g.,
 - To:header in an INVITE
 - The called party
 - From:header
 - The calling party
- Four main categories
 - General, Request, Response, and Entity headers



General Headers

- Used in both requests and responses
- Basic information
 - E.g., To:, From:, Call-ID: (uniquely identifies a specific invitation to a session), ...
- Contact:
 - Provides a URL for use in future communication regarding a particular session
 - **Examples 1:** In a SIP INVITE, the Contact header might be different from the From header.
 - An third-party administrator initiates a multiparty session.
 - **Example 2:** Used in response, it is useful for directing further requests directly to the called user.
 - **Example 3:** It is used to indicate a more appropriate address if an INVITE issued to a given URI failed to reach the user.



■ Request Headers

- Apply only to SIP requests
- Addition information about the request or the client
- E.g.,
 - Subject:
 - Priority: urgency of the request (emergency, urgent, normal, or non-urgent)

■ Response Headers

- Further information about the response that cannot be included in the status line
- E.g.,
 - Unsupported
 - Retry-After



Entity Headers

- Indicate the type and format of information included in the message body
- Content-Length: the length of the message body
- Content-Type: the media type of the message body
 - E.g., application/sdp
- Content-Encoding: for message compression
- Content Disposition: how a message part should be interpreted
 - session, alert, render ...