Outline

- Multimedia is different
- Real Time Protocol (RTP)
- Session Description Protocol (SDP)
- Session Initiation Protocol (SIP)

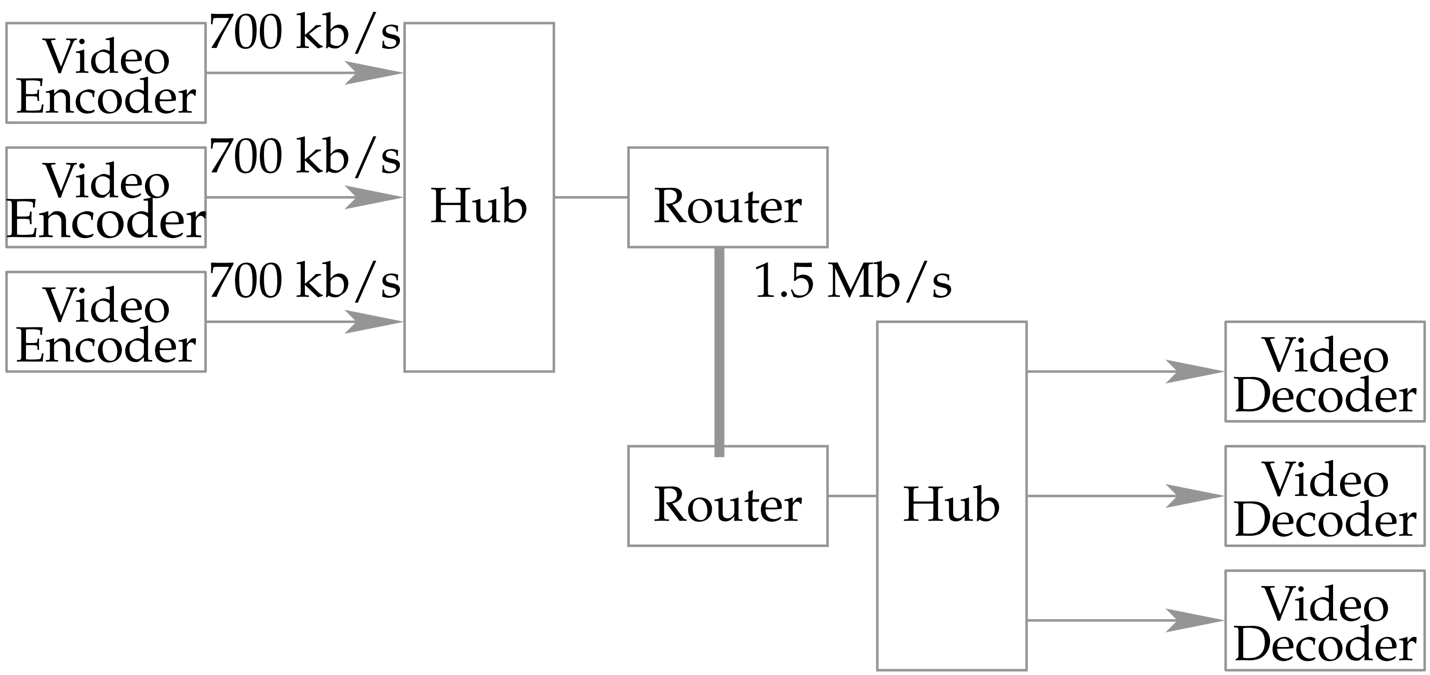
Elastic vs. Inelastic Workloads

- **Some applications adapt to network performance**
  - Examples: file transfer, printing.
  - Will perform the same task regardless of bandwidth
  - Such workloads are *elastic*—they adapt fine to lower performance
- **Other apps have traffic performance requirements:**
  - Example: video requires a certain available bandwidth
  - Workloads that fail if requirements not met are *inelastic*

Why is this an issue?

- **Most data networks provide *best effort delivery***
  - Try their best to deliver your data, but no guarantees
- **Traditional data applications:**
  - Use protocols such as TCP to deal with data loss and unknown network available bandwidth,
  - Tend to generate data in bursts, and
  - Normally do not require any kinds of guarantees from the network
- **Elastic workloads will often swamp inelastic ones**
  - BitTorrent can make Skype sound terrible
- **Traditional networks have not been designed with *Quality Of Service* (QoS) in mind**

Where do things break?



- **Things break when there is a contention for resources**
- **Service may be unacceptable for all**

Example of a Network with QoS: The Phone Network

- **Before you can communicate**
  - Must negotiate a channel with the network
  - If network lacks the resources, your call is not completed
- **If call is completed, you “own” circuit**
  - Yours for the duration of the call
  - Regardless of the additional traffic in the network
- **Circuit has appropriate capacity for voice**

QoS Components

- **Must know traffic characteristics and requirements**
  - Must describe them to the network to reserve bandwidth
- **Need *signaling protocol* betw. nodes & net. mgmt**
- **The Network must implement Admission Control**
  - Reject connections exceeding available bandwidth
- **The Network may also implement Traffic Policing**
  - Make sure traffic emitted by the nodes conform to the agreed parameters

Where to Provide?

- **1990s and early 2000s saw a lot of work on getting QoS into layer 3: Resource ReSerVation Protocol (RSVP)**
  - Fine for walled garden networks, e.g., IP-telephony backbones
  - Consumer applications do not want to assume QoS support
- **There's a lot more to multimedia than simple QoS**
  - Session establishment, media negotiation
- **Higher layers need to be involved**
  - Transport: Real Time Protocol (RTP)
  - Session: Session Description Protocol (SDP)
  - Application: Session Initiation Protocol (SIP)

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RTP [RFC 3550]

- **Provides end-to-end delivery services for data with real-time characteristics**
  - E.g. interactive audio and video
  - Services include: Source & payload type identification, Sequence numbering, Time-stamping, Delivery monitoring
- **Typically used on top of UDP**
  - Relies on UDP for multiplexing & checksums
- **Work over other network or transport protocols**
  - Lower-level must provide framing & length indication

RTP continued

- **Supports data transfer to multiple destinations**
  - Uses network-level multicast if available
- **Does *not* ensure timely delivery/QoS guarantees**
  - Relies on lower-layer services to do so
- **Does *not* guarantee in-order delivery**
- **Does *not* even guarantee delivery**
  - Underlying network need not be reliable
  - Underlying network need not deliver packets in sequence
  - RTP sequence numbers determine proper location of packet

Two parts to RTP

1. **RTP**
   - Carries data that has real time properties
2. **RTCP**
   - Monitors quality of service
   - Conveys information about participants in a session (for “loosely controlled” sessions)

RTP protocol framework

- **RTP is not a complete protocol**
  - It is a protocol *framework*, deliberately not complete
  - Tailored to applications through modification, not options
- **Each applications needs a *profile specification***
  - defines set of payload type codes
  - defines mapping of payload types to payload formats
- **Also need actual payload format specification**

RTP Session

- **Association among a set of participants communicating with RTP**
- **A session is defined by a particular pair of destination transport addresses**
  - One network address
  - A pair of ports (one for RTP and one for RTCP)
  - May be common to all participants (as in IP multicast)
  - May be different for each (individual network address + common port address)

Synchronization Source (SSRC)

- **Source of a stream of RTP packets**
  - Randomly chosen 32-bit numeric identifier
  - Intended to be globally unique
  - Carried in RTP header (not dependent on network address)
- **Timing & sequence number space are per-SSRC**
  - Lets receiver separately handle packets from same source
  - RTCP sender/receiver reports are per SSRC
- **One participant may use multiple SSRCs**
  - Should use one for each stream
  - Different streams may have different media clock rates
  - Or one stream may switch encodings mid-stream
- **Binding of SSRCs is provided through RTCP**
  - E.g., if participant uses multiple cameras in one session

RTP Translators and Mixers

- **Intermediate act as gateways at RTP layer**
  - Allow RTP traffic to pass through firewalls
  - Mix and/or recode data to fit over a low bandwidth link
- ***Translators* leave original SSRC intact**
  - So sources distinct even when all packets from translator
  - Translator may change payload type or combine packets
- ***Mixers* combine streams from multiple sources**
  - Output requires new SSRC with new seq/timestamps
  - Original sources conveyed in each packet using "contributing sources" (CSRC) list

RTP Packet

- **Fixed RTP packet header**
- **List of contributing sources (possibly empty)**
- **RTP Payload**
  - E.g. audio samples, compressed video data, etc. . . .

RTP Data Header

```
 0                   1                   2                   3
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|V=2|P|X|  CC   |M|     PT      |       sequence number         |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                           timestamp                           |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|           synchronization source (SSRC) identifier            |
+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+
|            contributing source (CSRC) identifiers             |
|                             ....                              |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

RTP Data Header details

- **Padding (P, 1 bit)**
  - Packet ends w. padding octets ending in padding length
  - E.g., useful when encrypting with block cipher
- **Extension (X, 1 bit)**
  - If set, fixed header followed by header extension
- **CSRC Count (CC, 4 bits)**
- **Profile-specific Marker (M, 1 bit)**
  - E.g., might indicate frame boundary

RTP Data Header details continued

- **Payload Type (PT, 7 bits)**
  - RFC 3551 defines some default audio/video types
  - But actual interpretation depends on profile
  - Can even define some types dynamically
- **Sequence Number (16 bits)**
  - Initial value random
  - Increments for each RTP *packet* (not byte) sent
  - Used by receiver to detect packet loss & misordering
- **Time Stamp (32 bits)**
  - Sampling instant of start of payload
  - Resolution depends on data format, must be sufficient for synchronization & measuring jitter

RTP Control Protocol (RTCP)

- **Uses separate port from data**
  - Save monitoring tools from sorting through data packets
  - For UDP, use even port n for RTP, and $n + 1$ for RTCP
    (So good idea for NATs to preserve even/odd port parity)
- **Multiple RTCP messages sent in *compound packets***
  - Reduces packet rate, saves bandwidth & processing cost
  - Figure out # segments based on packet length
- **Compound packets start w. *reception report***
  - Losses in multicast distribution can be quickly isolated
  - Senders can adapt to current network conditions

RTCP Bandwidth Allocation

- **Application decides bandwidth needed for session**
  - Included in session announcement or inferred by scope
- **Control b/w should be fixed fraction of total**
  - RTP currently recommends 5% of total session bandwidth
  - Profile can specify some other fraction
- **RTCP b/w kept constant by varying report interval**
  - Track number of active senders and receivers
  - Senders get 25% of total RTCP b/w, receivers the rest
  - Minimum report interval 5 seconds to avoid bursts on small sessions
  - Actual interval randomized to avoid synchronization

RTCP Receiver Report (RR)

<table>
<tr><td>V=2</td><td>P</td><td>RC</td><td>PT=RR=201</td><td>LENGTH</td></tr>
<tr><td colspan="5">SSRC OF SENDER</td></tr>
<tr><td colspan="5">SSRC_1 (SSRC OF FIRST SOURCE)</td></tr>
<tr><td colspan="3">FRACTION LOST</td><td colspan="2">CUMULATIVE NUMBER OF PACKETS LOST</td></tr>
<tr><td colspan="5">EXTENDED HIGHEST SEQUENCE NUMBER RECEIVED</td></tr>
<tr><td colspan="5">INTERARRIVAL JITTER</td></tr>
<tr><td colspan="5">TIME OF LAST SR (LSR)</td></tr>
<tr><td colspan="5">DELAY SINCE LAST SR (DLSR)</td></tr>
<tr><td colspan="5">. . . ADDITIONAL RECEPTION REPORTS. . .</td></tr>
<tr><td colspan="5">. . . PROFILE-SPECIFIC EXTENSIONS. . .</td></tr>
</table>

RTCP Sender Report (SR)

V=2	P	RC	PT=SR=200	LENGTH
SSRC OF SENDER				
NTP TIMESTAMP, MOST SIGNIFICANT WORD				
NTP TIMESTAMP, LEAST SIGNIFICANT WORD				
RTP TIMESTAMP				
SENDER S PACKET COUNT				
SENDER S OCTET COUNT				
. . . RECEPTION REPORTS . . .				
. . . PROFILE-SPECIFIC EXTENSIONS. . .				

- **Superset of RR, used when sender has sent RTP packets**

RTCP Source Description (SDES)

<table>
<tr><td>V=2</td><td>P</td><td>SC</td><td>PT=SDES=202</td><td>LENGTH</td></tr>
<tr><td colspan="5">SSRC / CSRC_1</td></tr>
<tr><td colspan="5">SDES ITEMS FOR SSRC / CSRC_1</td></tr>
<tr><td colspan="5">. . . ADDITIONAL SDES CHUNKS . . .</td></tr>
</table>

	LENGTH	USER AND DOMAIN NAME . . .
	LENGTH	COMMON NAME OF SOURCE. . .
	LENGTH	EMAIL ADDRESS OF SOURCE. . .

Canonical End-Point Identifiers

```
 0                   1                   2                   3
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|    CNAME=1    |     length    | user and domain name        ...
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

- **SDES that must appear in every compound packet**
- **CNAME identifies each participant uniquely**
  - Stays constant even if SSRC changes (by conflict or restart)
  - Binds streams from multiple media tools to same source
  - For monitoring, intended to be human-readable as well
  - Translators should translate RFC 1918 addresses to ensure global uniqueness

Analyzing RTCP Reports

- **Cumulative counts for long- and short-term analysis**
  - Subtract any two reports to get activity over interval
  - NTP timestamps in reports allow you to compute rates
  - Monitoring tools needn't understand data encodings
- **Sender reports give utilization information**
  - Average packet rate and average data rate over any interval
  - Monitoring tools can compute this without seeing the data
- **Receiver reports give loss and round-trip information**
  - Extended sequence number conveys # packets expected
  - Packets lost and packets expected give long term loss rate
  - Fraction lost field gives short-term loss rate
  - LSR and DLSR give senders ability to compute round-trip time

RTP Profiles

- **Provide interpretation of generic fields**
  - Mapping from payload type number to encodings
  - Use of marker bits
  - Frequency of timestamp counter
- **Other items which may be specified in a profile**
  - RTP header extensions
  - Additional RTCP packet types
  - RTCP report interval
  - Use of security/encryption
- **No support for parameter negotiation, membership control**
  - Protocols such as SDP and SIP handle this

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- **Session Initiation Protocol (SIP)**

SDP [RFC 4566]

- **Originally designed for multimedia multicast**
  - Session directory tool advertises multimedia conferences
  - Must communicate the conference addresses
  - Must communicate app-specific information necessary for participation.
- **SDP is designed to convey such information**
- **Can use multiple transport protocols including:**
  - SAP (Session Announcement Protocol)
  - Email using MIME extensions
  - HTTP

Session Information conveyed by SDP

- **Session name and purpose**
- **Time(s) the session is active**
  - Arbitrary list of start/stop times
  - Repeat times (e.g., every Tuesday at 2:45pm)
- **The media that constitute the session**
  - Type (audio, video), format, protocol
  - Network (possibly multicast) address, protocol, port
- **Other useful info:**
  - Bandwidth required
  - Contact information for a responsible person
  - etc.

Session Descriptions

- **Compact, but entirely text-based**
  - Lines of the form: $type = \langle value \rangle$
  - Facilitates embedding in various transport methods
- **Begins with session-level section**
  - Starts with line “texttv=0” (version 0)
  - Contains lines that apply to all media streams
  - Ends at first “`m=`…” line
- **Followed by zero or more media-level descriptions**
  - Starts with “`m=`…”, ends before next “`m=`…” line
  - Can contain directives overriding session-level section
- **With some transports, can concatenate sessions**
  - Each “`v=0`” starts a new session description

Session Description Syntax

- **Session description (* = optional)**
  - `v=` protocol version
  - `o=` owner/creator and session number
  - `s=` session name
  - `i=*` session information
  - `u=*` URI of description
  - `e=*` email address
  - `p=*` phone number
  - `c=*` connection information—optional if in all media
  - `b=*` bandwidth information
  - >> one or more time descriptions
  - `z=*` time zone adjustments
  - `k=*` encryption key
  - `a=*` zero or more session attribute lines
  - >> zero more media descriptions

Session Description Syntax continued

- **Time description**
  - `t=` time the session is active
  - `r=*` zero or more repeat times
- **Media description**
  - `m=` media name and transport address
  - `i=*` media title
  - `c=*` connection info.—optional defined at session level
  - `b=*` bandwidth information
  - `k=*` encryption key
  - `a=*` zero or more session attribute lines
- **The type set is small, not extensible**
  - SDP parsers must ignore unknown announcement types
- **Use attributes for media-specific details**

SDP Example

- **Example session description:**

```
v=0
o=cs144-staff 2890844526 2890842807 IN IPv4 171.16.64.4
s=SDP Lecture
i=A Leecture on the session description protocol
u=http://cs144.scs.stanford.edu/notes/l13.pdf
c=IN IP4 224.2.17.12/127
t=2873397496 2873404696
a=recvonly
m=audio 3456 RTP/AVP 0
m=video 2232 RTP/AVP 31
m=whiteboard 32416 udp wb
a=orient:portrait
```

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Session Initiation Protocol [RFC 3261]

- **SIP is a protocol designed to enable the invitation of users to participate in multimedia session**
  - Not tied to a specific conference control scheme
  - Supports loosely or tightly controlled sessions
  - Enables user mobility by relaying and redirecting invitations to a user's current location
- **Communication is between *users*, not hosts**
  - User identifiers define control path (whom to ask about user)
  - Data path (actual media) can be completely decoupled

Session Initiation Protocol (2)

- **SIP is a *control* protocol for creating/modifying/ terminating sessions w. one or more participants**
- **Examples of sessions are:**
  - Internet telephone calls
  - Internet multimedia conferences
  - Internet multimedia distribution
- **Communication among members in a session may be:**
  - Via multicast,
  - Via a mesh of unicast “relations”
  - Or via a combination of both

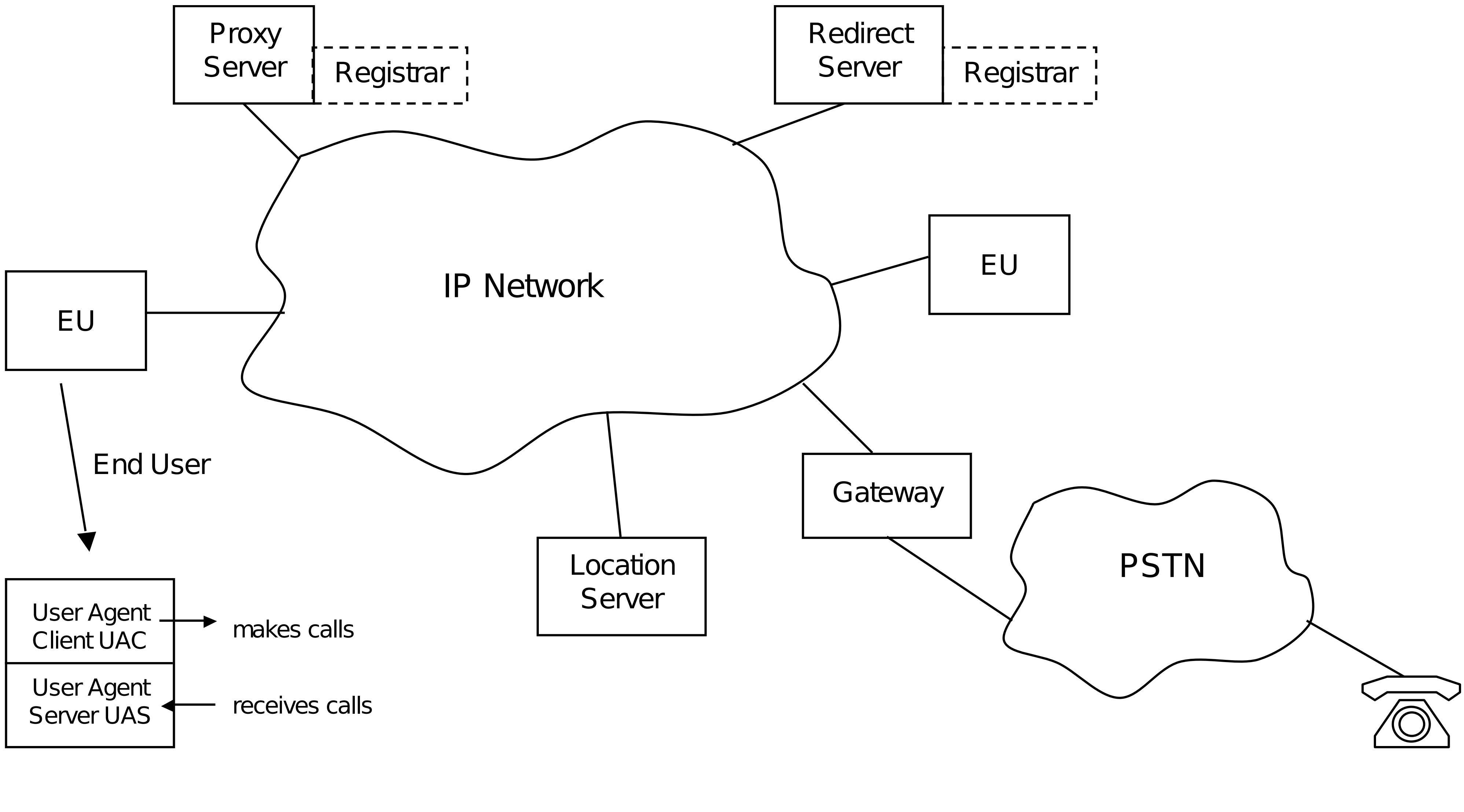
Functional Features

- Allows participants to agree on a set of compatible media types through SDP
- Supports user mobility by proxying and redirecting requests to the user's current location.
- Can be extended with additional capabilities.
- It is not tied to any particular conference control protocol
- It is independent of the lower layer transport protocol

Call Setup

- **Initial phase**
  - Client tries to find address at which to contact remote user
- **Subsequent phases**
  - Implement request-response protocol
  - Session description is sent with an invitation to join
- **Status code responses**
  - Informational – request received, continuing process
  - Success – action received, understood, and accepted
  - Redirection – client must undertake further action to complete request
  - Client error – request contains bad syntax
  - Server error – server couldn't complete valid request

Big Picture


Proxy Server
Registrar
Redirect Server
Registrar
EU
IP Network
EU
End User
Gateway
Location Server
PSTN
User Agent Client UAC
makes calls
User Agent Server UAS
receives calls

Addressing

- **SIP addresses are URLs: user@host**
  - user: user name or telephone number
  - host: domain name or numeric IP address
- **Examples:**
  - `sip:dm@scs.stanford.edu`
  - `sip:16505555555@sip.future-nine.com`
- **To send a message, a SIP client can send it to a pre-configured proxy, or use DNS:**
  - Check for DNS SRV records
  - Then check for MX records
  - Finally, use an A record

Components (1)

- **Clients**
  - End systems
  - Send SIP requests
  - Usually also contain SIP user agent server (UAS), which listens for call requests, prompts user or executes program to determine response
- **Proxy server**
  - Proxies request to another server
  - Possibly translates and rewrites request
  - Can “fork” request to multiple servers, creating search tree

Components (2)

- **Redirect Server**
  - Redirects users to try another server (user agent may act as redirect server)
- **Location Server (or service)**
  - Used by SIP redirect or proxy server to obtain information about a user’s possible location(s)
  - May be co-located with a SIP server but the manner in which a SIP server requests location services is beyond the scope of SIP
  - May be anything (LDAP, whois, local file, result of program execution)

Components (3)

- **Registrar**
  - A server that accepts REGISTER requests
  - Typically co-located with a proxy or redirect server
  - Allows a client to let the proxy or redirect server know at which address(es) it can be reached

Methods (1)

- **There are 6 methods in SIP**
- **Invite**
  - Invites a participant to a conference
  - Conference can be unicast, multicast, new or in existence
- **Bye**
  - Ends a client's participation in a call
- **Cancel**
  - Terminates a search
- **Options**
  - Queries a participant about their media capabilities, and finds them, but doesn't invite

Methods (2)

- **Ack**
  - Call acceptance
  - Reliability
- **Register**
  - Informs a SIP server about the location of a user

Responses (1)

- **1xx – Informational**
  - Request received, continuing to process request
  - Examples: 100 trying, 180 ringing, 181 call is being forwarded, 182 queued
- **2xx – Success**
  - Action successfully received, understood, and accepted
  - Example: 200 OK
- **3xx – Redirection**
  - Further action must be taken in order to complete the request
  - Examples: 300 multiple choices, 301 moved permanently, 302 moved temporarily, 305 use proxy

Responses (2)

- **4xx – Client error**
  - Request contains bad syntax
  - Examples: 400 bad request, 401 unauthorized, 402 payment required
- **5xx – Server error**
  - Server failed to fulfill an apparently valid request
  - Examples: 500 internal server error, 501 not implemented, 502 bad gateway
- **6xx – Global failure**
  - The request cannot be fulfilled at any server
  - Examples: 600 busy everywhere, 604 does not exist everywhere, 606 not acceptable

Message Syntax

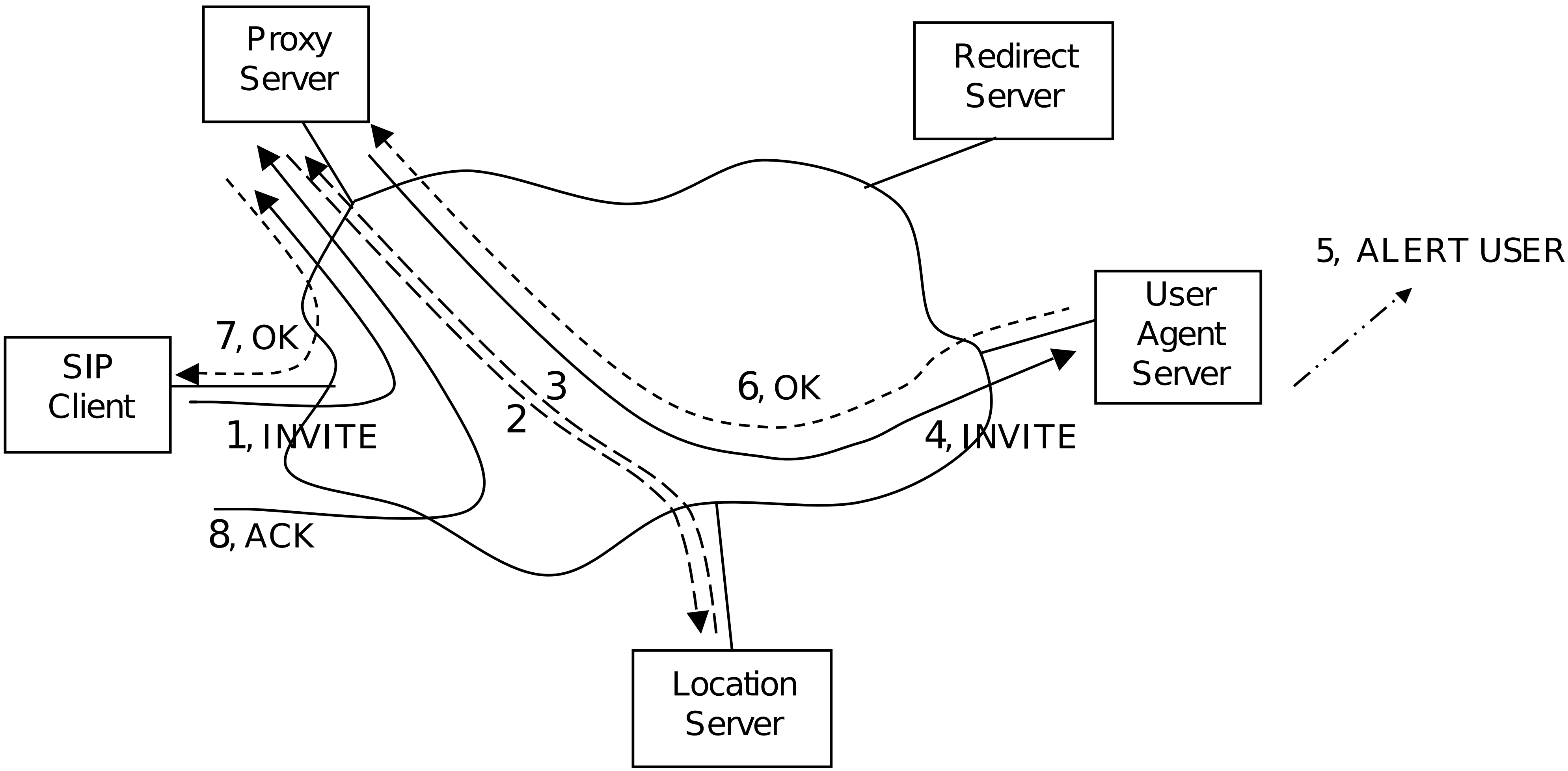
- **Text based**
- **Many header fields from http**
- **Some new ones**
  - Via
- **Payload may contain media description**
  - typically uses SDP, Session Description Protocol

```
INVITE sip:dm@scs.stanford.edu SIP/2.0
From: sip:pal@scs.stanford.edu
To: dm@scs.stanford.edu
Call-ID: 19990321@scs.stanford.edu
Cseq: 10 INVITE
v=0
o=user1 12345 6789 IN IP4 171.0.1.2
s=Multimedia Networks
i=Presentation Multimedia Networks and
Communication
e=pal@scs.stanford.edu
c=IN IP4 224.2.0.1/127
t=0 0
m=audio 3456 RTP/AVP 0
```

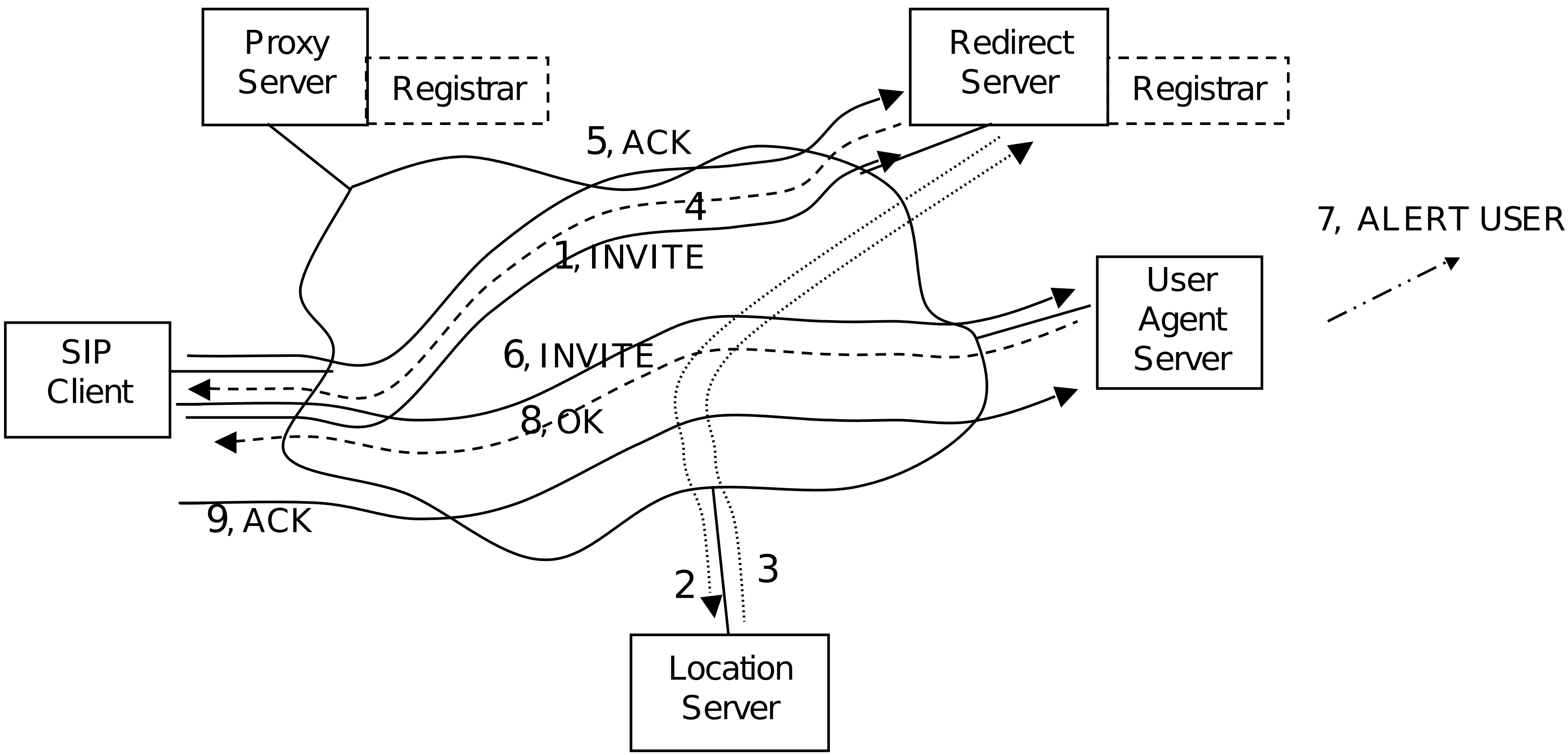
Basic Operation

- **Client sends req. to locally configured proxy**
  - Or obtains domain server IP address (using DNS)
- **Call initiator contacts SIP server for domain**
- **Location server locates receiver**
- **Call is established**
  - Initiator sends an INVITE request
  - Invited party answers (agrees)
  - Initiator receives OK indication
  - Initiator sends an ACK request

Proxy Example



Redirect Example



Other Features

- **Multiple call acceptances**
  - Client control
  - Client selection
  - Multiparty conferencing
- **Security**
  - Encryption and authentication end-to-end
  - Uses existing mechanisms
- **Messaging**
  - SIP is behind AIM and many other IM systems
  - SIMPLE: SIP for Instant Messaging and Presence Leveraging Extensions (simple)

Services

- Call forwarding
- Hold
- Blind call transfer
- Transfer with notification
- Operator assisted transfer
- Full mesh unicast conferences
- Multicast conferences
- Authenticated Caller ID
- *66
- Local services: call waiting, mute, *69