

## Control-Plane Signaling in the Internet and Multimedia Networking Basis

Dr. Xiaoming Fu

Telematics group  
University of Göttingen, Germany

## Overview

- Review of functional composition of an IP router
- Review of MPLS switching v.s. IP forwarding
- Control-plane Signaling: RSVP(-TE), (CR-)LDP, NSIS
- Missing components with multimedia networking
  - Multimedia application characteristics and classification
  - Multimedia control/quality enhancement in application layer

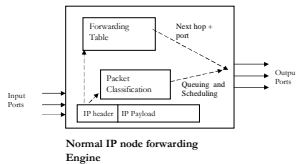
### Credits:

- Kurose and K. Ross, Computer Networking: A Top-Down Approach Featuring the Internet, 2nd Ed., Addison-Wesley, 2002
- Henning Schulzrinne, Columbia University

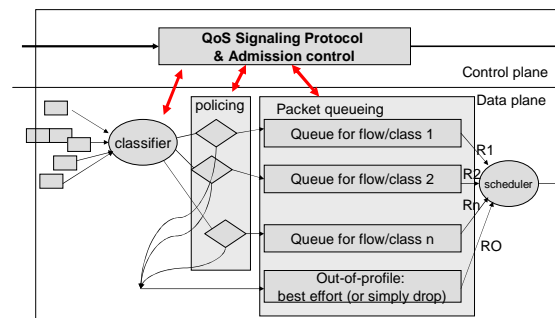
## IP Node Forwarding

Two key router functions:

- *Control plane*: run routing algorithms/protocol (RIP, OSPF, BGP), and write the results (periodically) to the data path (as forwarding table)
- *Data (forwarding) plane*: switching datagrams from incoming to outgoing link (Longest-prefix match based on packet's destination IP address)



## A logical architecture for IP QoS



## Signaling for QoS reservation setup

- From the previous figure, QoS signaling is an important component in control plane to manipulate parameters for use in data plane
- In the Internet, QoS signaling can rely on an IP signaling protocol
  - RSVP (Resource reSeVation Protocol) is the well-known protocol
  - NSIS (Next Steps in Signaling) is a recent proposal for IP signaling

## Resource Reservation: RSVP

- RSVP initially was designed for establishing a guaranteed QoS path between a sender and receiver(s).
- RSVP establishes end-to-end reservations over a connectionless network.
- RSVP is robust when routers/links fail: traffic is re-routed and new reservations are established.
- RSVP is receiver-initiated and so is designed with multicast in mind.

## Resource Reservation: RSVP

The network needs to know the TSpec, the RSpec and the Path followed by packets.

- The TSpec (specification of the transmitted traffic) is only known by the source.
- The Path is only known by the network.
- The RSpec (specification of what the receiver would like to receive).

## Resource Reservation: RSVP

So, the sender periodically sends the Tspec to the whole multicast group ("PATH messages").

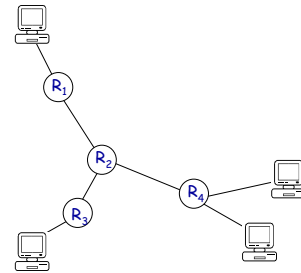
- The network learns the Path taken by packets in the multicast group.
- The receiver/network learns the TSpec.

## Resource Reservation: RSVP

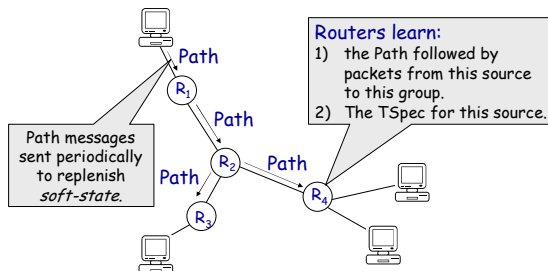
To initiate a new reservation, a receiver sends messages to reserve resources "up" the multicast tree ("RESV messages").

- The routers forward RESV messages towards the source.
- The routers determine if the reservation can be fulfilled.
- If necessary/possible, the routers merge the requests from different receivers.

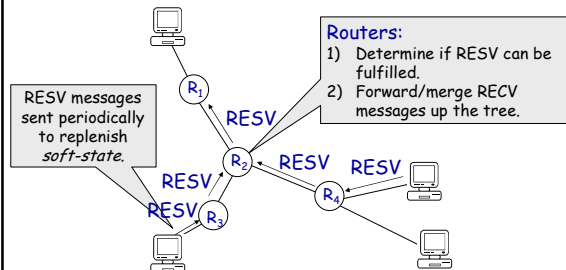
## Establishing a reservation 1: The multicast group is established



## Establishing a reservation 2: RSVP Path messages sent by source(s)



## Establishing a reservation 3: RSVP RESV messages sent by receiver(s)



### Establishing a reservation Merging RESV messages

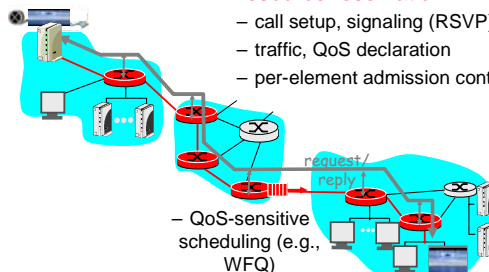
Examples:

1. Router receives RESVs from two receivers, A and B, asking for 100ms delay and 50ms delay respectively. Router passes up request for 50ms.
2. Router receives RESVs for a audio teleconference call with 100 participants requesting 1.5Mb/s each. The tree need support only 1.5Mb/s total data-rate.

RSVP supports many styles of RESV merging.

### RSVP QoS Signaling for Intserv

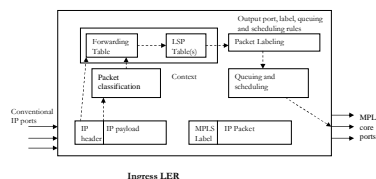
- Resource reservation
  - call setup, signaling (RSVP)
  - traffic, QoS declaration
  - per-element admission control



### MPLS vs. (native) IP networks

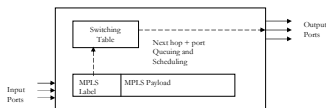
- Recall the difference between MPLS switching and IP routing networks
  - An MPLS LSR switches MPLS-encapsulated received IP packets based on MPLS labels (in hardware) to a next LSR
    - per-label switching: faster
  - An IP router lookup IP routing table and switches IP packets based on IP header information to next IP router
    - per-packet header lookup: slower
- Common requirements: control plane information setup & maintenance
  - MPLS: distribution of routing info & label, (can also carry QoS reservations if wanted)
  - IP: routing tables, QoS reservations
  - We call those protocols (except routing) as control plane signaling protocols

### Label Edge Router



- LER terminates and originates LSP's and performs both label based forwarding and conventional NIF functions.
- **Ingress LER** - labels unlabelled packets and creates an initial MPLS frame by pushing one or more MPLS label entries.
- **Egress LER** - terminates LSP by popping the top MPLS stack entry.

### Label Based Forwarding



- At each LSR, forwarding is done by the single index lookup into the switching table using the packet's MPLS label.
- The switching table is loaded a priori with a unique next-hop label, output port and queuing and scheduling rules.
- The establishment of mapping information is responsibility of control part
  - done using Label Distribution Protocols: e.g., CR-LDP, RSVP-TE

### Label Distribution Protocol (LDP)



LDP and its Constraint Routed (CR) extension:

- Hard State Protocol
- UDP used for peer discovery
- TCP used for session, advertisement, notification, and LDP messages
- Supports Diffserv and Operator configurable QOS classes
- Failure reported using the reliable TCP

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## Label Distribution Using RSVP-TE

- RSVP extension for Traffic Engineering (RSVP-TE): RFC3209
  - Additions to RSVP signaling protocol
- Leverage the admission control mechanism of RSVP
- Label requests are sent in PATH messages and binding is done with RESV messages
- EXPLICIT-ROUTE object defines the path over which setup messages should be routed
  - Different from original RSVP where an IP alert option is attached for discovery of the path

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## CR-LDP Vs RSVP-TE

- Choice of Transport Protocol
  - UDP + TCP sessions Vs Raw IP
- Multipoint Support
  - Multipoint-to-Point : (Both support)
  - Multicast (Point-to-Multipoint) : Yet to be addressed
  - Question: *is this really important?*
- Scalability
  - Network Flows
    - RSVP extensions (baseRFC2205+RFC2961+RFC3209) for refreshing many LSPs in a single RSVP BUNDLE message and ability to indicate state index rather than having to send the entire payload
  - Others: CPU Utilization, Data Storage Requirements

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## Next steps in signaling (NSIS)

- RSVP not widely used for resource reservation
  - but is used for MPLS path setup
  - design heavily biased by multicast needs
  - Limited ability signaling delivery any size of signaling & over congested situation
  - marginal and after-the-fact security
  - limited support for IP mobility
- Thus, IETF NSIS working group developing new framework for general state management protocol
  - resource reservation
  - NAT and firewall control
  - traffic and QoS measurement
  - Any other signaling purposes for IP-supported nodes
- Split into two components:
  - NSLP: services
  - NTLP: transport using UDP and TCP (similar to CR-LDP)

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## NSIS

- On-path vs. off-path
  - off-path → bandwidth brokers
- Discovery of next NTLP or NTLF+NSLP hop
  - use router alert option

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## Review: What is quality of service?

- Many applications are sensitive to the effects of delay (+ jitter) and packet loss
  - may have "floor" below which utility drops to zero
- The existing Internet architecture provides a best effort service.
  - All traffic is treated equally (generally, FIFO queuing)
  - No mechanism for distinguishing between delay sensitive and best effort traffic
- Original IP architecture (IPv4) has TOS (type-of-service byte) in packet header
  - RFC 795: defined multiple axes (delay, throughput, reliability)
  - rarely used outside some (rumor) military networks
- IP QoS slow to deploy, what about application layer (assuming best effort networks)?
  - **Let's check features of multimedia applications**

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## MM Networking Applications

**Classes of MM applications:**

- 1) Streaming stored audio and video
- 2) Streaming live audio and video
- 3) Real-time interactive audio and video

**Fundamental characteristics:**

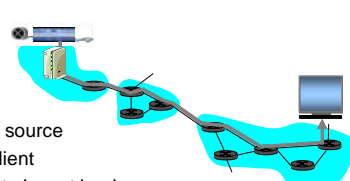
- Typically **delay sensitive**
  - end-to-end delay
  - delay jitter
- But **loss tolerant**: infrequent losses cause minor glitches
- Antithesis of data, which are loss intolerant but delay tolerant.

**Jitter** is the variability of packet delays within the same packet stream

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### Streaming Stored Multimedia



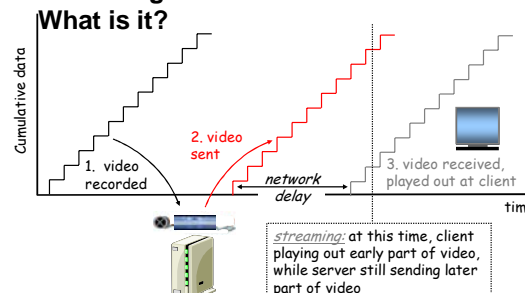
**Streaming:**

- media stored at source
- transmitted to client
- streaming: client playout begins *before* all data has arrived
  - timing constraint for still-to-be transmitted data: in time for playout

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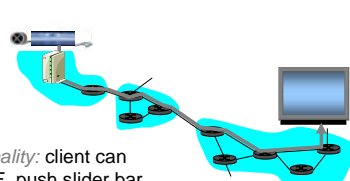
### Streaming Stored Multimedia: What is it?



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### Streaming Stored Multimedia: Interactivity



- *VCR-like functionality*: client can pause, rewind, FF, push slider bar
  - 10 sec initial delay OK
  - 1-2 sec until command effect OK
  - RTSP often used (more later)
- timing constraint for still-to-be transmitted data: in time for playout

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### Streaming Live Multimedia

**Examples:**

- Internet radio talk show
- Live sporting event

**Streaming**

- playback buffer
- playback can lag tens of seconds after transmission
- still have timing constraint

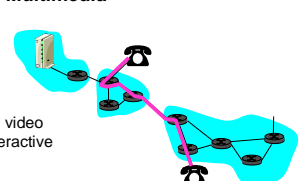
**Interactivity**

- fast forward impossible
- rewind, pause possible!

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### Interactive, Real-Time Multimedia



- **applications**: IP telephony, video conference, distributed interactive worlds
- **end-end delay requirements**:
  - audio: < 150 msec good, < 400 msec OK
    - includes application-level (packetization) and network delays
    - higher delays noticeable, impair interactivity
- **session initialization**
  - how does callee advertise its IP address, port number, encoding algorithms?

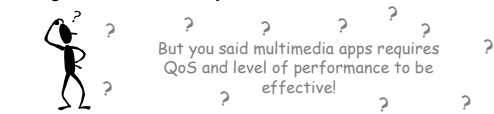
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### Multimedia Over Today's Internet

**TCP/UDP/IP: "best-effort service"**

- *no* guarantees on delay, loss



But you said multimedia apps requires QoS and level of performance to be effective!

Today's Internet multimedia applications use application-level techniques to mitigate (as best possible) effects of delay, loss

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## How should the Internet evolve to better support multimedia?

**Integrated services philosophy:**


- Fundamental changes in Internet so that apps can reserve end-to-end bandwidth
- Requires new, complex software in hosts & routers

**Laissez-faire**

- no major changes
- more bandwidth when needed
- content distribution, application-layer multicast
  - application layer

**Differentiated services philosophy:**

- Fewer changes to Internet infrastructure, yet provide 1st and 2nd class service.



What's your opinion?

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## Streaming Stored Multimedia

Application-level streaming techniques for making the best out of best effort service:

- client side buffering
- use of UDP versus TCP
- multiple encodings of multimedia

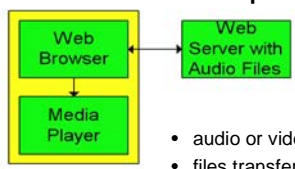
**Media Player**

- jitter removal
- decompression
- error concealment
- graphical user interface w/ controls for interactivity

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## Internet multimedia: simplest approach



- audio or video stored in file
- files transferred as HTTP object
  - received in entirety at client
  - then passed to player

**audio, video not streamed:**

- no, "pipelining," long delays until playback!

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## Streaming Multimedia: UDP or TCP?

**UDP**

- server sends at rate appropriate for client (oblivious to network congestion!)
  - often send rate = encoding rate = constant rate
  - then, fill rate = constant rate - packet loss
- short playout delay (2-5 seconds) to compensate for network delay jitter
- error recover: time permitting

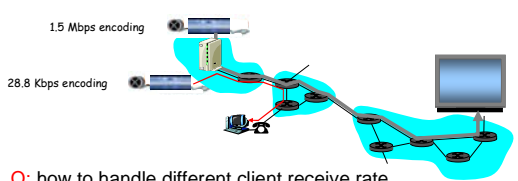
**TCP**

- send at maximum possible rate under TCP
- fill rate fluctuates due to TCP congestion control
- larger playout delay: smooth TCP delivery rate
- HTTP/TCP passes more easily through firewalls

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## Streaming Multimedia: client rate(s)



**Q:** how to handle different client receive rate capabilities?

- 28.8 Kbps dialup
- 100Mbps Ethernet

**A:** server stores, transmits multiple copies of video, encoded at different rates

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## User Control of Streaming Media: RTSP

**HTTP**

- Does not target multimedia content
- No commands for fast forward, etc.

**RTSP: RFC 2326**

- Client-server application layer protocol.
- For user to control display: rewind, fast forward, pause, resume, repositioning, etc...

**What it doesn't do:**

- does not define how audio/video is encapsulated for streaming over network
- does not restrict how streamed media is transported; it can be transported over UDP or TCP
- does not specify how the media player buffers audio/video

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### RTSP: out of band control

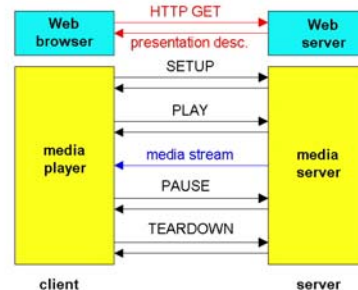
**FTP uses an "out-of-band" control channel:**

- A file is transferred over one TCP connection.
- Control information (directory changes, file deletion, file renaming, etc.) is sent over a separate TCP connection.
- The "out-of-band" and "in-band" channels use different port numbers.

**RTSP messages are also sent out-of-band:**

- RTSP control messages use different port numbers than the media stream: out-of-band.
  - Port 554
- The media stream is considered "in-band".

### RTSP Operation



### Real-time interactive applications

- PC-2-PC phone
  - instant messaging services are providing this
- PC-2-phone
  - Dialpad
  - Net2phone
- videoconference with Webcams

Going to now look at a PC-2-PC Internet phone example in detail

### Interactive Multimedia: Internet Phone

**Introduce Internet Phone by way of an example**

- speaker's audio: alternating talk spurts, silent periods.
  - 64 kbps during talk spurt
- pkts generated only during talk spurts
  - 20 msec chunks at 8 Kbytes/sec: 160 bytes data
- application-layer header added to each chunk.
- Chunk+header encapsulated into UDP segment.
- application sends UDP segment into socket every 20 msec during talkspurt.

### Internet Phone: Packet Loss and Delay

- **network loss:** IP datagram lost due to network congestion (router buffer overflow)
- **delay loss:** IP datagram arrives too late for playout at receiver
  - delays: processing, queueing in network; end-system (sender, receiver) delays
  - typical maximum tolerable delay: 400 ms
- loss tolerance: depending on voice encoding, losses concealed, packet loss rates between 1% and 10% can be tolerated.

### Recovery from packet loss (1)

**forward error correction (FEC): simple scheme**

- for every group of  $n$  chunks create a redundant chunk by exclusive OR-ing the  $n$  original chunks
- send out  $n+1$  chunks, increasing the bandwidth by factor  $1/n$ .
- can reconstruct the original  $n$  chunks if there is at most one lost chunk from the  $n+1$  chunks
- Tradeoff:
  - increase  $n$ , less bandwidth waste
  - increase  $n$ , longer playout delay
  - increase  $n$ , higher probability that 2 or more chunks will be lost

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### Recovery from packet loss (2)

**2nd FEC scheme**

- "piggyback lower quality stream"
- send lower resolution audio stream as the redundant information
- for example, nominal stream PCM at 64 kbps and redundant stream GSM at 13 kbps.

- Whenever there is non-consecutive loss, the receiver can conceal the loss.
- Can also append (n-1)st and (n-2)nd low-bit rate chunk

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### Recovery from packet loss (3)

**Interleaving**

- chunks are broken up into smaller units
- for example, 4 5 msec units per chunk
- Packet contains small units from different chunks
- if packet is lost, still have most of every chunk
- has no redundancy overhead
- but adds to playout delay

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### Real-Time Protocol (RTP)

- RTP specifies a packet structure for packets carrying audio and video data
- RFC 1889.
- RTP packet provides
  - payload type identification
  - packet sequence numbering
  - timestamping
- RTP runs in the end systems.
- RTP packets are encapsulated in UDP segments
- Interoperability: If two Internet phone applications run RTP, then they may be able to work together

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### RTP runs on top of UDP

RTP libraries provide a transport-layer interface that extend UDP:

- port numbers, IP addresses
- payload type identification
- packet sequence numbering
- time-stamping

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### RTP Example

- Consider sending 64 kbps PCM-encoded voice over RTP.
- Application collects the encoded data in chunks, e.g., every 20 msec = 160 bytes in a chunk.
- The audio chunk along with the RTP header form the RTP packet, which is encapsulated into a UDP segment.
- RTP header indicates type of audio encoding in each packet
  - sender can change encoding during a conference.
- RTP header also contains sequence numbers and timestamps.

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### RTP and QoS

- RTP does **not** provide any mechanism to ensure timely delivery of data or provide other quality of service guarantees.
- RTP encapsulation is only seen at the end systems: it is not seen by intermediate routers.
  - Routers providing best-effort service do not make any special effort to ensure that RTP packets arrive at the destination in a timely matter.

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## RTP Header



**Payload Type (7 bits):** Indicates type of encoding currently being used. If sender changes encoding in middle of conference, sender informs the receiver through this payload type field.

- Payload type 0: PCM mu-law, 64 kbps
- Payload type 3, GSM, 13 kbps
- Payload type 7, LPC, 2.4 kbps
- Payload type 26, Motion JPEG
- Payload type 31, H.261
- Payload type 33, MPEG2 video

**Sequence Number (16 bits):** Increments by one for each RTP packet sent, and may be used to detect packet loss and to restore packet sequence.

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## RTP Header (2)

- **Timestamp field (32 bytes long).** Reflects the sampling instant of the first byte in the RTP data packet.
  - For audio, timestamp clock typically increments by one for each sampling period (for example, each 125 usecs for a 8 KHz sampling clock)
  - if application generates chunks of 160 encoded samples, then timestamp increases by 160 for each RTP packet when source is active. Timestamp clock continues to increase at constant rate when source is inactive.
- **SSRC field (32 bits long).** Identifies the source of the RTP stream. Each stream in a RTP session should have a distinct SSRC.

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## SIP

- Session Initiation Protocol
- Comes from IETF
- SIP long-term vision
- All telephone calls and video conference calls take place over the Internet
- People are identified by names or e-mail addresses, rather than by phone numbers.
- You can reach the callee, no matter where the callee roams, no matter what IP device the callee is currently using.

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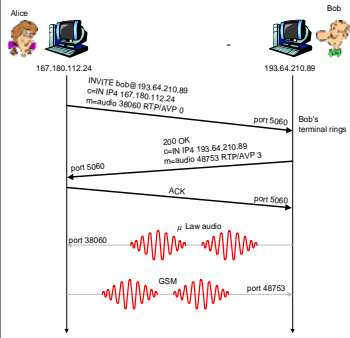
## SIP Services

- Setting up a call
  - Provides mechanisms for caller to let callee know she wants to establish a call
  - Provides mechanisms so that caller and callee can agree on media type and encoding.
  - Provides mechanisms to end call.
- Determine current IP address of callee.
  - Maps mnemonic identifier to current IP address
- Call management
  - Add new media streams during call
  - Change encoding during call
  - Invite others
  - Transfer and hold calls

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## Setting up a call to a known IP address



• Alice's SIP invite message indicates her port number & IP address. Indicates encoding that Alice prefers to receive (PCM ulaw)

• Bob's 200 OK message indicates his port number, IP address & preferred encoding (GSM)

• SIP messages can be sent over TCP or UDP; here sent over RTP/UDP.

• Default SIP port number is 5060.

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## Setting up a call (more)

- Codec negotiation:
  - Suppose Bob doesn't have PCM ulaw encoder.
  - Bob will instead reply with 606 Not Acceptable Reply and list encoders he can use.
  - Alice can then send a new INVITE message, advertising an appropriate encoder.
- Rejecting the call
  - Bob can reject with replies "busy," "gone," "payment required," "forbidden".
- Media can be sent over RTP or some other protocol.

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**Summary: Internet Multimedia: bag of tricks**

- Network core: **QoS** enabled (better control of forwarding resource)
- **use UDP** to avoid TCP congestion control (delays) for time-sensitive traffic
- client-side **adaptive playout delay**: to compensate for delay
- server side **matches stream bandwidth** to available client-to-server path bandwidth
  - chose among pre-encoded stream rates
  - dynamic server encoding rate
- error recovery (on top of UDP)
  - FEC, interleaving
  - retransmissions, time permitting
  - conceal errors: repeat nearby data
- Interactive multimedia: RTP for enhancing certain service quality; SIP for session initialization and parameter negotiation
- Control of streaming media: RTSP for (re)winding, pausing, ... (but fast forwarding for live stream impossible)  
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