The Integrated Services Internet: Multicast, Resource Reservations and the Real-Time Transport Protocol

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Overview

- multicast
- resource reservation protocols
- transport services for real-time data

Broadcast and Multicast

broadcast: all hosts on (small, local) network

directed broadcast: all hosts on remote network

multicast: multiple recipients (group)

Applications for Multicast

- audio-video distribution (1-to-many) and symmetric (all-to-all)
- distributed simulation (war gaming)
- resource discovery
- file distribution (stock market quotes, new software, …)

Connection-Oriented Multicast

- enumerate sources explicitly
- examples:
  - ATM (explicit connection to each end point)
  - ST-II (several end points in setup message)
  - enumeration of end points in packet
- only connection-oriented (packet header size!)
- source needs to know destinations ↔ resource discovery, dynamic groups

Host Group Model

Deering, 1991:

- senders need not be members;
- groups may have any number of members;
- there are no topological restrictions on group membership;
- membership is dynamic and autonomous;
- host groups may be transient or permanent.

Local Multicast

Some local networks are by nature multi/broadcast: Ethernet, Token Ring, FDDI, …

Ethernet, Tokenring:

- broadcast: all ones
- multicast: 01.xx.xx.xx.xx.xx
- adapter hardware can filter dynamic list of addresses

ATM: point-to-point links ➠ need ATM multicast server

IP Multicast

- host-group model
- network-level; data packets same, only address changes
- need help of routers
- special IP addresses (class D): 224.0.0.0 through 239.255.255.255
- 28 bits ➠ 268 million groups (plus scope)
- 224.0.0.1: all hosts
- some pre-assigned (224.0.1.2: SGI Dogfight)
- others dynamic (224.2.x.x for multimedia conferencing)
- map into Ethernet: 01.00.5E.00.00.00 + lower 23 bits

IGMP

Multicast for local (broadcast) networks, between router and hosts

- router listens to all multicast packets on all interfaces
- hosts sends report for first process to join group to multicast group
- host *does not* send report when processes all have left
- router multicasts query to all hosts ≈ once a minute
- host waits and listens for others; if nobody else, send response

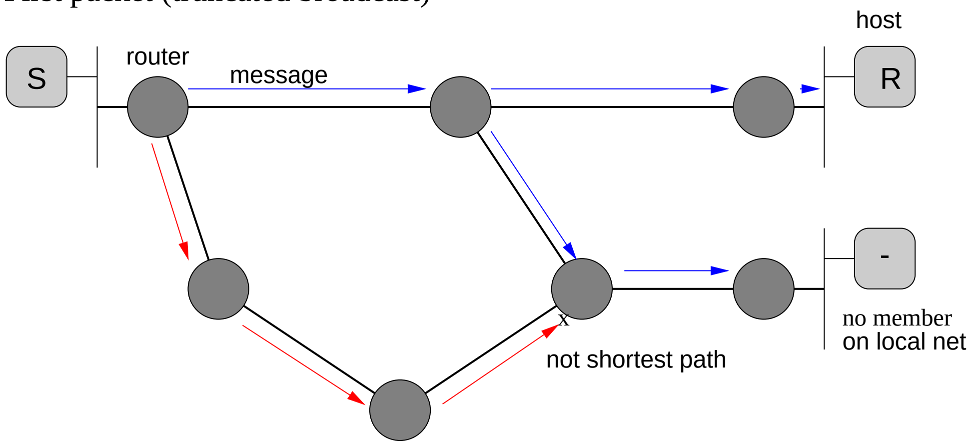
Multicast Forwarding

1. check incoming interface: discard if not on shortest path to source
2. forward to all outgoing interface except incoming
3. don't forward if interface has been *pruned*
4. prunes time out every minute
5. routers may send *grafts* upstream

routing information: DVMRP

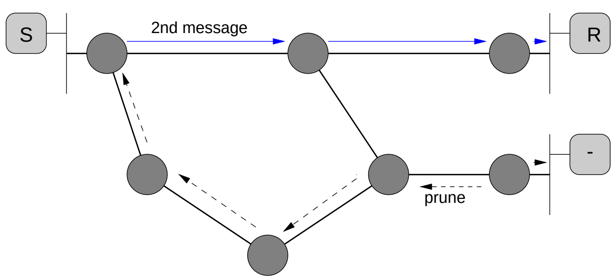
Multicast Forwarding

First packet (truncated broadcast)



Multicast Forwarding

With pruning:



➠ router needs to keep “negative” list for groups

Multicast Programming

UDP, not TCP (obviously…)

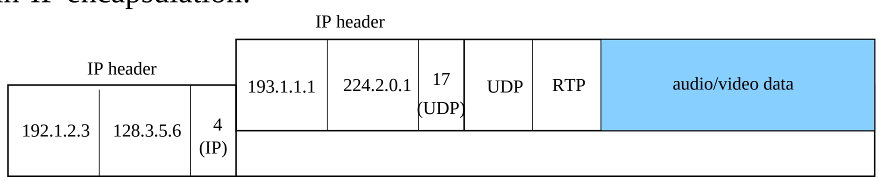
```
struct sockaddr_in name;
struct ip_mreq imr;

sock = socket(AF_INET, SOCK_DGRAM, 0);
imr.imr_multiaddr.s_addr = htonl(groupaddr);
imr.imr_interface.s_addr = htonl(INADDR_ANY);
setsockopt(sock, IPPROTO_IP, IP_ADD_MEMBERSHIP,
  &imr, sizeof(struct ip_mreq));
name.sin_addr.s_addr = htonl(groupaddr);
name.sin_port = htons(groupport);
bind(sock, &name, sizeof(name));
recv(sock, (char *)buf, sizeof(buf), 0);
```


Problems

- “multicast storms”
- state in routers for sparse groups vs. optimal trees
- multicast routing vs. unicast routing (reverse path)
- hierarchical routing

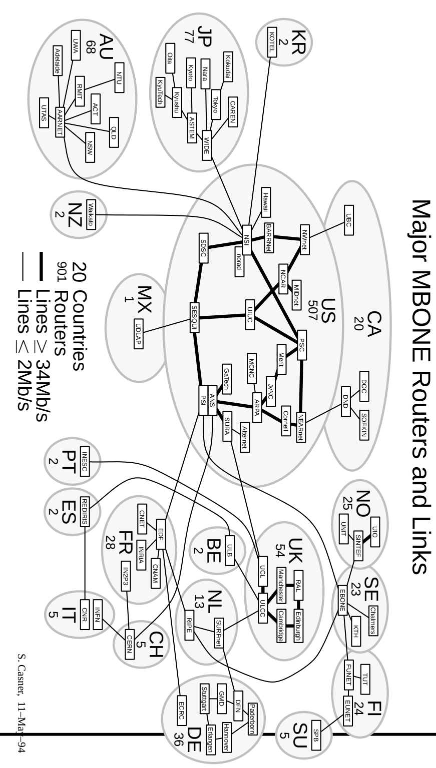
MBONE

- MBONE ≡ multicast backbone
- overlay network over Internet
- needed until deployment of multicast-capable backbone routers
- IP-in-IP encapsulation:



source: 193.1.1.1; group: 224.2.0.1; MBONE tunnel: 192.1.2.3 to 128.3.5.6

- limited capacity, resilience



Multicast: Further Information

- S. Deering, RFC 1112
- D. Comer, *Internetworking with TCP/IP*
- R. Perlman, *Interconnections - Bridges and Routers* (“IP multicast is bad”)
- C. Huitema, *Routing in the Internet*

Resource Reservation

Resource Reservation Issues

sender-oriented: sender specifies bandwidth, receivers ➠ ATM, ST-II

receiver-oriented: receivers notify sender ➠ RSVP

hard state: reliable connection set-up (handshake), explicit tear-down ➠ Q.2931, ST-II

soft state: probabilistic set up, time-outs ↔ periodic state refresh

Any protocol should almost never say “no” (unless it is run by a monopoly…)

RSVP

receiver-oriented reservation protocol being standardized by IETF:

- not a routing protocol, but interacts with routing
- transports *opaque* QOS and policy parameters for sessions
- simplex ➠ setup for unidirectional data flows
- does not prescribe admission or policy control
- sets up packet classifier, but does not handle packets
- independent sessions (can’t tie video and audio session)
- multicast (and unicast)
- either own protocol type or UDP encapsulated

RSVP Objects

Flow descriptor =

Flowspec: • service class

- Rspec ➠ desired QoS
- Tspec ➠ describes traffic characteristics

Filterspec: which packets get this treatment ➠ sender IP address/port, protocol, other fields ➠ complex (regular expressions? IP options!) ➠ currently, sender IP address and UDP/TCP port ➠ no fragmentation

Reservation Styles

sender selection	reservations distinct for each sender	shared
explicit	fixed filter	shared-explicit
wildcard (all)	–	wildcard filter

➠ mutually incompatible

shared: only one active data source ➠ e.g., reserve for twice needed for audio

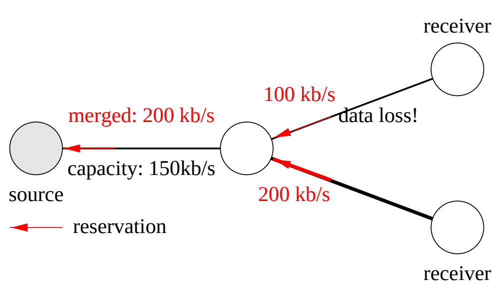
distinct: video

RSVP: Basic Operation

- source sends PATH messages to receivers: previous hop to source, Tspec, OPWA
- receivers send RESV messages back to senders
- reservations are merged at each node for same sender (max. flowspec)
- merge point or data sender may send confirmation (if requested)
- reservations *may* get merged between senders (audio!)
- one-pass ➠ receiver doesn't know final QoS ➠ One Pass With Advertising
- application *should* explicitly tear down reservations

Killer Reservations

1. small reservation in place; another receiver larger reservation ➠ failure? ➠ keep old
2. large reservation fails again and again ➠ blocks new, smaller one

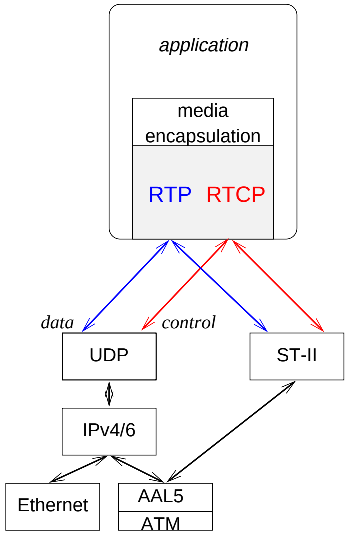


Mapping RSVP onto ATM

	IP, RSVP	ATM
multicast tree, reservation	sequential	same time
origin	receiver	sender (root) ➠ UNI4.0
change reservations	yes	no
routing changes	time-out	re-establish VC
flow merging (audio)	yes	no (separate VCs)
receiver diversity	not yet	no

Real-Time Transport Protocol

RTP: The Big Picture



RTP: Real-Time Transport Protocol

- only part of puzzle
- product of Internet Engineering Task Force, AVT WG
- RFC 1889, 1890
- ITU H.323, Netscape
- support for functions, but does not restrict implementation
- compression for low-bandwidth networks under study

RTP: Goals

lightweight: specification and implementation

flexible: provide mechanism, don't dictate algorithms

protocol-neutral: UDP/IP, ST-II, IPX, ATM-AALx, ...

scalable: unicast, multicast from 2 to $O(1000)$

separate control/data: some functions may be taken over by conference control protocol

secure: support for encryption, possibly authentication

RTP: Functions

- segmentation/reassembly done by UDP (or similar)
- resequencing (if needed)
- loss detection for quality estimation, recovery
- intra-media synchronization: remove delay jitter through playout buffer
- intra-media synchronization: drifting sampling clocks
- inter-media synchronization (lip sync)
- quality-of-service feedback and rate adaptation
- source identification

RTP: Mixers, Translators, ...

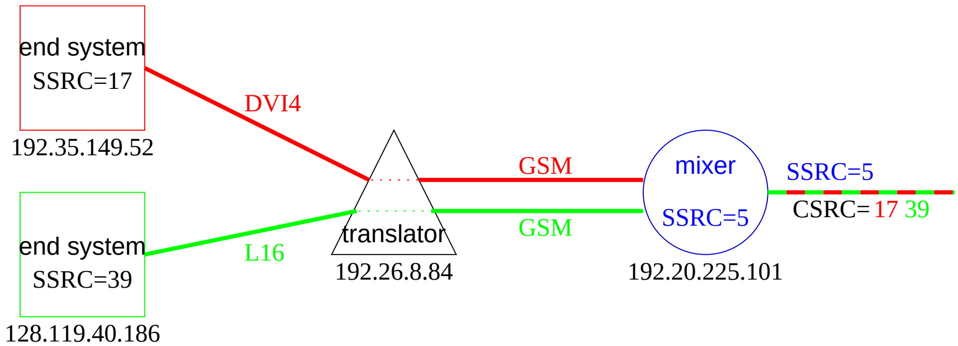
mixer:

- several media stream ➠ one new stream (new encoding)
- mixer: reduced bandwidth networks (dial-up)
- appears as new source

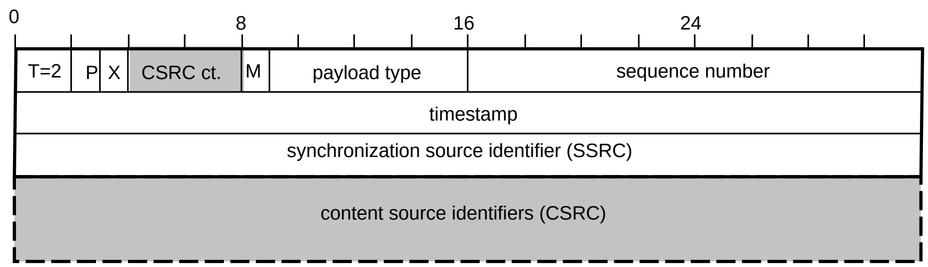
translator:

- single media stream
- *may* convert encoding
- protocol translation, firewall

RTP: Mixers, Translators, ...



RTP: Packet format



P: padding (for encryption)

M: marker bit; indicates frame, talkspurt

CC: content source count

Payload type: audio, video encoding method

RTP: Control Protocol – Algorithm

Goals:

- estimate current number of participants – dynamic
- participant information ➠ talker indication
- quality-of-service feedback ➠ adjust sender rate
- scale to $O(1000)$ participants, small fraction of data bandwidth

➠ randomized response with rate ↓ as members ↑

- limited by tolerable age of status
- gives active senders more bandwidth

RTP: Control Protocol – Types

stackable packets, similar to data packets

sender report: bytes send ➠ estimate rate;
timestamp ➠ synchronization

reception reports: number of packets sent and expected ➠ loss,
interarrival jitter, round-trip delay

source description: name, email, location, …

explicit leave: in addition to time-out

extensions: application-specific

RTP: Control Protocol — SR

0 8 16 24

T=2	P	receiver count	payload type = SR	packet length
synchronization source identifier (SSRC)				
NTP real−time timestamp				
RTP sample−clock timestamp				
sender's packet count				
sender's byte count				
synchronization source identifier of remote source				
cumulative number of packets received				
cumulative number of packets expected				
interarrival jitter				
last SR NTP timestamp seen from this source				
delay since receiving that SR NTP timestamp				

for each site received

RTP Implementations

tool	who	media	RSVP
NeVoT	GMD Fokus	audio	soon
NeViT	GMD Fokus	video	yes
vic	LBNL	video	no
vat	LBNL	audio	no

`http://www.fokus.gmd.de/step/rtp/`

Conclusion

- introduced range of approaches:
  - adaptive applications
  - layered applications
  - resource reservation
  - transport protocols
- authentication
- charging for shared reservations?