MBONE: Multicast + Real-Time Audio and Video over the Internet

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Overview

- Internet as "integrated services network"
- Multicast: why, how
- The MBONE
- real-time audio and video
  - requirements
  - network support and resource reservation
  - transport protocols

“Classic” Internet to “Integrated” Internet

“Classic” Internet:

- data services
- email, file transfer, WWW, …
- unicast
- single class of service

“Integrated Services” Internet adds…

- real-time services (audio, video, …)
- resource reservation, differentiated service?
- multicast

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 $\approx$ once a minute
- host waits and listens for others; if nobody else, send response

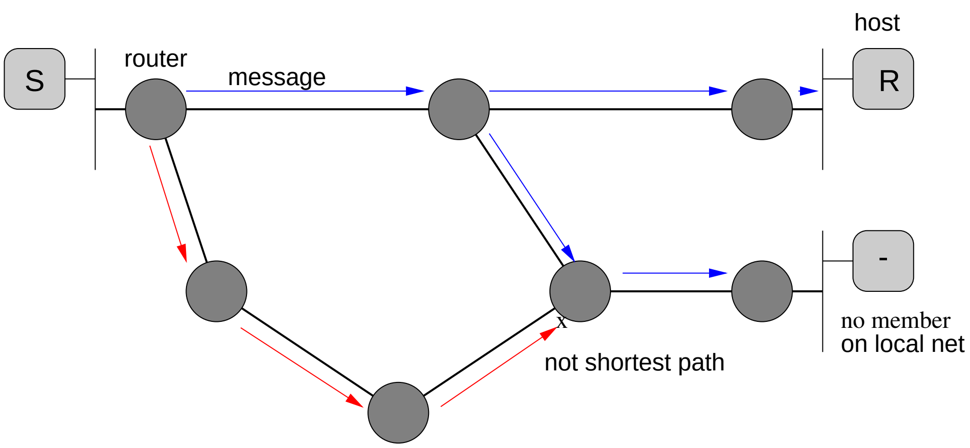
Multicast Forwarding

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

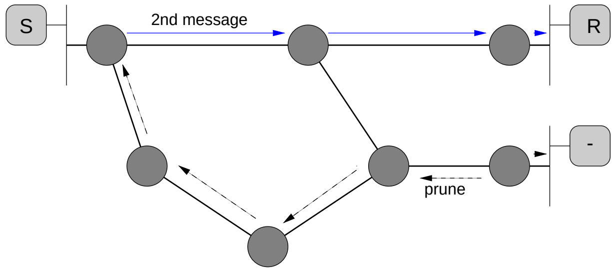
Multicast Forwarding

First packet (truncated broadcast)



Multicast Forwarding

With pruning:



➠ router needs to keep “negative” list for groups

Multicast Programming

UDP, not TCP (obviously…)

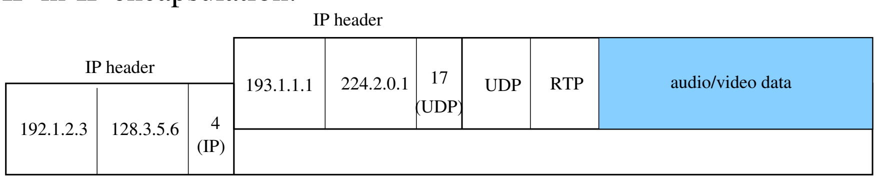
```
struct sockaddr_in name;
struct ip_mreq imr;

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

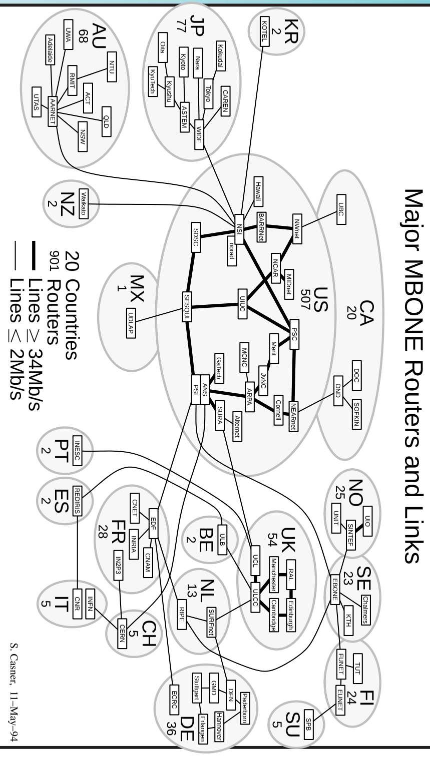
MBONE

- 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*

Real-Time Services

Why bother integrating?

- use existing workstations as audio/video terminals
- use existing LAN/WAN infrastructure
- efficiency:
  - true (LAN/WAN) multicast instead of MCUs and bridges – same application scales from two to hundreds of receivers
  - variable-bit rate (VBR) video, but interoperation with H.261 standards
  - audio silence suppression – important for large-scale conferences

Why bother integrating (cont'd)?

- added control functionality:
  - directory services
  - visual speaker indication
  - source selection at receiver
- integration of application-sharing and data applications, WWW

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Disadvantages of packet audio/video

- no resource reservation ➠ quality may suffer (but: RSVP)
- may push overloaded networks over the edge
- low frame rates for workstation video codecs
- hands-free speaking in infancy (echo)
- packetization overhead, delay
- operating systems ill-suited for real-time applications ➠ single-user, no background load
- office environment: acoustics, lighting, …

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Characteristics of digital audio and video

	audio	video
rate (kb/s)	13…64…1500	200…1500…6000
loss tolerance	$\leq 5\%$	$10^{-5}\ldots 1\%$
packet size	small	large
traffic	interrupted CBR	VBR

One-way delay tolerance:

conference audio without echo cancellation: 40 ms

conference audio with echo cancellation: 150 ms

playback audio/video: $\geq$ 500 ms ("VCR response")

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Integration of Real-Time and Data Traffic

- TCP (and TP4, …) not suited to carry real-time traffic:
  - flow control ➠ window backoff and slow start
  - retransmission delay
- TCP backoff nice for making room for real-time traffic
- options:
  - low load, rely on data traffic backoff (but NFS doesn't)
  - priorities at MAC/link level (starvation!)
  - bandwidth-allocating scheduling at link level
- transport protocols can't help (except with set-up)!

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Integration of Real-Time and Data Traffic

- single-priority token rings cannot guarantee bounds (token loss), but may be sufficient
- MAC priorities (token ring, FDDI) won't help (much)
- priorities often not even implemented
- ATM: promises, but (most) switches don't implement traffic control

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Resource Reservation

- only makes sense with differential charging (or administrative controls)
- reserve resources ➠ preferential treatment for packets
- could have many reservation protocols
- sender or receiver oriented

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RSVP

receiver-oriented reservation protocol being standardized by IETF:

- multicast (and unicast)
- source sends PATH messages to receivers: path, max. flowspec
- receivers send RESV messages back to senders
- reservations are merged for same sender (max.)
- reservations *may* get merged between senders (audio!)

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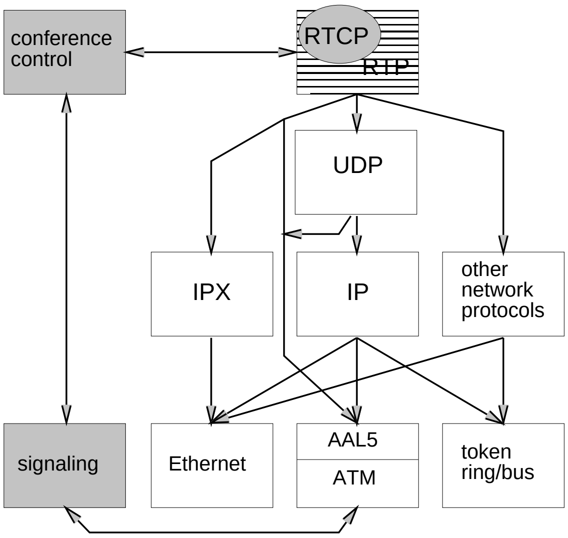
RTP: The Big Picture



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RTP: Real-Time Transport Protocol

- only part of puzzle
- product of Internet Engineering Task Force, AVT WG
- final draft stages
- under consideration by Interactive Multimedia Association (IMA)
- support for functions, but does not restrict implementation

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: Bridges, Translators, ...

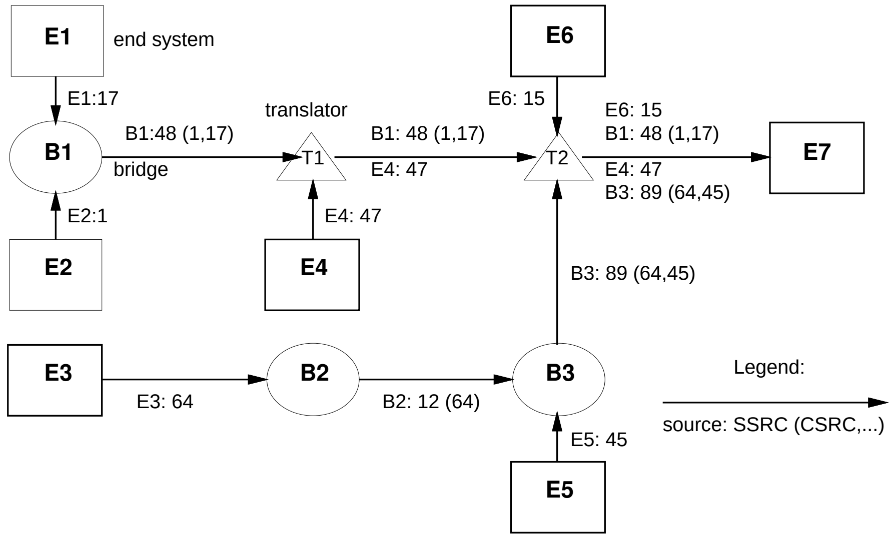
bridge:

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

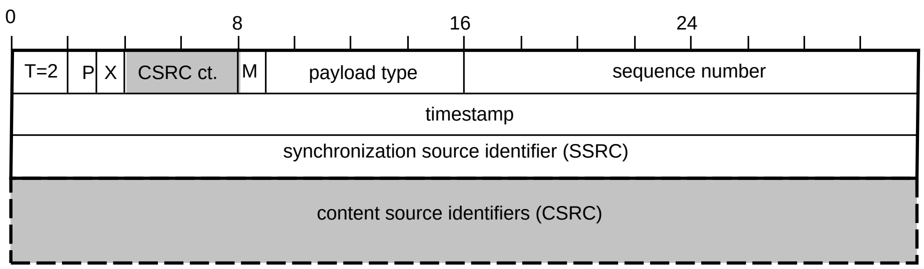
translator:

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

RTP: Bridges, 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 $\downarrow$ as members $\uparrow$

- limited by tolerable age of status
- give 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

format mapping: dynamic definition of media formats

extensions: application-specific

RTP: Control Protocol — SR

Bit positions: 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				
for each site received	synchronization source identifier of remote source				
for each site received	cumulative number of packets received				
for each site received	cumulative number of packets expected				
for each site received	interarrival jitter				
for each site received	last SR NTP timestamp seen from this source				
for each site received	delay since receiving that SR NTP timestamp				

RTP Implementations

"Adaptive applications" ➠ adjust to current network delay

tool	where	application
nv	Xerox PARC	Unix video (proprietary video encoding)
NeVoT	GMD Fokus	Unix audio
ivs	INRIA	Unix audio/video (H.261)
vic	LBL	Unix video

more in progress (PC, Mac, …)

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

Conclusion

- current LANs/WANs can support small number of audio/video connections
- need range of approaches:
  - adaptive applications
  - signaling
  - resource reservation
  - transport protocols
  - switch and router support for QoS
- charging for shared reservations?