Voice over IP: Issues and Protocols

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- Key Developments and Issues
- Voice/Multimedia over LANs
- Voice/Multimedia over IP
- Quality of Service over IP
- IP Multicasting
- Voice over ATM

Key Developments and Issues

- Voice over IP: Why?
- Sample Products and Services
- 13 Technical Issues
- 4 Other Issues
- Protocols
- H.323 Standard

Voice/Multimedia over LANs

- IEEE 802.1p standard on traffic classes in LANs and Dynamic multicast
- Generic Attribute Registration Protocol (GARP)
- Virtual LANs

Multimedia over IP

- Real-time Transport Protocol: RTP, RTCP
- Real-Time Streaming Protocol: RTSP
- Session Initiation Protocol (SIP)
- Session Description Protocol (SDP)

Quality of Service over IP

- Integrated services
- Resource Reservation Protocol: RSVP
- Differentiated Services
- QoS routing
- Multiprotocol Label Switching (MPLS) CoS

Multicasting over IP

- Multicast addressing and registration: IGMP
- Multicast Backbone: Mbone
- Multicast Routing: MOSPF, PIM

Voice and Telephony over ATM

- VTOA: Protocol Stack and Services
- ATM Adaptations Layers: AAL1, AAL5
- New AAL2

Schedule (Tentative)

- 9:00-9:15 Tutorial Overview
- 9:15-10:30 Voice over IP: Key Issues
- 10:30-10:45 *Coffee Break*
- 10:45-11:15 Voice over IP: Key Issues
- 11:15-12:00 Voice/Multimedia over LANs
- 12:00-1:00 *Lunch*
- 1:00-1:45 Voice/Multimedia over IP
- 1:45-2:00 *Stretch Break*
- 2:00-3:30 QoS over IP
- 3:30-3:45 *Coffee Break*
- 3:45-4:15 Multicasting over IP
- 4:15-4:45 Voice/Telephony over ATM
- 4:45-5:00 Final Review

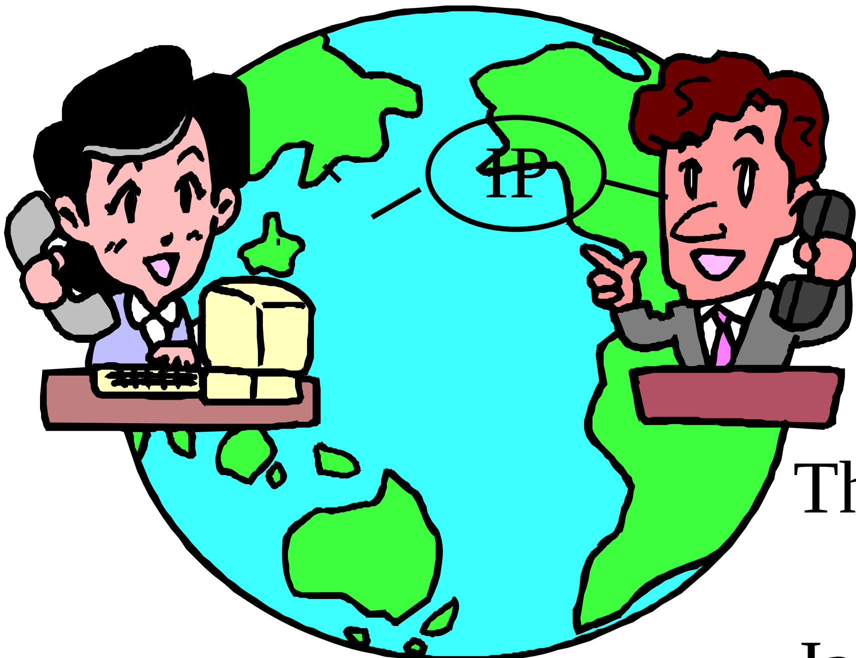
Disclaimer

- The technologies are currently evolving.
  $\Rightarrow$ Many statements are subject to change.
- Features not in a technology may be implemented later in that technology.
- Problems claimed to be in a technology may later not be a problem.

References

- You can get to all on-line references via: http://www.cis.ohio-state.edu/~jain/refs/lcn98ref.htm

Voice over IP: Key Developments and Issues



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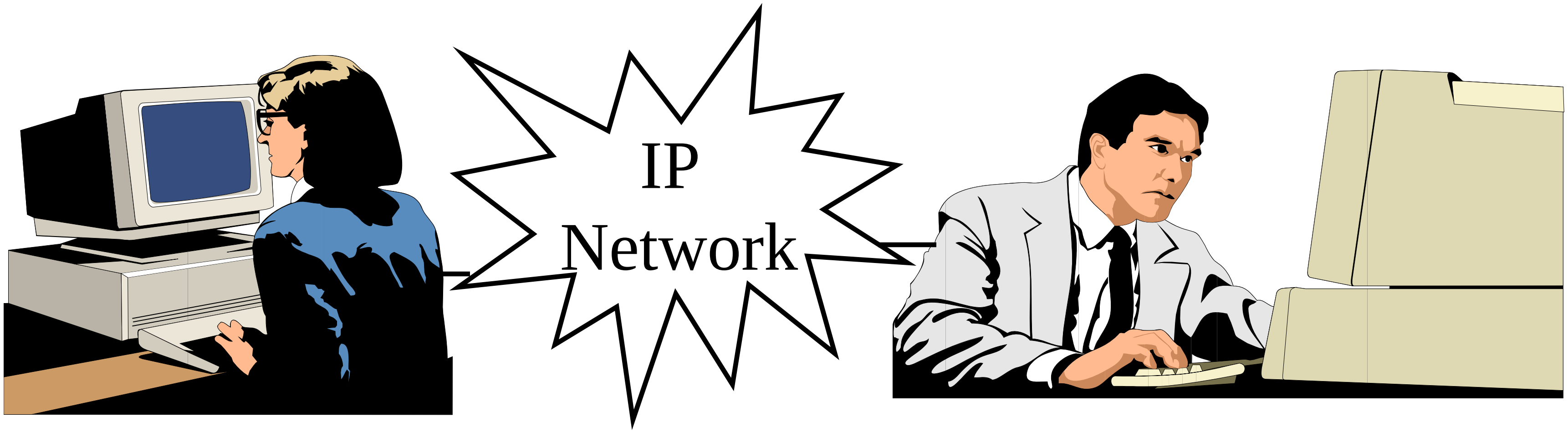
- Sample Products and Services
- 13 Technical Issues
- 4 Other Issues
- Protocols
- H.323 Standard

Market

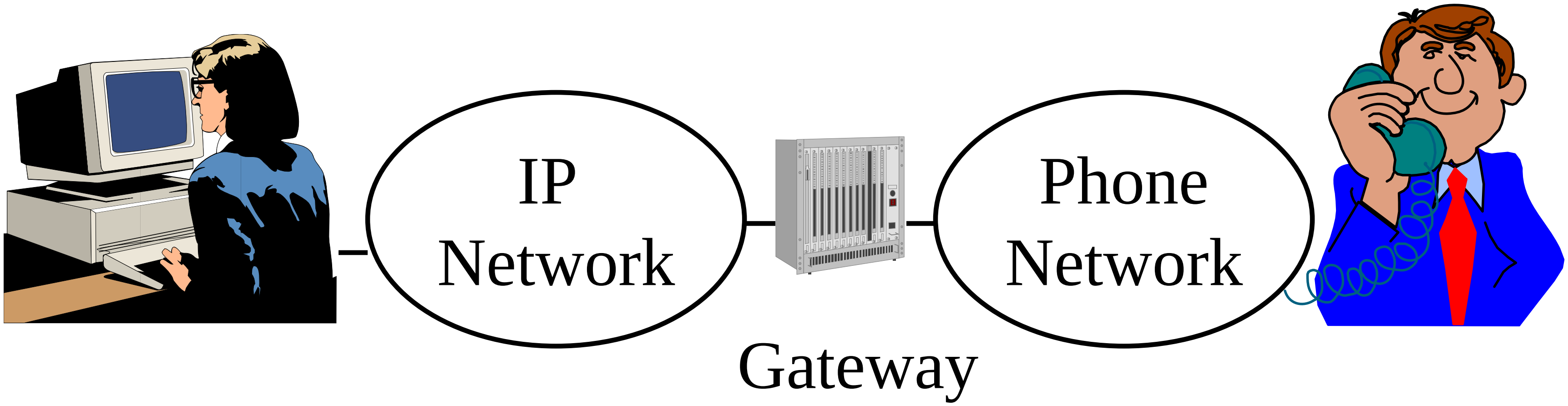
- International VOIP calls could cost 1/5th of normal rates ⇒ Big share of $18B US to foreign calls. $15B within Europe.
- 500,000 IP telephony users at the end of 1995.
- 15% of all voice calls on IP/Internet by 2000 ⇒ 10M users and $500M in VOIP product sales in 1999 [IDC]
- US VOIP service will grow from $30M in 1998 to $2B in 2004 [Forester Research]
  $2B in 2001 and $16B by 2004 [Frost & Sullivan]

Scenario 1: PC to PC



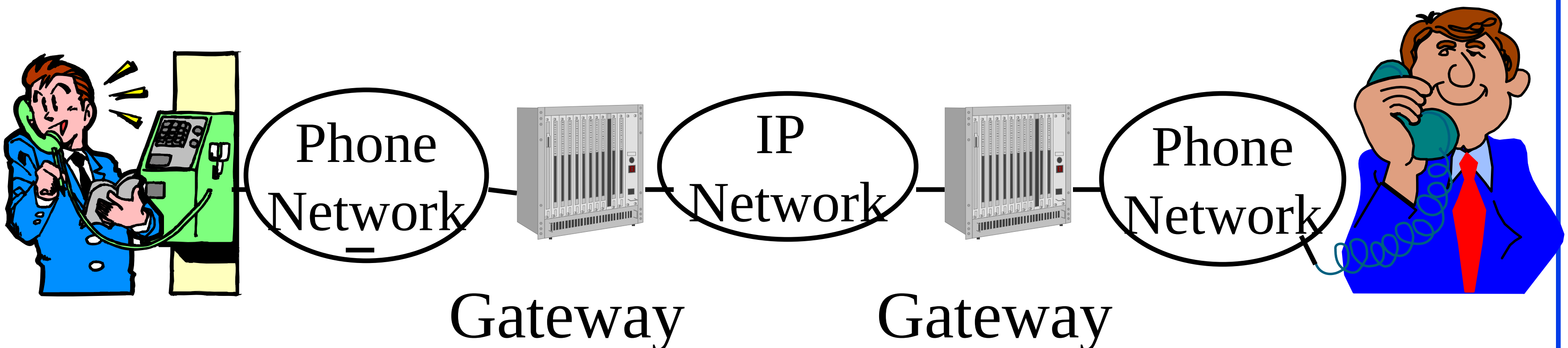
- Need a PC with sound card
- IP Telephony software: CUSeeMe, Internet Phone, ...
- Video optional

Scenario 2: PC to Phone



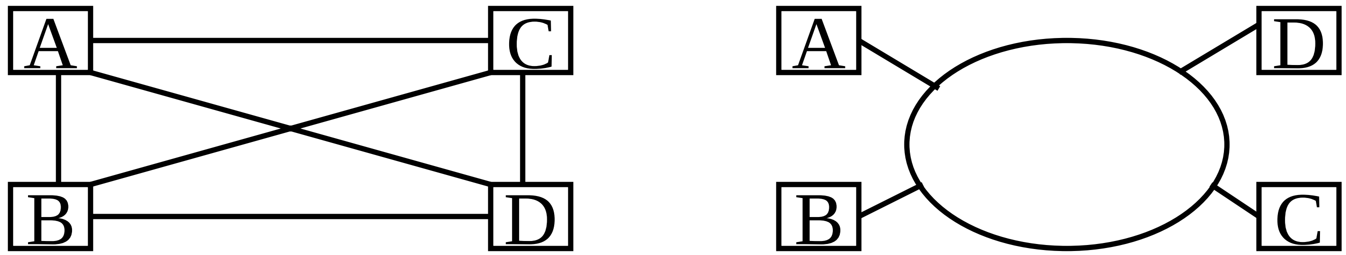
- Need a gateway that connects IP network to phone network (Router to PBX)

Scenario 3: Phone to Phone



- Need more gateways that connect IP network to phone networks
- The IP network could be dedicated intra-net or the Internet.
- The phone networks could be intra-company PBXs or the carrier switches

Advantages



- Voice has per-minute distance sensitive charge
  Data has flat time-insensitive distance-insensitive charge
- Private voice networks require n(n-1)/2 access links.
  Shared Internet requires only n access links.
- Easy alternate routing ⇒ More reliability
- No 64kbps bandwidth limitation
  ⇒ Easy to provide high-fidelity voice

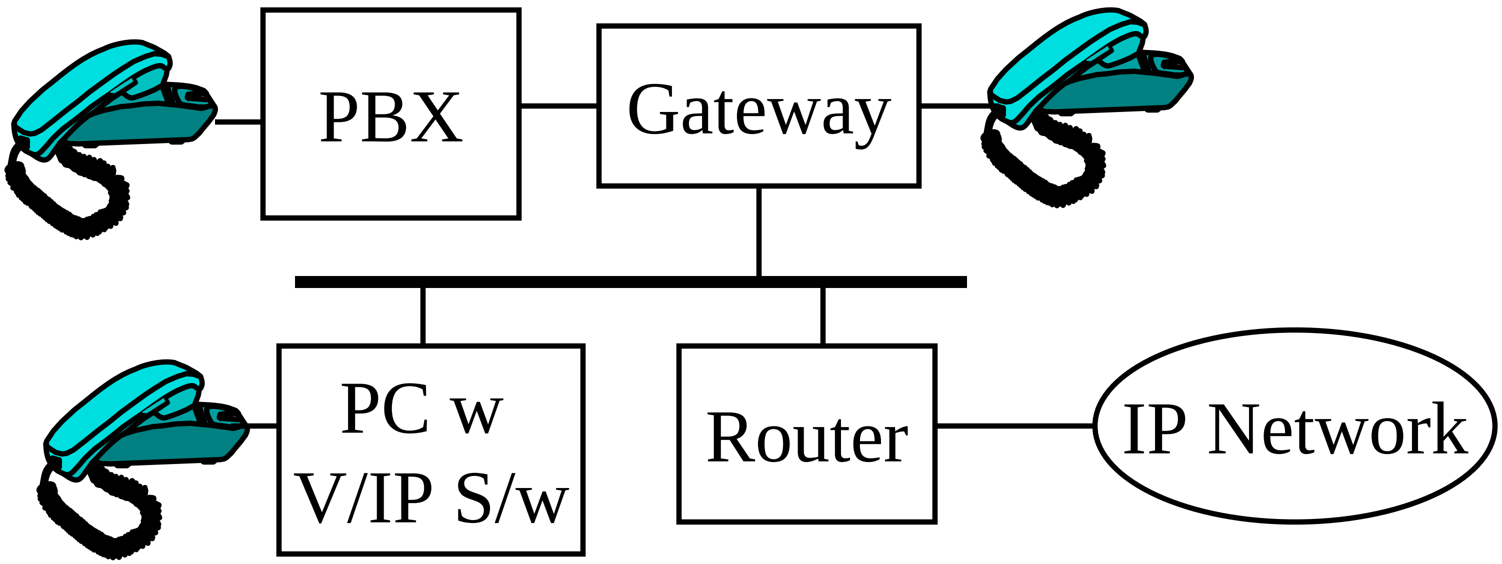
Applications

- Any voice communication where PC is already used:
  - Document conferencing
  - Helpdesk access
  - On-line order placement
- International callbacks
  (many operators use voice over frame relay)
- Intranet telephony
- Internet fax

Sample Products

- VocalTec Internet Phone: PC to PC.
- Microsoft NetMeeting: PC to PC. Free.
- Internet PhoneJACK: ISA card to connect a standard phone to PC. Works with NetMeeting, InternetPhone etc. Provides compression.
- Internet LineJACK: Single-line gateway.
- Micom V/IP Family:
  - Analog and digital voice interface cards
  - PC and/or gateway

Products (Cont)



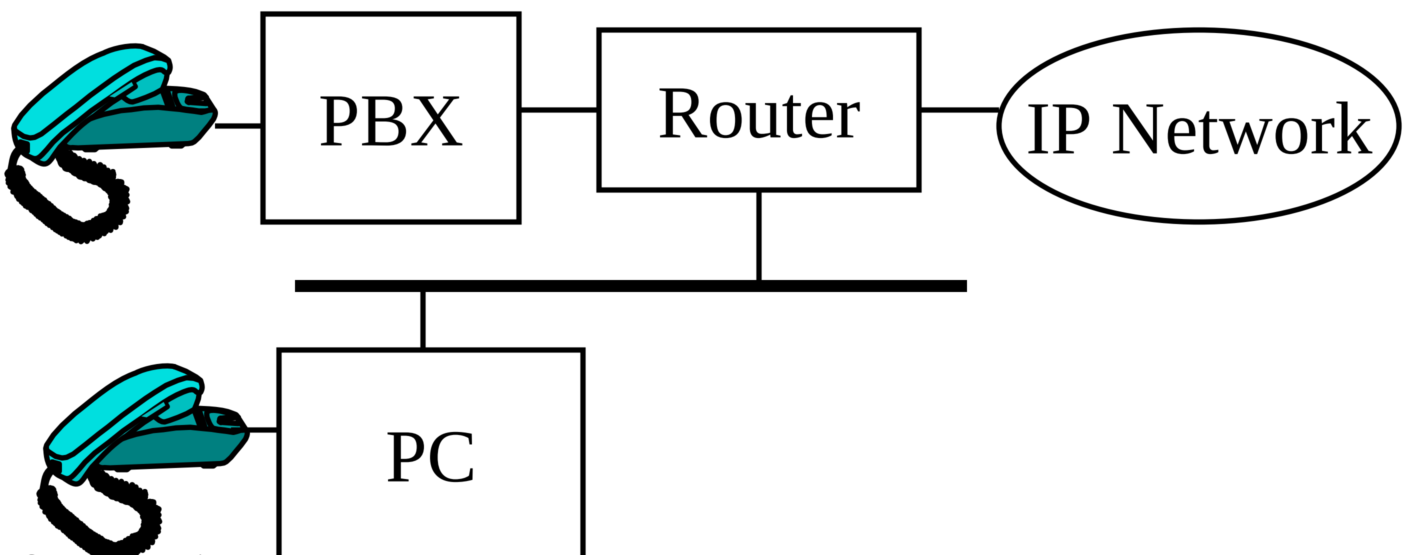
- Features:
  - Compression
  - Phone number to IP address translation.
  - Supports RSVP.
  - Limits number of calls.

Products (Cont)

- VocalTec Internet Telephony Gateway:
  - Similar to Micom V/IP
  - Interactive voice response system for problem reporting
  - Allows WWW plug in
  - Can monitor other gateways and use alternate routes including PSTN
  - Sold to Telecom Finland. New Zealand Telecom.
- Lucent's Internet Telephony Server: Gateway
  Lucent PathStar Access Server

Products (Cont)

- CISCO 2600 Routers: Voice interface cards (VICs) Reduces one hop.
- Baynetworks, 3COM, and other router vendors have announced product plans



Sample Services

- IDT Corporation offers Net2Phone, Carrier2Phone, Phone2Phone services.
- Global Exchange Carrier offers international calls using VocalTec InternetPhone s/w and gateways
- Quest offers 7.5¢/min VOIP Q.talk service in 16 cities.
- ITXC provides infrastructure and management to 'Internet Telephone Service Providers (ITSPs)'
- America On-line offers 9¢/min service.
- AT&T announced 7.5¢/min VOIP trials in 9 US cities.

Services (Cont)

- Other trials: USA Global link, Delta 3, WorldCom, MCI, U.S. West, Bell Atlantic, Sprint, AT&T/Japan, KDD/Japan, Dacom/Korea, Deutsche Telekom in Germany, France Telecom, Telecom Finland, and New Zealand Telecom.
- Level 3 is building a nation wide IP network for telephony.
- Bell Canada has formed 'Emergis' division.
- Bellcore has formed 'Soliant Internet Systems' unit
- Bell Labs has formed 'Elemedia' division

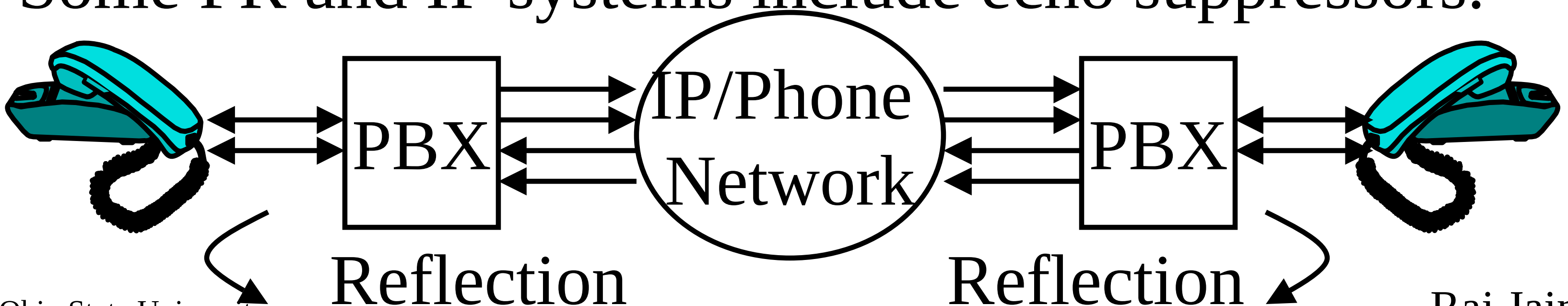
Technical Issues

1. Large Delay

- Normal Phone: 10 ms/kmile
  ⇒ 30 ms coast-to-coast
- G.729: 10 ms to serialize the frame + 5 ms look ahead + 10 ms computation = 25 ms one way algorithmic delay
- G.723.1 = 100 ms one-way algorithmic delay
- Jitter buffer = 40-60 ms
- Poor implementations ⇒ 400 ms in the PC
- In a survey, 77% users found delay unacceptable.

Technical Issues (Cont)

2. Delay Jitter: Need priority for voice packets. Shorter packets? IP precedence (TOS) field.
3. Frame length: 9 kB at 64 kbps = 1.125 s Smaller MTU ⇒ Fragment large packets
4. Lost Packets: Replace lost packets by silence, extrapolate previous waveform
5. Echo cancellation: 2-wire to 4-wire. Some FR and IP systems include echo suppressors.



Technical Issues (Cont)

6. Silence suppression
7. Address translation: Phone # to IP. Directory servers.
8. Telephony signaling: Different PBXs may use different signaling methods.
9. Bandwidth Reservations: Need RSVP.
10. Multiplexing: Subchannel multiplexing
    $\Rightarrow$ Multiple voice calls in one packet.
11. Security: Firewalls may not allow incoming IP traffic
12. Insecurity of internet
13. Voice compression: Load reduction

Other Issues

1. Per-minute distance-sensitive charge vs
   flat time-insensitive distance-insensitive charge
2. Video requires a bulk of bits but costs little.
   Voice is expensive. On IP, bits are bits.
3. National regulations and government monopolies
   $\Rightarrow$ Many countries forbid voice over IP
   In Hungary, Portugal, etc., it is illegal to access a web site with VOIP s/w. In USA, Association of Telecommunications Carriers (ACTA) petitioned FCC to levy universal access charges in ISPs
4. Modem traffic can't get more than 2400 bps.

Compression Standards

- G.711: 64 kbps Pulse Code Modulation (PCM)
- G.721:
  - 32 kbps Adaptive Differential PCM (ADPCM).
  - Difference between actual and predicted sample.
  - Used on international circuits
- G.728: 16 kbps Code Excited Linear Prediction (CELP).
- G.729: 8 kbps Conjugate-Structure Algebraic Code Excited Linear Prediction (CS-ACELP).

Compression (Cont)

- G.729A:
  - A reduced complexity version in Annex A of G.729.
  - Supported by AT&T, Lucent, NTT.
  - Used in simultaneous voice and data (SVD) modems.
  - Used in Voice over Frame Relay (VFRADs).
  - 4 kbps with proprietary silence suppression.

Compression (Cont)

- G.723.1: Dual rates (5.3 and 6.3 kbps).
  - Packet loss tolerant.
  - Silence suppression option.
  - Recommended by International Multimedia Teleconferencing Consortium (IMTC)'s VOIP forum as default for H.323.
  - Supported by Microsoft, Intel.
  - Mean opinion score (MOS) of 3.8.
    4.0 = Toll quality.

Protocols

- Multimedia over LANs:
  - Priority and Dynamic Multicast (IEEE 802.1p)
  - Virtual LAN (IEEE 802.1Q)
- Audio-Visual Transport over IP:
  - RTP: Real-time Transport Protocol.
    Sequencing, timestamp, payload identification, and delivery monitoring. [RFC 1889]
  - RTCP: RTP Control Protocol. Provides delivery feedback.
  - RTSP: Real-time Streaming Protocol. Allows controlling streaming audio/video. [RFC 2326]

Protocols (Cont)

- Multi-party Multimedia Control:
  - SIP: Session Initiation Protocol [IETF mmusic]
  - SDP: Session Description Protocol [RFC 2327]
  - SAP: Session Announcement Protocol [IETF mmusic]
  - SCCP: Simple Conference Control Protocol [IETF mmusic]

Protocols (Cont)

- Quality of Service over IP:
  - Integrated Services: Guaranteed (CBR) and controlled-load (nrt-VBR) services. [RFC 2211+2212]
  - RSVP: Resource Reservation protocol [RFC 2205]
  - IP over ATM: MPOA allows QoS.
  - MPLS: Multiprotocol Label Switching. Will support QoS. [IETF mpls]
  - ST-II: Stream Protocol V2. Connection oriented IP. IPv5. Provides resource reservations. [RFC 1819]

Protocols (Cont)

- Multicasting:
    - Internet Group Management Protocol (IGMP)
- Multicast Routing:
    - MOSPF
    - DVMRP
    - PIM

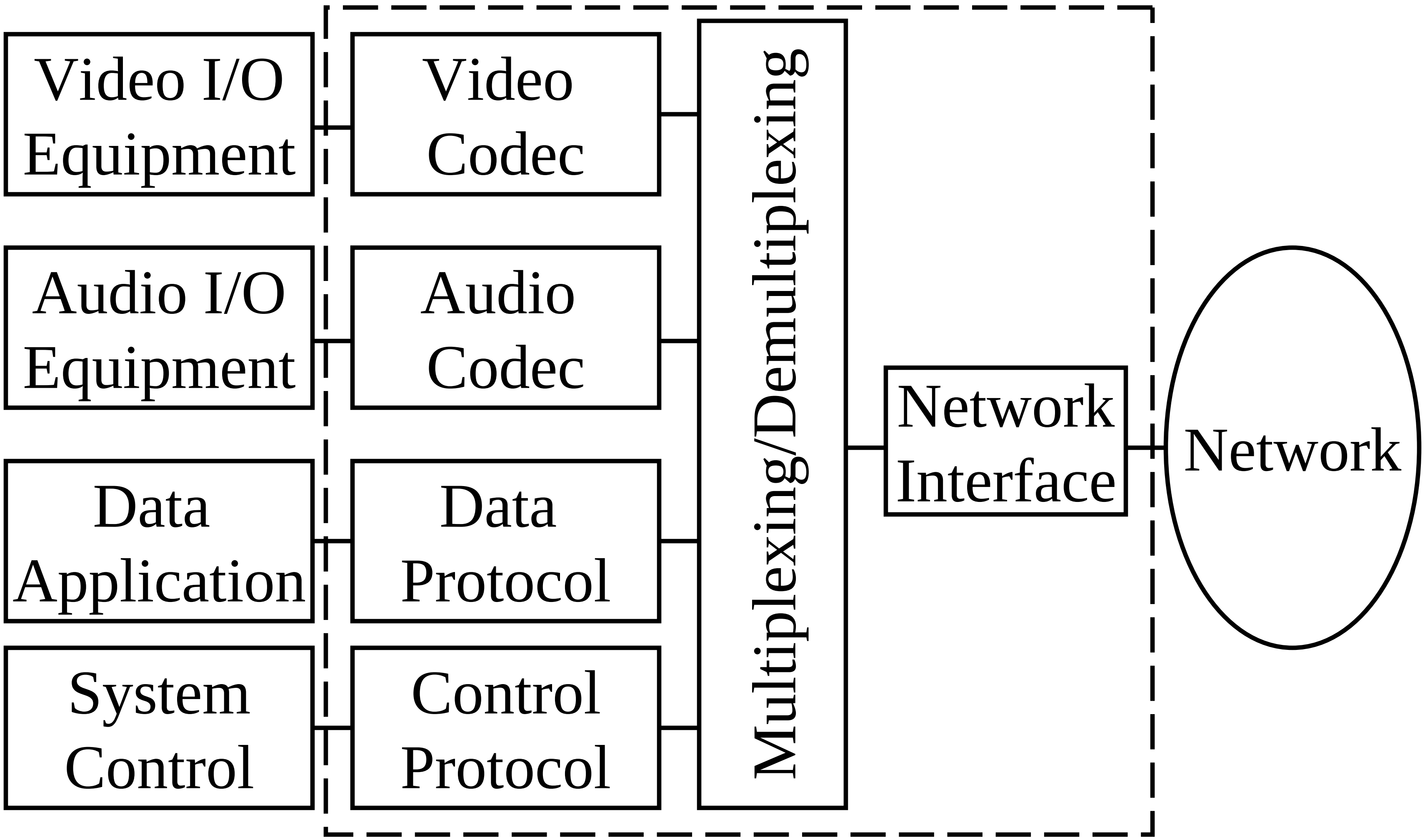
Protocols (Cont)

- VPIM: Voice Profile for Internet Mail [RFC 1911]. Voice messages in email. MIME profile.
- SCTP: Simple Computer Telephony Protocol. Like SMTP for mail. [www.phonezone.com]
- S.100: Standard application programming interface for computer telephony.
  - Endorsed by the Enterprise Computer Telephony Forum.
  - Will allow applications from different vendors to share the telephony server resources for switching, routing, and media processing.

Protocols (Cont)

- SCbus: High-speed TDM bus for computer telephony.
  - Endorsed by ANSI.
  - Telephony products from Micom, VocalTec, IDT are all based on SCbus.
- Internet Fax Routing standard: Allows routing communication among fax servers.
- H.323 Internet telephony (video conferencing) standard

Telephony/Conferencing Systems



Conferencing Standards

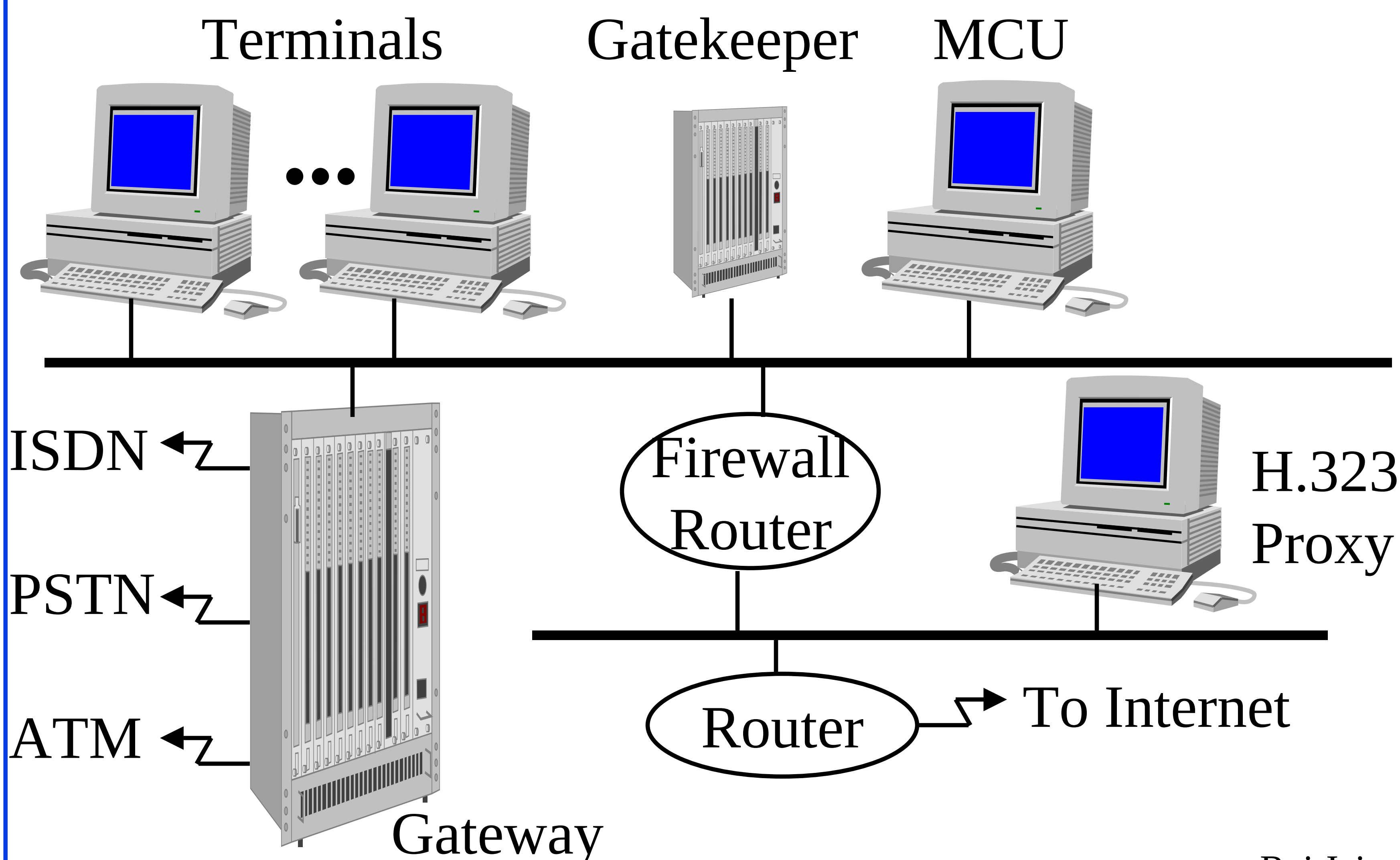
Network	ISDN	ATM	PSTN	LAN	POTs
Conf. Std.	H.320	H.321	H.322	H.323 V1/V2	H.324
Year Audio Codec	1990 G.711, G.722, G.728	1995 G.711, G.722, G.728	1995 G.711, G.722, G.728	1996/1998 G.711, G.722, G.723.1, G.728, G.729	1996 G.723.1, G.729
Audio Rates kbps Video Codec	64, 48-64 H.261	64, 48-64, 16 H.261, H.263	64, 48-64, 16 H.261, H.263	64, 48-64, 16, 8, 5.3/6.3 H.261 H.263	8, 5.3/6.3 H.261 H.263
Data Sharing	T.120	T.120	T.120	T.120	T.120
Control	H.230, H.242	H.242	H.242, H.230	H.245	H.245
Multiplexing	H.221	H.221	H.221	H.225.0	H.223
Signaling	Q.931	Q.2931	Q.931	Q.931	-

H.323 Protocols

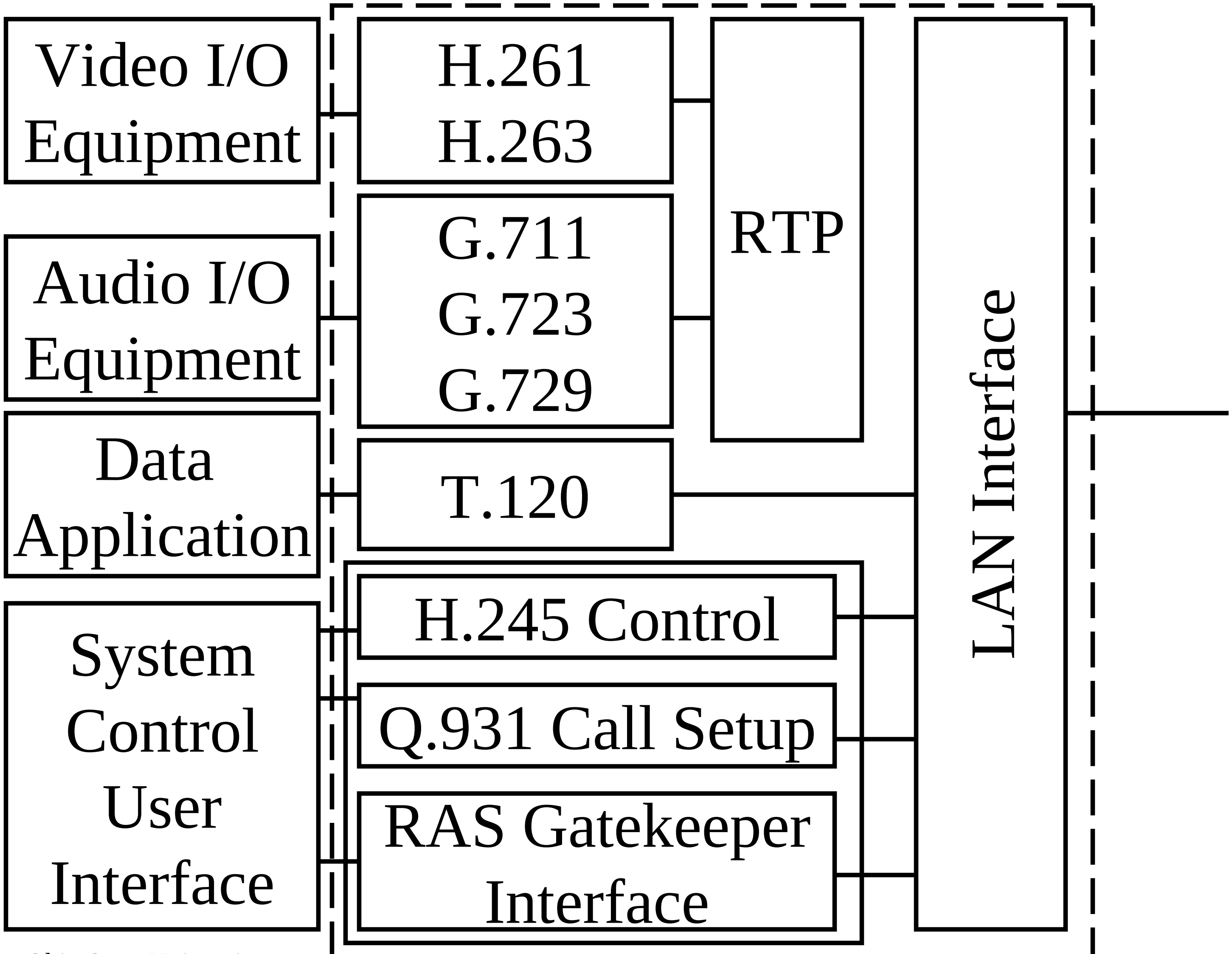
- Multimedia over LANs
- Provides component descriptions, signaling procedures, call control, system control, audio/video codecs, data protocols

<table>
<tr><th>Video</th><th>Audio</th><th colspan="4">Control and Management</th><th>Data</th></tr>
<tr><td>H.261
H.263</td><td>G.711, G.722,
G.723.1, G.728,
G.729</td><td rowspan="2">RTCP</td><td>H.225.0
RAS</td><td>H.225.0
Signaling</td><td>H.245
Control</td><td>T.124</td></tr>
<tr><td colspan="2">RTP</td><td colspan="3">X.224 Class 0</td><td>T.125</td></tr>
<tr><td colspan="3">UDP</td><td colspan="3">TCP</td><td rowspan="2">T.123</td></tr>
<tr><td colspan="6">Network (IP)</td></tr>
<tr><td colspan="7">Datalink (IEEE 802.3)</td></tr>
</table>

H.323 Components



H.323 Terminals

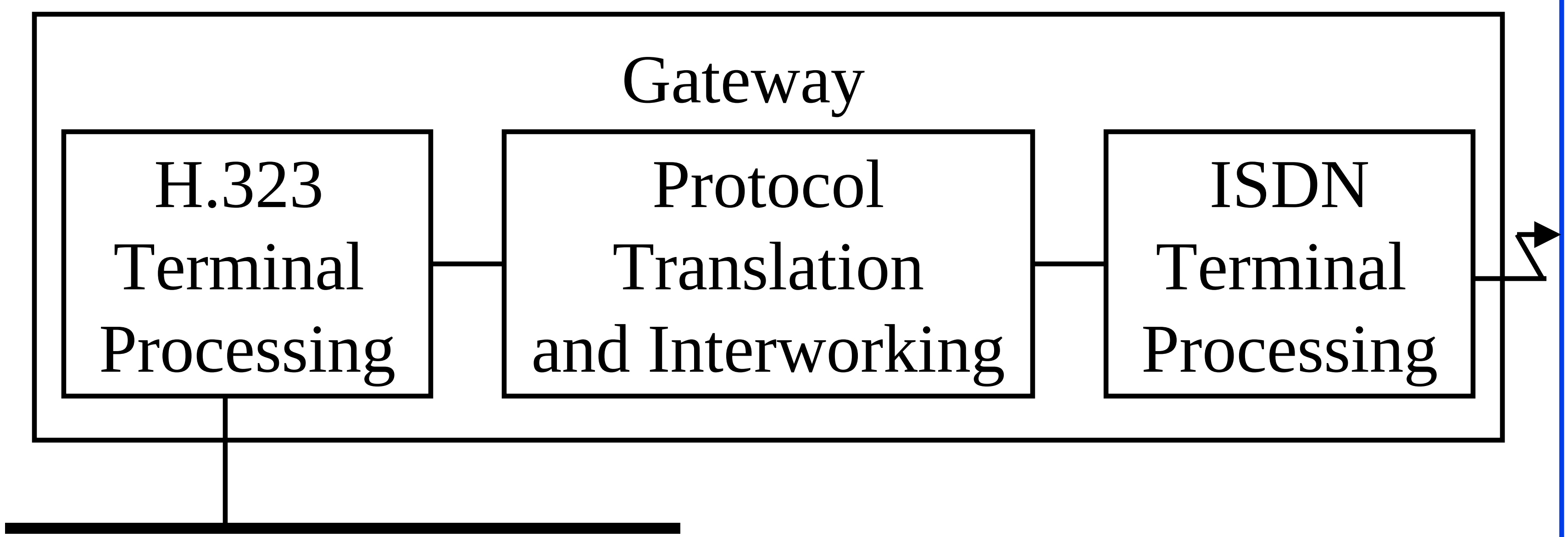


H.323 Terminals

- Client end points. PCs.
- H.245 to negotiate channel usage and capabilities.
- Q.931 for call signaling and call setup.
- Registration/Admission/Status (RAS) protocol to communicate with gatekeepers.
- RTP/RTCP for sequencing audio and video packets.

H.323 Gateways

- Provide translation between H.323 and other terminal types (PSTN, ISDN, H.324)
- Not required for communication with H.323 terminals on the same LAN.



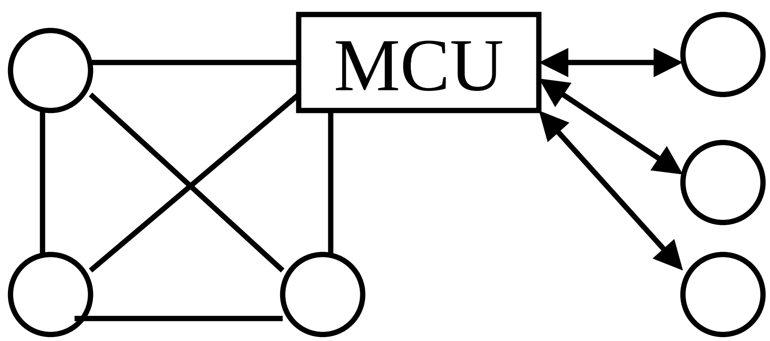
H.323 Gatekeepers

- Provide call control services to registered end points.
- One gatekeeper can serve multiple LANs
- Address translation (LAN-IP)
- Admission Control: Authorization
- Bandwidth management
  (Limit number of calls on the LAN)
- Zone Management: Serve all registered users within its zone of control
- Forward unanswered calls
- May optionally handle Q.931 call control

H.323 MCUs

- Multipoint Control Units
- Support multipoint conferences
- Multipoint controller (MC) determines common capabilities.
- Multipoint processor (MP) mixes, switches, processes media streams.
- MP is optional. Terminals multicast if no MP.

Multicast



Unicast

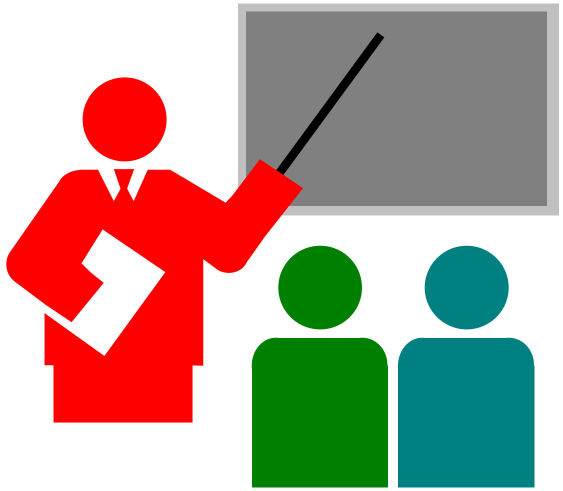
H.323 V2

- Approved in January 1998.
- Enhanced security: H.235 standard for authentication, integrity, privacy, and non-repudiation.
- Fast call setup: No delay after answering the call
- Supplementary services: Call transfer, Call diversion (forwarding)
- T.120/H.323 Integration: Clarified.

Voice Support in Routers

- Smart Queueing:
  - Priority queueing,
  - Weighted fair queueing (WFQ),
  - Weighted Random Early Detection (WRED)
- Traffic Shaping
- Resource Reservation: RSVP

Summary



- Voice over IP products and services are being rolled out
- Ideal for computer-based communications
- IP needs QoS for acceptable quality
- A number of working group at IETF are working on it
- H.323 provides interoperability

Organizations

- IMTC: International Multimedia Teleconferencing Consortium
- VOIP: Founded in 1996 by Cisco, Microsoft, VocalTec, 3Com/USR, Dialogic etc. to augment H.323. Folded into IMTC.
- Enterprise Computer Telephony Forum (ECTF), http://www.ectf.org
- Internet Fax Routing Forum
- Voice Profile for Internet Mail (VPIM) Work Group of EMA
- VON Coalition, Inc., http://www.von.org

IETF Working Groups

Multimedia:

- IP Telephony (iptel): RFC 1789 on INETPhone servers. Will develop Gateway attribute distribution protocol and call processing syntax.
- Internet Fax (fax): Data representation, addressing, and transport of faxes over IP.
- PSTN and Internet Interworking (pint):
  - Initiation of telephone services from IP hosts.
  - Web users can request call back, fax, fax-back services.
  - Phone users can request web pages (via speech).

IETF (Cont)

- Audio/Video Transport (avt): Real-time transmission of audio and video over UDP and IP multicast. RTP, RTCP.
- Multiparty Multimedia Session Control (mmusic): Internet Teleconferencing. SDP, SAP, RTSP, SIP, SCCP.
- Also several working groups on multicasting and QoS

References

- See http://www.cis.ohio-state.edu/~jain/refs/ref_voip.htm for a detailed list of references.

Voice/Multimedia over LANs

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- IEEE 802.1p standard on traffic classes in LANs and Dynamic multicast
- Generic Attribute Registration Protocol (GARP)
- Virtual LANs

Traffic Classes in LANs

- IEEE 802.1p working group
- Goal: To support time-critical (continuous media) traffic
- Method:
  - 1. Prioritization of traffic
  - 2. Efficient support of multicasting
- Bridge filtering database for each port indicates whether any members of the group exist on the port $\Rightarrow$ Need Group registration protocol

What's in a Name?

- The "p" in 802.1p is lower case.
- Uppercase letter ⇒ Base standard
- Lowercase letter ⇒ Supplement
- 802.1p is a supplement to 802.1D bridge standard
- 802.1Q is a base VLAN standard
- 802.3z is a 1000 Mbps supplement to Ethernet Standard

IEEE 802.1p: Features

- Allows up to 8 traffic classes (priorities)
- Priority $\Rightarrow$ Both queueing and access
- Allows queueing priority on LANs that have no access priorities, e.g., Ethernet
- Different number of priorities on different ports
- Allows dynamic multicast filtering
- Applies to all 802 MAC protocols + FDDI
- **802 MAC Protocols**: 802.3 (Ethernet), 802.4 (Token Bus), 802.5 (Token Ring), 802.6 (DQDB), 802.9 (Integrated Services), 802.12 (Demand Priority)

Number of Priorities

- Up to 8 traffic classes (0 through n-1).
  0 = Normal service = Low priority.
- Different ports/bridges may have different number of traffic classes
  $\Rightarrow$ Low-speed ports need priorities first
- Recommended four priorities:
  - Time and safety critical
  - Time critical
  - Non-time critical, loss sensitive
  - Non-time critical, loss insensitive

How is Priority Set?

- Priority may be set by user, destination address, input port, output port, access priority, or by VLAN
- A priority may be assigned for a port
  $\Rightarrow$ For a source station connected to a switch
- In some LANs, priority can be encoded in frames.
- In some LANs, priority cannot be encoded in frames. 802.1p does not have a mechanism to communicate priority in such LANs.
  - It has to be regenerated locally using local database, or use 802.1Q VLAN tags.

Multicast: Today

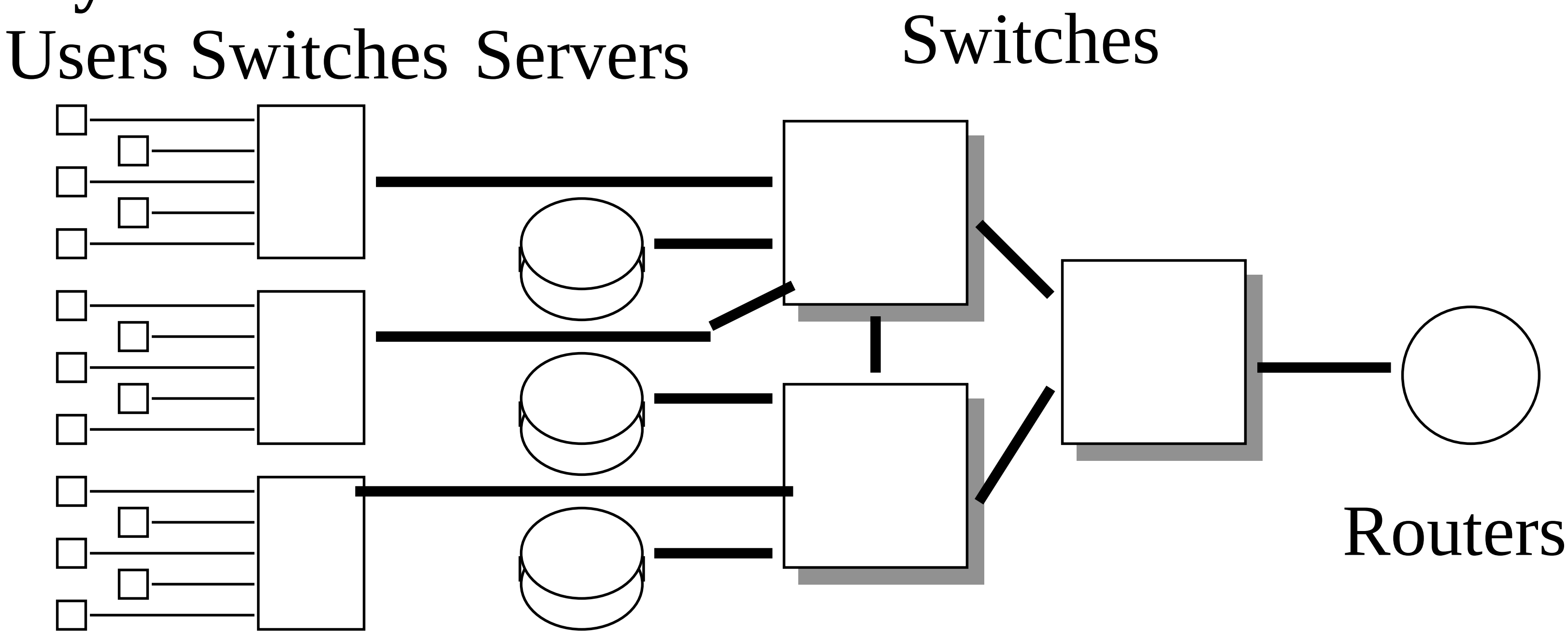
- Bridges forward multicast on all active ports
- A spanning tree is formed to avoid loops

Dynamic Multicast Filtering

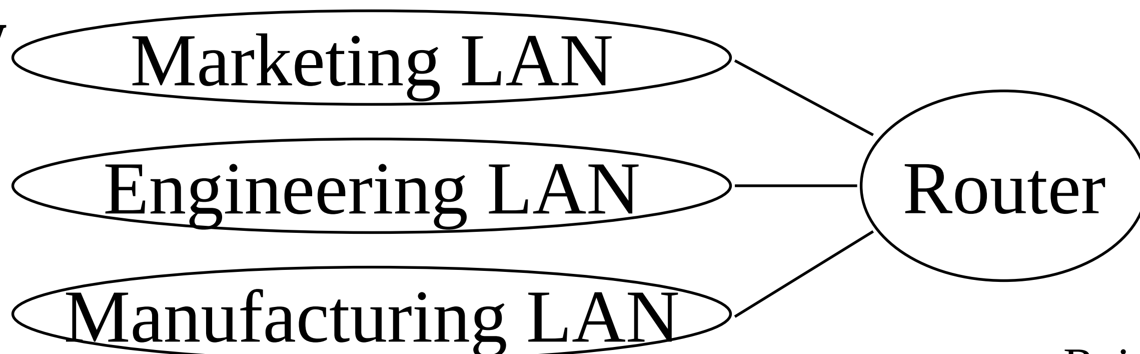
- Send multicast frames only on LANs where receivers exist
- Multicast address registration: Join/leave a group
- Legacy multicast addresses: Unregistered
- Join/leave “all groups” (Used on legacy segments)
- Join/leave “all unregistered groups” (For coexistence of legacy and new stations during migration.)
- Static entries can exclude some multicast addresses from "all groups"
- Membership information is forwarded to other bridges

What is a Virtual LAN

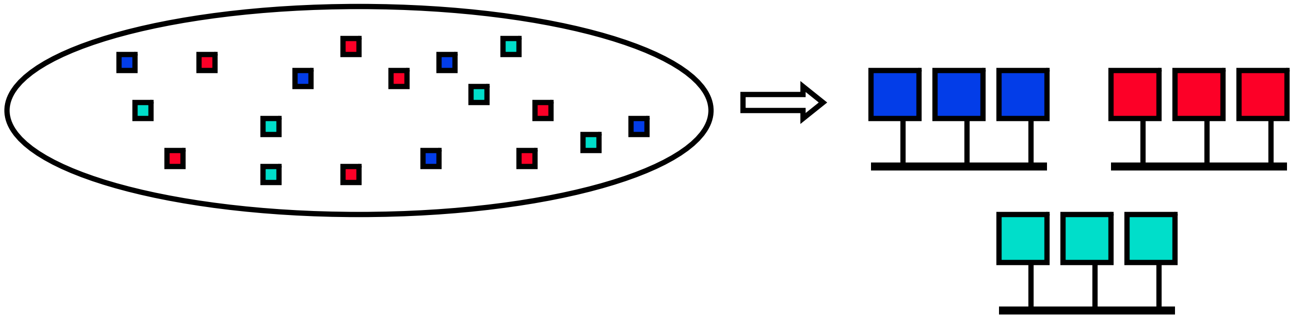
- **Physical View**



- **Logical View**



Virtual LAN

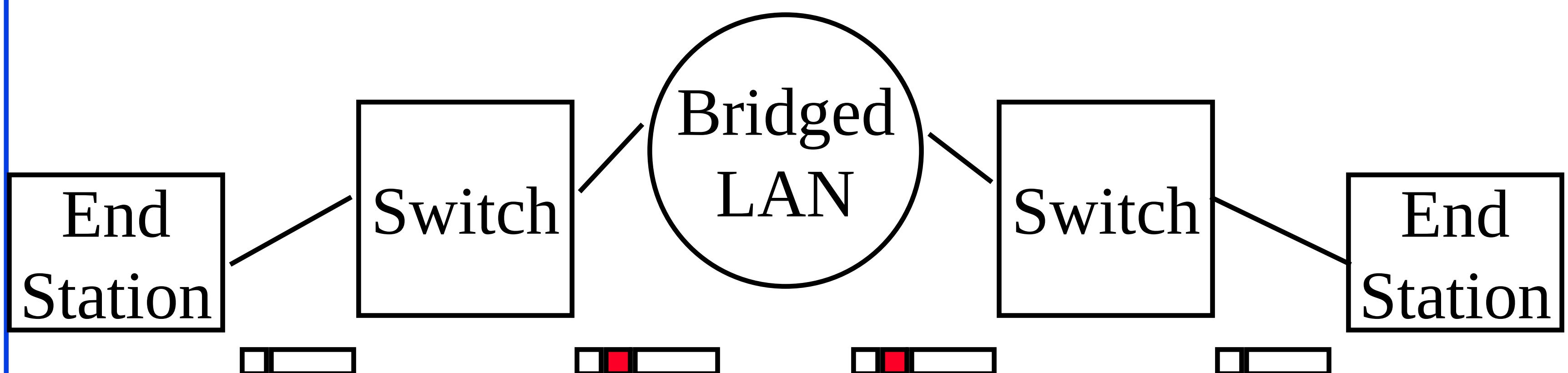


- Virtual LAN = Broadcasts and multicast goes only to the nodes in the virtual LAN
- LAN membership defined by the network manager $\Rightarrow$ Virtual

VLAN Tagging

Dest. Addr	Src. Addr	VLAN Tag	Prot. Type

- First switch adds tag containing VLAN id to all incoming packets
- Intermediate switches do not recompute the VLAN id
- Last switch removes tags from all outgoing packets
- Tag is not swapped at every hop like VC Id or labels



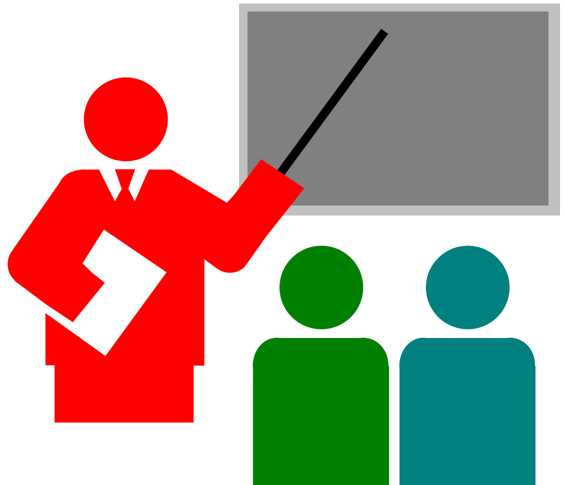
Synonyms

- Tag
- Label
- Mark
- Sticker
- Brand

IEEE 802.1Q: Features

- Upward compatible with existing hubs and bridges
- Extends 802.1p priority mechanism to priority based on VLAN membership
- Allows priority associated with each VLAN
- VLAN-based priority takes precedence over other priority considerations
- Allows signaling priority information on non-priority (CSMA/CD) LANs
- Operation with/without explicit VLAN tag: +4 bytes
  $\Rightarrow$ Reduce data field or increase max frame size.
  Both allowed.

Summary



- Traffic classes and dynamic multicast on LANs to allow multimedia
- IEEE 802.1p allows 8 priorities
- Distributed multicast registration protocol
- Virtual LANs $\Rightarrow$ Location independent LAN Groups
- IEEE 802.1Q allows both explicit and implicit tagging

References

- For a detailed list of references, see http://www.cis.ohio-state.edu/~jain/refs/lsw_refs.htm
- IEEE 802.1 Home page, http://grouper.ieee.org/groups/802/1/index.html
- IEEE 802.1D MAC Bridges: Revision, (Incorporating IEEE 802.1p)," IEEE P802.1D/D17, May 25, 1998, ftp://p8021:-go_wildcats@p8021.hep.net/8021/d-drafts/d17/fdis-15802-3.pdf

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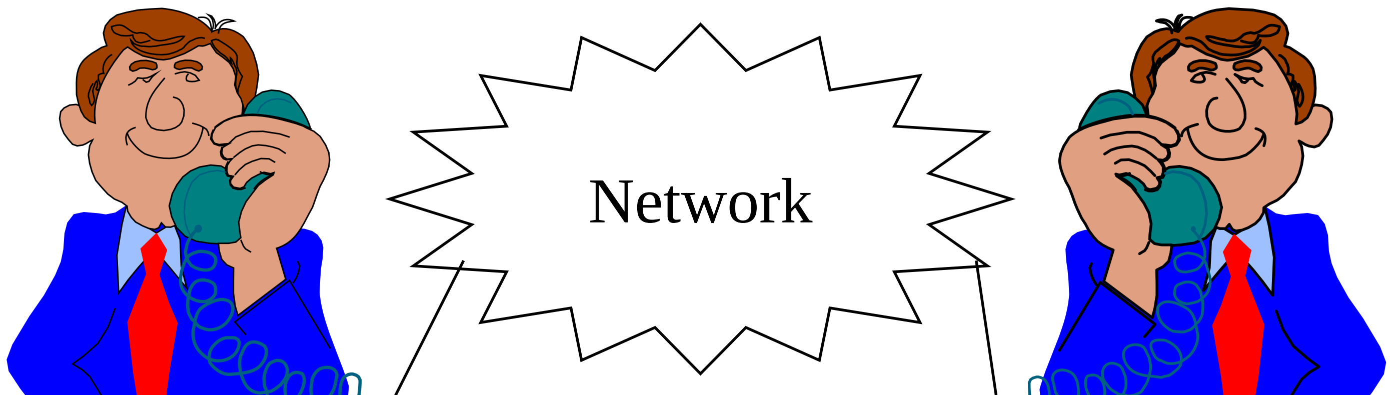
- Draft Standard for Virtual Local Area Networks, IEEE P802.1Q/D11, July 30, 1998, ftp://p8021:-go_wildcats@p8021.hep.net/8021/q-drafts/d11/q-d11.pdf

Protocols for Multimedia Over IP



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- Real-time Transport Protocol: RTP, RTCP
- Real-Time Streaming Protocol: RTSP
- Session Initiation Protocol (SIP)
- Session Description Protocol (SDP)

Multimedia

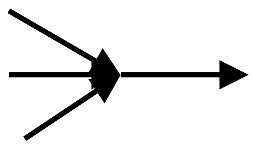
Application	Video Conferencing, Telephony, Fax
Transport	Timing Sync, payload id, error recovery
Network	QoS, Multicast, Signaling
Datalink	Access Control, Multicast, signaling
Physical	Multiple channels via SDM, FDM, TDM

IETF Groups

Application	Iptel, fax, pint
Transport	Avt (RTP), mmusic (RTSP)
Network	Qosr, MPLS, IntServ, Issl, Diff-serv, RSVP
Datalink	IEEE 802.1p
Physical	Broadband Ethernet 10Broad36

RTP

- Real-Time "Transport" Protocol
- Not really an L4 protocol.
  Common parts of several applications.
  Uses UDP for multiplexing and checksum.
- Supports unicast and multicast delivery
- Source and payload type identification
- Sequencing, Timing, and Synchronization
- Source merging: Multiple contributing sources for a combined stream produced by an RTP mixer.
  32-bit Synchronizing source (SSRC) id.



- Stream translation: High-speed to low speed

RTP (Cont)

- What RTP Does not Do?
  - Reliable data delivery
  - Quality of service guarantees
  - Resource reservations (RSVP)
  - Delivery of encryption key to participants
- RTP provides a general framework for applications to be able to do these $\Rightarrow$ Application Level Framing
- Two components: RTP and Control (RTCP) $\Rightarrow$ Simple RTP header
- Particular codings need additional parameters $\Rightarrow$ RTP Profiles documents

RTCP

- Real-Time Transport Control Protocol
- Convey information about participants
- Convey information about relationships among sessions
- Monitor application performance
  $\Rightarrow$ Feedback on quality of data
- Automatically adjusts overhead
  (Report frequency based on participant count)

RTCP Packet Types

- Sender Report (SR):
  Packets/bytes sent, lost
- Receiver Report (RR):
  Packets/bytes received, lost, jitter
- Source Description (SDES)
- End of participation (BYE)
- Application Specific functions (APP)

Audio Encodings

- L16: Linear 16-bit Uncompressed sampled at 48 kHz, 44.1 kHz, 22.05 kHz, 11.025 kHz
- PCMU: CCITT/ITU-T G.711 8 bits/sample with -law companding
- PCMA: CCITT/ITU-T G.711 8 bits/sample with A-law companding
- G721 through G729: CCITT/ITU-T Recommendations
- GSM: Group Speciale Mobile 13 kb/s

Audio Encodings (Cont)

- 1016: Federal standard FED-STD 1016 uses code-excited liner prediction (CELP)
- IDVI: Intel/DVI ADPCM recommended by Interactive Multimedia Association
- And more ...

Video Encodings

- CPV: Compressed packet video implemented by Concept, Bolter, and ViewPoint Systems
- JPEG: ISO Standard DIS 10918-1, and 10918-2.
- H261: CCITT/ITU-T standard
- nv: developed by Ron Frederick at Xerox
- CUSM: Used in CUSeeMe
- PicW: Used in PictureWindow developed at BBN

RTSP

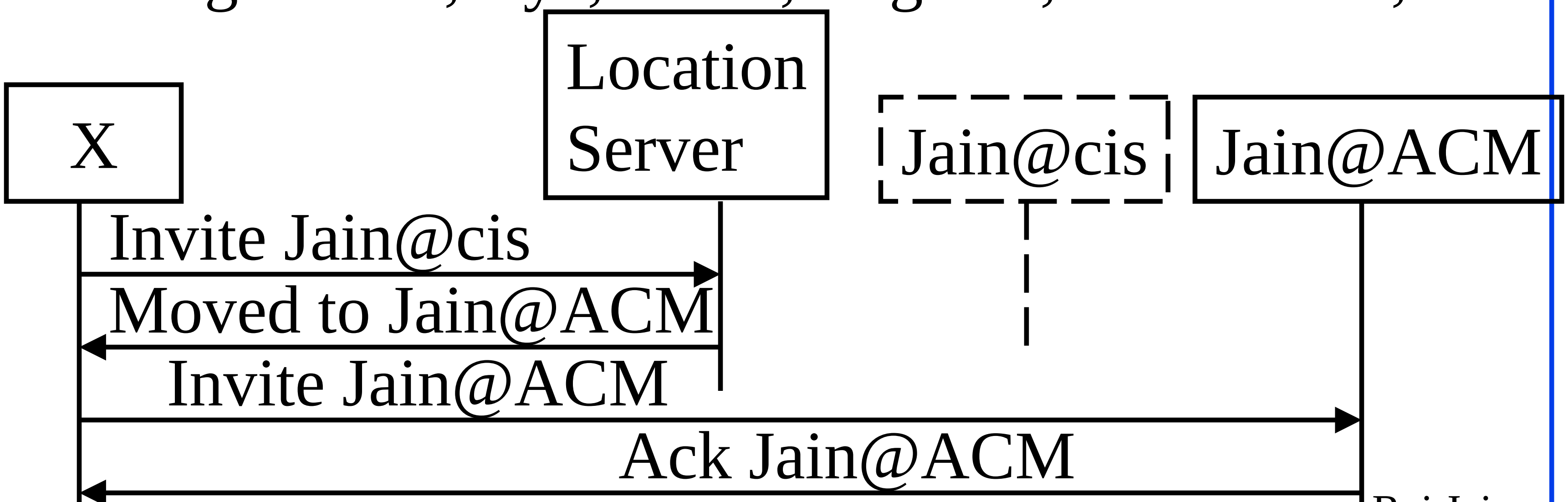
- Real time streaming protocol
- Application level protocol similar to hyper-text transfer protocol (HTTP/1.1) for audio/video
- Maintains state $\Rightarrow$ Setup/teardown messages
- RTSP messages use TCP, UDP, ...
- Data transfer is done separately using TCP, RTP/UDP, ...
- Uses URLs, e.g., rtsp://media.example.com:554/twister/audiotrack
- Both servers and clients can issue requests. HTTP servers do not issue requests.

RTSP Methods

- Setup: Start a new session
- Teardown
- Redirect
- Play
- Record
- Pause
- Describe: Tell me about session X
- Announce: A session X will take place at t
- Get_parameter: Get server/client statistics
- Set_parameter
- Options: I can accept only these options.

SIP

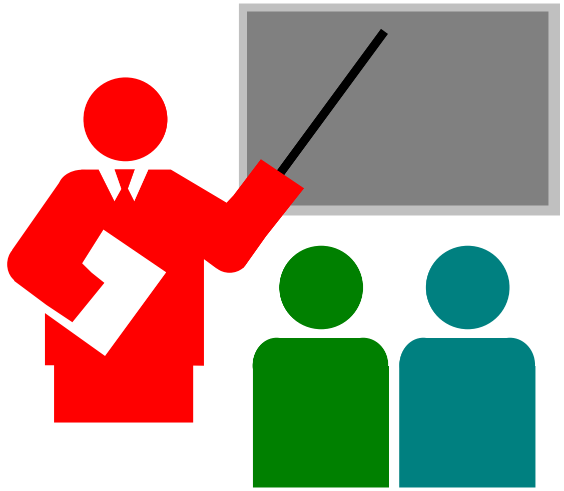
- Session Initiation Protocol
- Allows locating a callee at different locations
- Callee registers different locations with SIP Server
- Servers can also use finger, rwhois, LDAP to find callee
- Messages: Ack, Bye, Invite, Register, Redirection, ...



Session Description Protocol (SDP)

- Developed by Multiparty Multimedia Session Control (MMUSIC) working group
- Allows development of compatible session announcement tools
- Session description contains session name, description, URL, owner/creator, email address, phone number, originating host, connection info, bandwidth info, zero or more repeat times, encryption key, session attributes
- Media description includes media name and transport addr, connection info, encryption key, media attributes

Summary



- TCP/IP protocols suite is being extended to allow multimedia on Internet
- "Transport" Protocol: RTP, RTCP, RTSP
- Session protocols: SIP, SDP

References

- For a detailed list of references see: http://www.cis.ohio-state.edu/~jain/refs/mul_refs.htm

- RFC 1889, “RTP: A Transport Protocol for Real-Time Applications,”
- RFC 2327, “SDP: Session Description Protocol,” 4/14/98.
- RFC 2326, "Real Time Streaming Protocol (RTSP)", 04/14/1998.
- SIP - Session Initiation Protocol, draft-ietf-mmusic-sip-08.txt, 8/11/98.

Quality of Service Over IP

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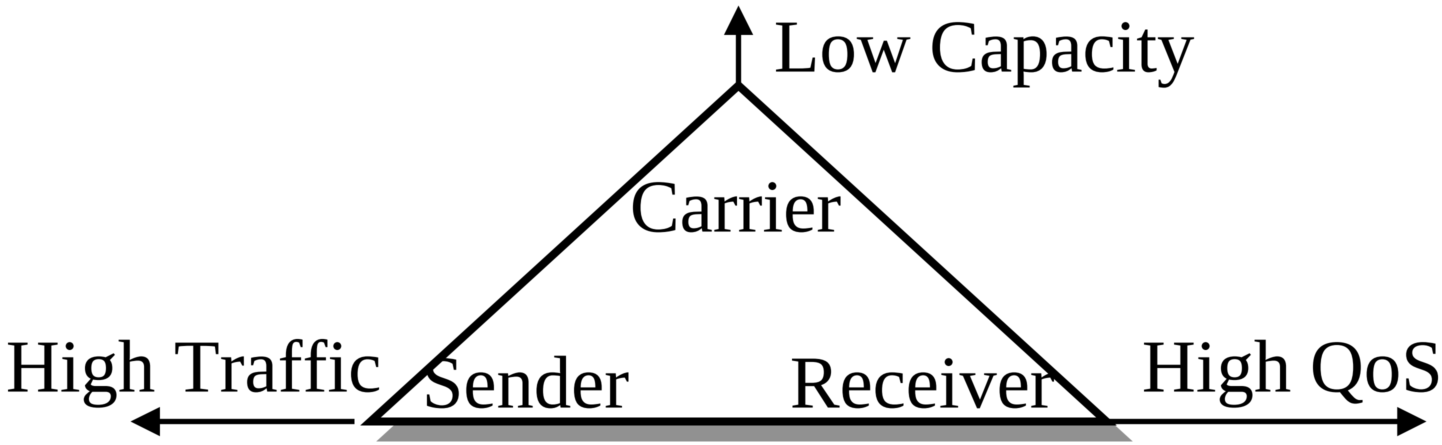
- Integrated services
- Resource Reservation Protocol: RSVP
- Differentiated Services
- QoS routing
- Multiprotocol Label Switching (MPLS) CoS

QoS Triangle



- Senders want to send traffic any time with high load, high burstiness
- Receivers expect low delay and high throughput
- Since links are expensive, providers want to minimize the infrastructure
- If one of the three gives in $\Rightarrow$ no problem

Components of QoS Architecture

1. Services with different QoS: Service definitions
2. Ways for users to communicate what they need: Signaling or admission control, policy management
3. Ways for providers to ensure that users are following their commitment: Policing/shaping
4. Ways for providers to find the routes: QoS based routing
5. QoS based forwarding: Buffer Allocation and Drop Policy, Queueing Discipline and Service Policy, Traffic Management

Integrated Services

- Best Effort Service
- Controlled-Load Service: Performance as good as in an unloaded datagram network. No quantitative assurances. (Min throughput)
- Guaranteed Service: rt-VBR
  - Firm bound on data throughput and delay.
  - Delay jitter or average delay not guaranteed or minimized.
  - Every element along the path must provide delay bound.
  - Is not always implementable, e.g., Shared Ethernet.

Flow Specification

- TSpec: Peak rate (p), bucket rate (r), bucket size (b), max datagram size (M), min policed unit (m)
  - All datagrams less than m are counted as m bytes
  - Peak rate may be unknown or unspecified
- RSpec (QoS): Allocated Rate (R) and delay slack (S)
  S = Extra acceptable delay over that obtainable with R
  Zero slack ⇒ Reserve exactly R.
- RSpec specified only for guaranteed rate service.
  Not for controlled load service.

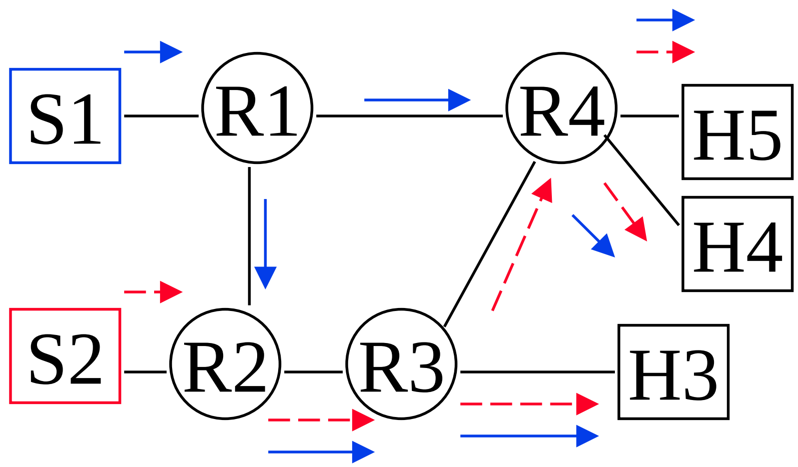
AdSpec Parameters

- Break Bit ⇒ Non-IntServ-Hop
  ⇒ IntServ-unaware router on the path
- Number of IntServ Hops
- Available Path Bandwidth: Bytes per second
- Minimum Path Latency:
  Σ Propagation+Min processing delays
  Does not include queueing delays
- Path MTU: Maximum transmission unit for QoS
  May be less than physical MTU
  Physical MTU available only to sources

RSVP

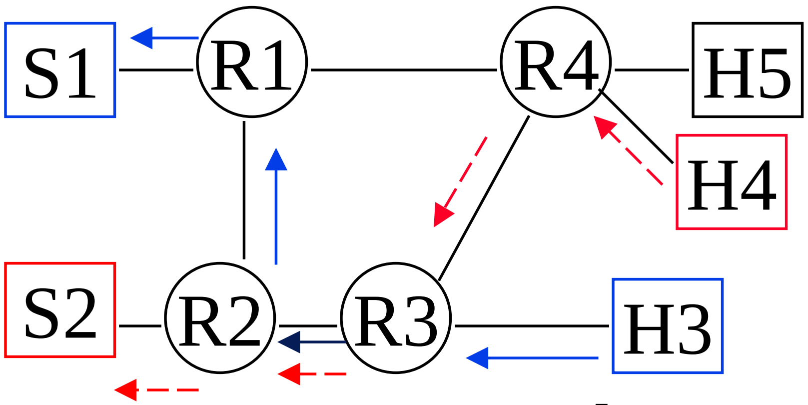
- Resource ReSerVation Protocol
- Internet signaling protocol
- Carries resource reservation requests through the network including traffic specs, QoS specs, network resource availability
- Sets up reservations at each hop
- RSVP does not find routes.
  Multicast routing protocols do.

Path Messages



- Sources send quasi-periodic PATH messages to multicast address
- Path message contain:
  - Sender Template: Data format, Src Address, Src Port
  - Sender TSpec: Traffic Characteristics. Not changed.
  - ADSpec: Network path resource/service availability Accumulated along the path.

Reservation Requests



- Receivers must join multicast address to receive path messages
- Receivers generate reservation (RESV) requests
- RESV messages contain resources to be reserved
- RESV messages are forwarded along the reverse path of PATH messages

Reservation (Cont)

- Requests are checked for resource availability (admission control) and administrative permissions (policy control)
- Two or more RESV messages for the same source over the same link are merged.
- Routers maintain a soft state.
  The receivers have to refresh periodically.
- Heterogeneous Receivers: Sources divide traffic into several flows. Each flow is a separate RSVP flow. Receivers join one or more flows. Each RSVP flow is homogeneous.

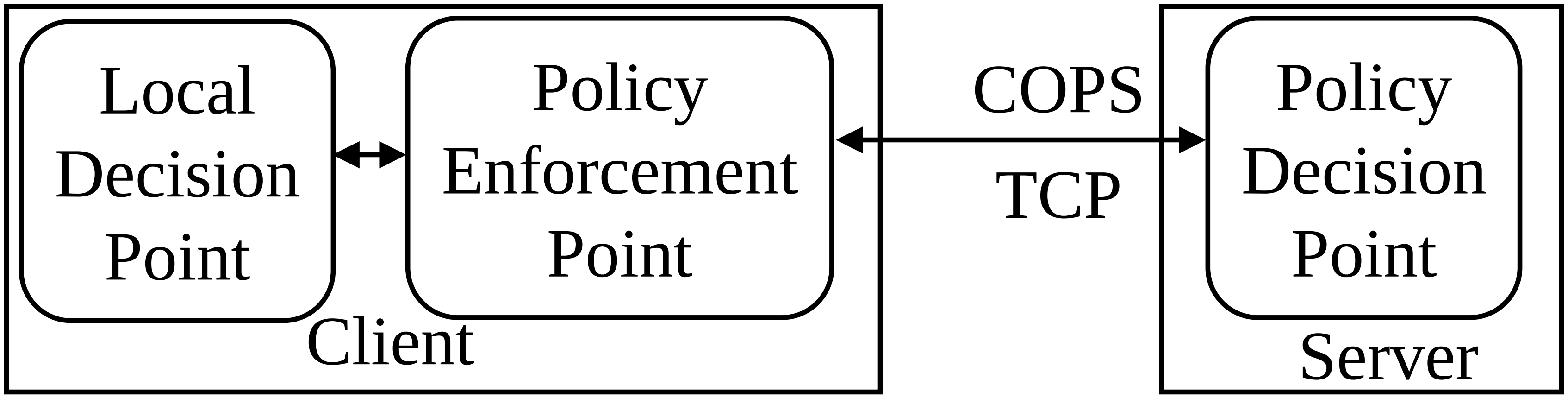
Problems with RSVP and Integrated Services

- Complexity in routers: packet classification, scheduling
- Scalable in number of receivers per flow but Per-Flow State: O(n) ⇒ Not scalable with # of flows. Number of flows in the backbone may be large.
  ⇒ Suitable for small private networks
- Need a concept of "Virtual Paths" or aggregated flow groups for the backbone
- Need policy controls: Who can make reservations? Support for accounting and security.
  ⇒ RSVP admission policy (rap) working group.

Problems (Cont)

- Receiver Based:
  Need sender control/notifications in some cases.
  Which receiver pays for shared part of the tree?
- Soft State: Need route/path pinning (stability).
  Limit number of changes during a session.
- Throughput and delay guarantees require support of lower layers. Shared Ethernet $\Rightarrow$ IP can’t do GS or CLS. Need switched full-duplex LANs.
- Can’t easily do RSVP on ATM either
- Most of these arguments also apply to integrated services.

COPS Protocol



- Common Open Policy Service Protocol
- When the routers (clients) receive a RSVP message, they send the request to server and obtain authorization
- Will work with other (non-RSVP) signaling
- Routers can make local decisions but should keep servers informed
- Servers can send unsolicited responses for changes later

IP ToS Field

Ver	Hdr Len	Precedence	ToS	Unused	Tot Len
4b	4b	3b	4b	1b	16b

- IPv4: 3-bit precedence + 4-bit ToS
- RFC791: ToS determines packet treatment and monitory considerations
- RFC1349: $bit_1 \Rightarrow$ min delay, $bit_2 \Rightarrow$ max throughput, $bit_3 \Rightarrow$ max reliability, $bit_4 \Rightarrow$ min cost
- OSPF and integrated IS-IS can compute paths for each ToS

Differentiated Services Working Group

- August 97: BOF started
- Feb 98: Working group formed
- Dec 98: Final document (?)
- Email: majordomo@baynetworks.com in body: subscribe diff-serv
- Archive: http://www-nrg.ee.lbl.gov/diff-serv-arch/
- Charter: define ds byte (IPv4 ToS or IPv6 traffic class octets)

Diff-Serv Terminology

- Service: Offered by the protocol layer
  - Application: Mail, FTP, WWW, Video,...
  - Transport: Delivery, Express Delivery,...
    Best effort, controlled load, guaranteed service
- Per-Hop Behavior (PHB): Mechanisms - Drop threshold, Queue assignment, Service priority, Service Rate
- Flow: Packets with specific header fields, Destination Address, Source Address, Port, Flow Label
- Aggregates: Stream of packets with the same DS byte pattern

Initial proposals

- Assured service (Jacobson): traffic profile (VBR or CLS like), in-profile and out-profile
- Premium Service (Clark): Peak rate (CBR or GS like), Virtual leased line
- 2 Priority bits, 2 drop bits
- Bits for delay class: 2 bits $\Rightarrow$ 4 classes
  Bits for Drop preference: 2 bits
  $\Rightarrow$ Up to 4 drop preferences

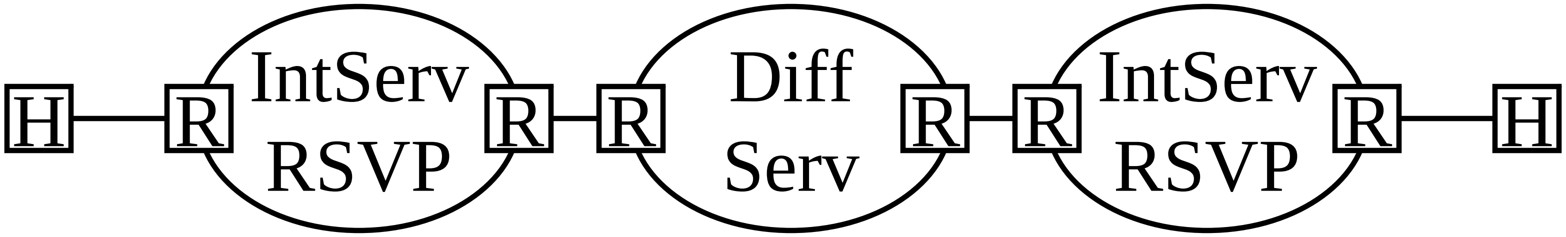
Sample PHB Allocation

- ppp i 00
- ppp = Precedence (Higher is generally better)
- i = in/out bit ⇒ In profile/out Profile
  ⇒ Drop preference. Allows in/out pkts in same Queue
- Aug'98 IETF: Four queues and 3 drop preferences
- Compatible with current usage
- Precedence is used as an index to select a queue, or VC, ...
- In IEEE-802 switches, only 1, 2, or 3 msbs used
- Unrecognized code points ⇒ Default forwarding

PHB Allocation (Cont)

- Plan: 32 code points standard,
  16 Experimental/local use, 16 reserved
  xxxxx0 Standard
  xxxx11 Experimental/Local Use
  xxxx01 Reserved for future

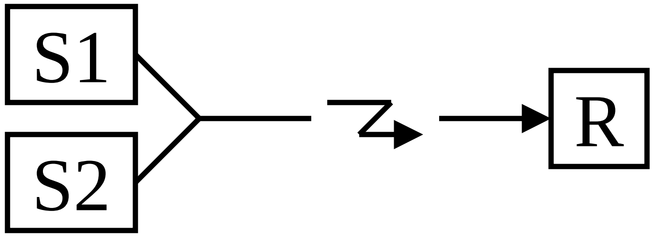
End-to-end QoS



- Hosts may mark DS byte or use RSVP signaling or both or none.
- Why hosts? 1. Encryption, 2. Hosts know the importance of info even if the header fields are same
- Routers may mark DS byte if necessary.
- Routers at the intserv diff-serv boundary accept/reject RSVP requests based on current load

- Service between intserv and diff-serv regions can be statically or dynamically provisioned
- Current integrated services (CLS, GS) may or may not be practical
- DS byte may be modified at network boundary

Issues



- Standard code points (behaviors)
- Receiver control over incoming low-speed link
- Signaling: Should users signal
  or network managers set resource allocations
- Dynamic or Static management controls?
- Billing: Bit for receiver billing. If receiver billing, the receiver should be able to deny/drop packets received.
- Congestion Check Bit: If set, network indicates highest priority for which packets are being dropped in the ToS byte.

QoS Extensions to OSPF

- Open shortest path first
- Separate metric can be specified for each ToS supported
- OSPF options field has a T-bit
  T-bit = 1 $\Rightarrow$ Router can compute routes for each ToS
- Work to extend OSPF is currently underway
- QoS $\Rightarrow$ Frequent updates
  $\Rightarrow$ Instability: Underloaded links become overloaded
  Also, complexity

Inter-Domain QoS Routing

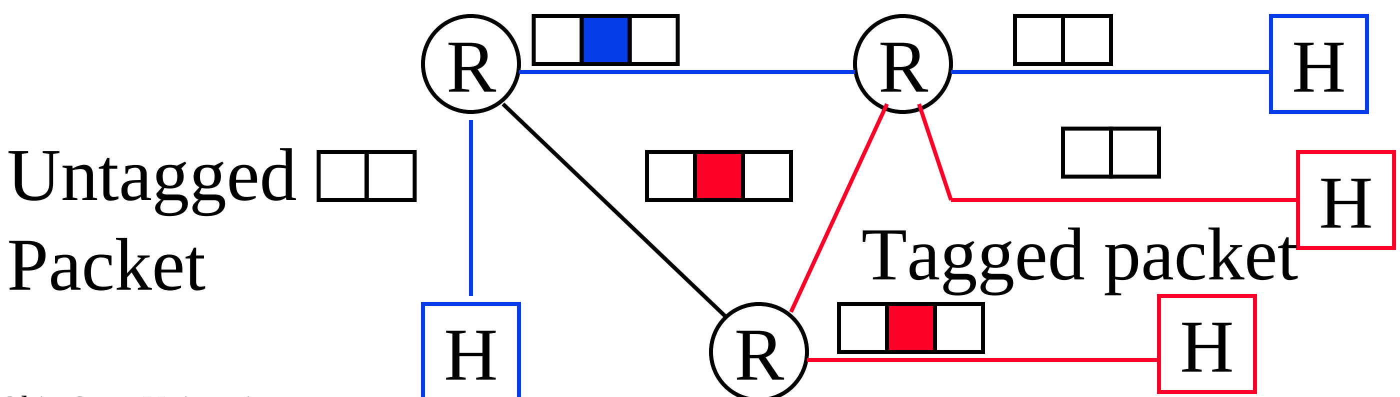
- Domains want to limit the frequency and amount of information exchanged $\Rightarrow$ Stability
- QoS based routing may cause frequent changes and instability
- QoS extensions to Border Gateway Protocol (BGP) proposed but may or may not happen
- Need hierarchical aggregation for scalability Crank-back

Tag Switching

- Proposed by CISCO
- Similar to VLAN tags
- Tags can be explicit or implicit L2 header

L2 Header	**Tag**	

- Ingress router/host puts a tag. Exit router strips it off.

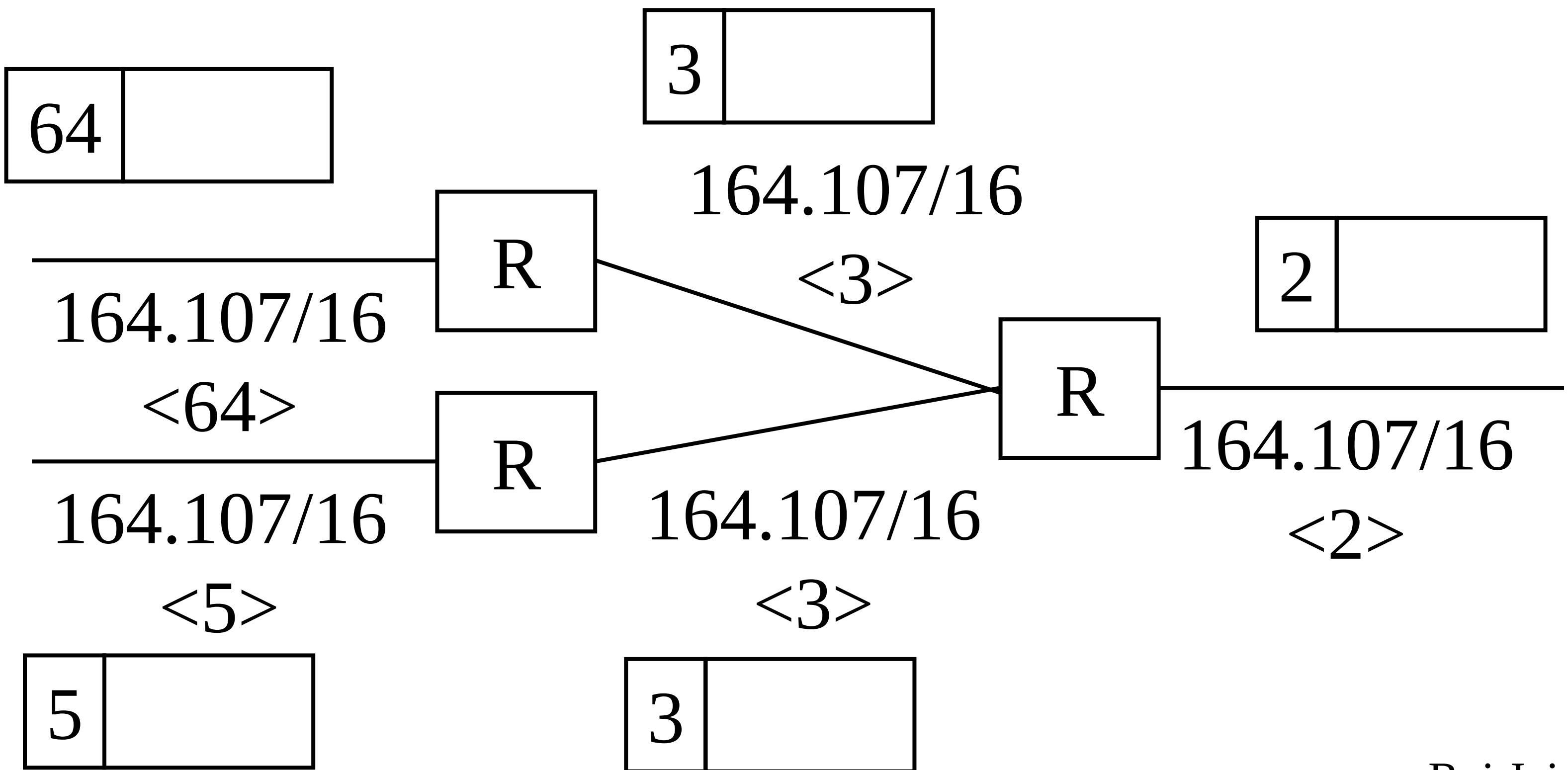


Tag Switching (Cont)

- Switches switch packets based on labels.
  Do not need to look inside $\Rightarrow$ Fast.
- One memory reference compared to 4-16 in router
- Tags have local significance
  $\Rightarrow$ Different tag at each hop (similar to VC #)

Tag Switching (Cont)

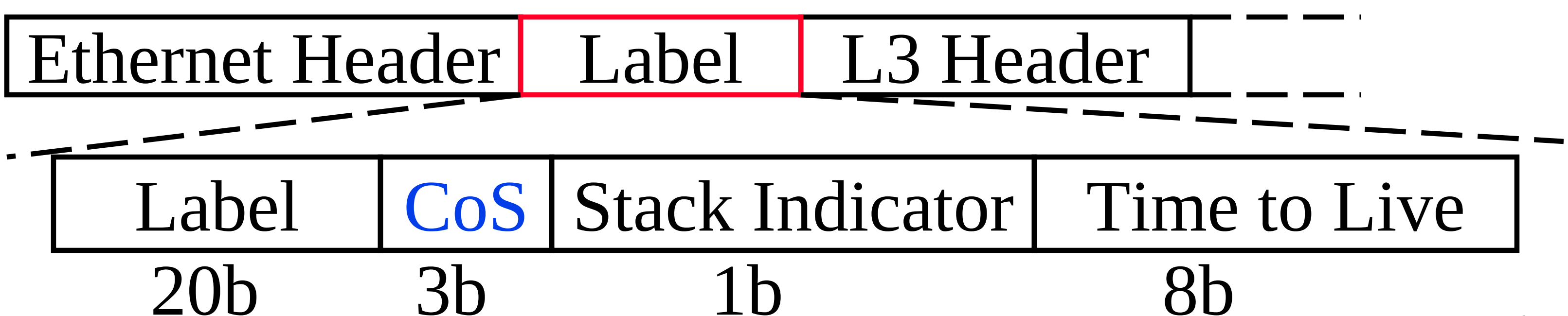
- One VC per routing table entry



MPLS

Prefix	Label	Out-Port	Out-Label
164.107.0.0/16	1	2	3
164.107.0.0/24	2	3	4
…	…	…	…

- Multiprotocol Label Switching
- Current: Longest prefix match on the dest address
- With Labels: Search can be replaced by indexing
- MPLS labels contain 3-bit CoS

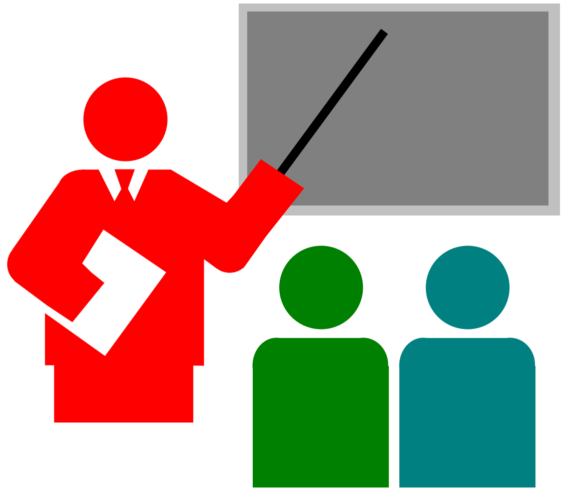


Label Stacks

- Labels are pushed/popped as they enter/leave MPLS domain
- Routers in the interior will use Interior Gateway Protocol (IGP) labels. Border gateway protocol (BGP) labels outside.

L2 Header	Label 1	Label 2	...	Label n	

Summary



- Internet protocols suite is being extended to allow QoS
- Integrated Services: GS = rtVBR, CLS = nrt-VBR
- Signaling protocol: RSVP
- Differentiated Services will use the DS byte
- QoS Routing: QOSPF
- Multiprotocol Label Switching has 3-bit CoS

IETF QoS Working Groups

- Integrated Services (intserv)
- Integrated Services over Specific Link Layers (issll)
- Resource Reservation Setup Protocol (rsvp)
- QoS-based Routing (qosr)
- Differentiated services (diff-serv)

References

- For a detailed list of references see: http://www.cis.ohio-state.edu/~jain/refs/ipqs_ref.htm
- RFC 2212, "Specification of Guaranteed Quality of Service", 9/97
- RFC 2211 "Specification of the Controlled-Load Network Element Service", 9/97
- RFC 2205, "Resource ReSerVation Protocol (RSVP) -- Version 1 Functional Specification", 11/97
- Diff-Serv Mail Thread index, http://www-nrg.ee.lbl.gov/diff-serv-arch/

Multicasting Over Internet

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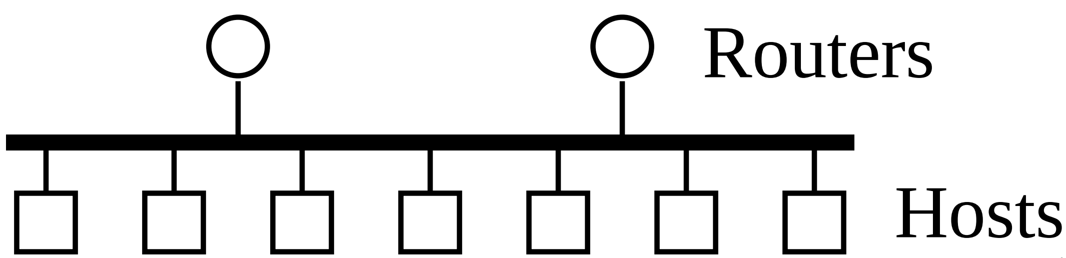
- Multicast addressing and registration: IGMP
- Multicast Backbone: Mbone
- Multicast Routing: MOSPF

IGMP

Ver	Type	Reserved	Checksum
Group Address			

- Internet Group Management Protocol
- Used by hosts to report multicast membership
- Join-IP-Multicast Group (address, interface)
- Leave-IP-Multicast Group (address, interface)



IGMP Operation

- One "Querier" router per link
- Every 60-90 seconds, querier broadcasts "query" to all-systems (224.0.0.1) with TTL = 1
- After a random delay of 0-10 seconds, hosts respond for each multicast group
- Everyone hears responses and stops the delay timer $\Rightarrow$ One response per group
- Non-responding groups are timed-out

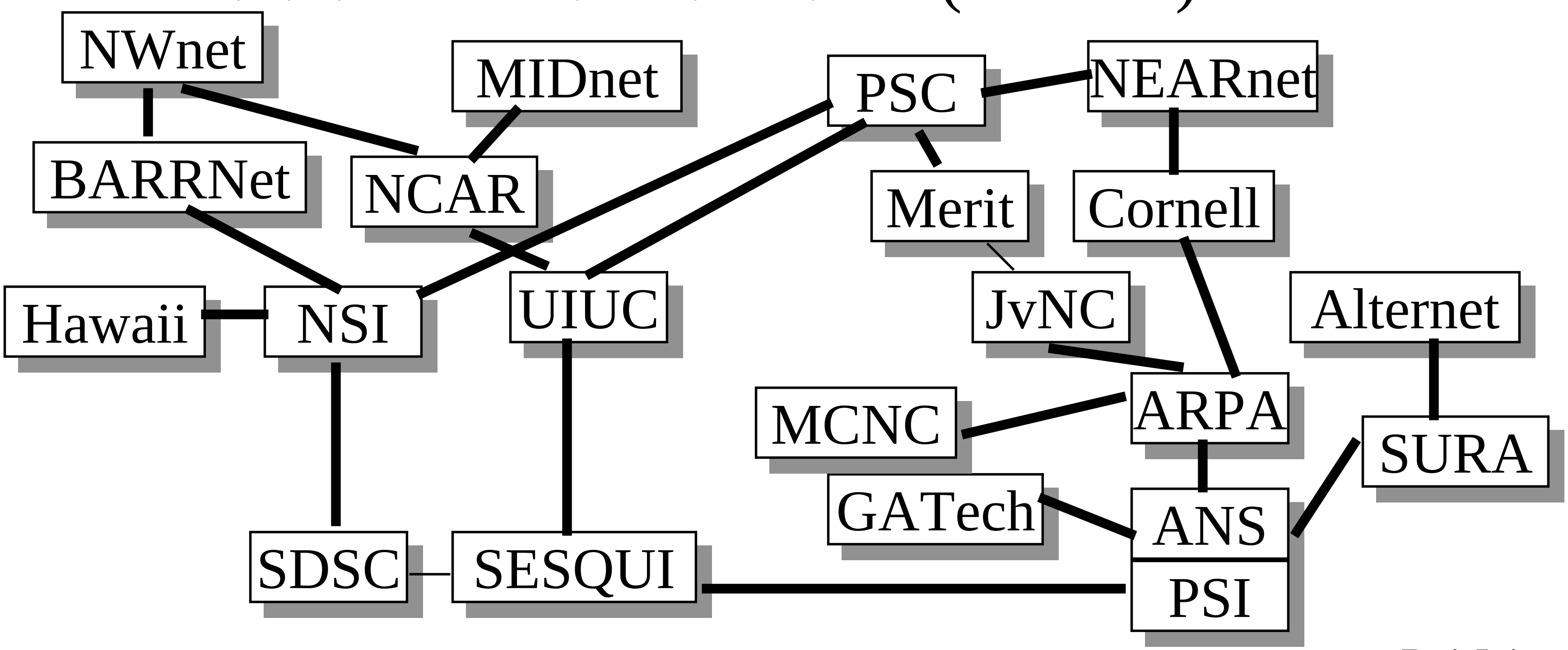
IGMP Version 2

Type	Max Resp	Checksum
Group Address		

- Querier election method
- Messages include "maximum response time"
- "Leave group" message to reduce leave latency
  Sent only if the host that responded to the last query leaves
- Querier then issues a "membership query" with a short response time
- Already implemented. RFC 2236.
- IGMP V3 is being designed.

MBone

- Internet Multicast backbone
- A set of routers that implement IP multicasting
- IP multicast address: start with 1110... (binary), 224.0.0.0 to 239.255.255.255 (decimal)



MBone (Cont)

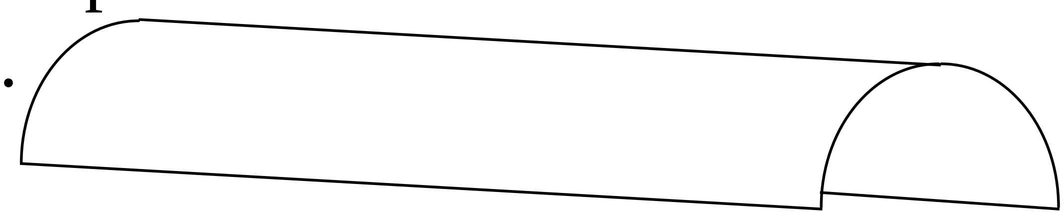
- Uses radio/TV station paradigm: Sender is allocated a multicast address and it starts transmitting on that address
- Anyone can listen by tuning into the multicast address by sending an Internet Group Management Protocol (IGMP) request to router to join the multicast
- The router provides a connection to the nearest point
- Sender has no idea of who is listening
  Sender controlled multicasts does not scale well.

MBone (Cont)

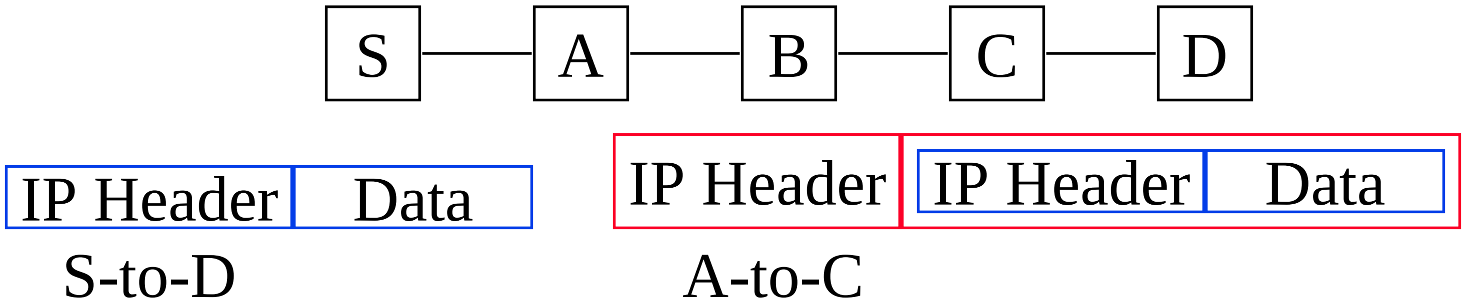
- First audiocast in March 1992: IETF meeting to 20 sites
- Now over 600 hosts in over 15 countries
- Programs include space shuttle, conferences, IETF,...
- President Clinton and VP Gore have appeared
- Is a source of heavy traffic, congestion, and complaints

Mrouted

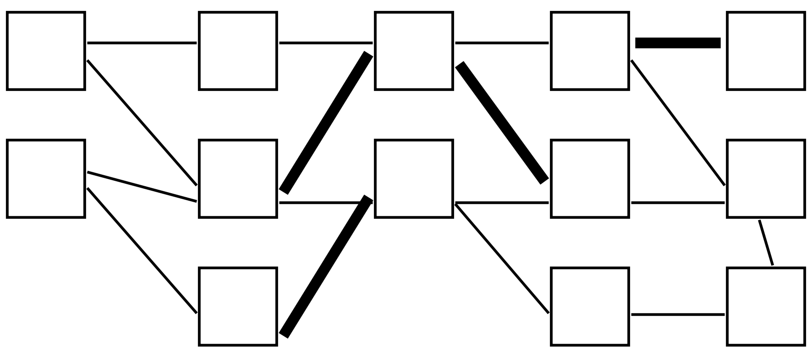
- The routing protocol that allows IP multicast
- Software available on the Internet.
  Join the MBone mailing list.
- Many vendors implement it already in their routers
- To connect find the nearest Mrouted.
  Maps available on the net.
- Mrouteds setup tunnels between them.
  Tunnel = direct connection
- Routers on the path of the tunnel do not need to know multicasting.



Tunnels



- Implemented by encapsulating the entire packet in another IP header.
- Each tunnel has a cost. Least cost path is found by exchanging distance-vectors with neighbors.



Tunnels (Cont)

- Each tunnel requires 100 to 300 kbps.
  Use 500 kbps for design.
  A few tunnels can saturate the host.
  Four on SPARC 1, six on SPARC 10.
  Fifteen tunnels can saturate an Ethernet.
  Maximum two tunnels over T1.
- Each packet has a time to live (TTL).
  TTL is decremented at each router.
  The packet is forwarded iff its TTL is over a threshold.
- Periodically, leaf mrouteds poll to see if there are any listeners.

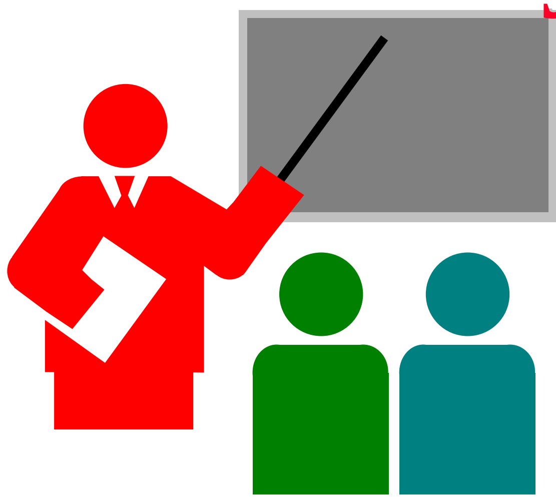
Tunnels (Cont)

- Pruning: If an mrouted gets a packet for which it has no listeners, it sends a message to the upstream mrouted to stop sending.

MOSPF

- Multicast Extensions to OSPF
- Alternative to Mrouted (IP multicasting)
- MOSPF routers use Internet Group Management Protocol (IGMP) same as IP
- OSPF is a link-state routing protocol
- Each router builds a complete source-rooted tree to all multicast destinations
- Designed for use within a single autonomous system
- Need inter-AS protocols for larger multicasts, e.g. DVMRP

Summary



- IP multicast addresses
- Multicast registration using IGMP
- Multicast propagation on Mbone using Mrouted
- Multicast routing: MOSPF

References

- For a detailed list of references see: http://www.cis.ohio-state.edu/~jain/refs/ipm_refs.htm

Voice And Telephony over ATM

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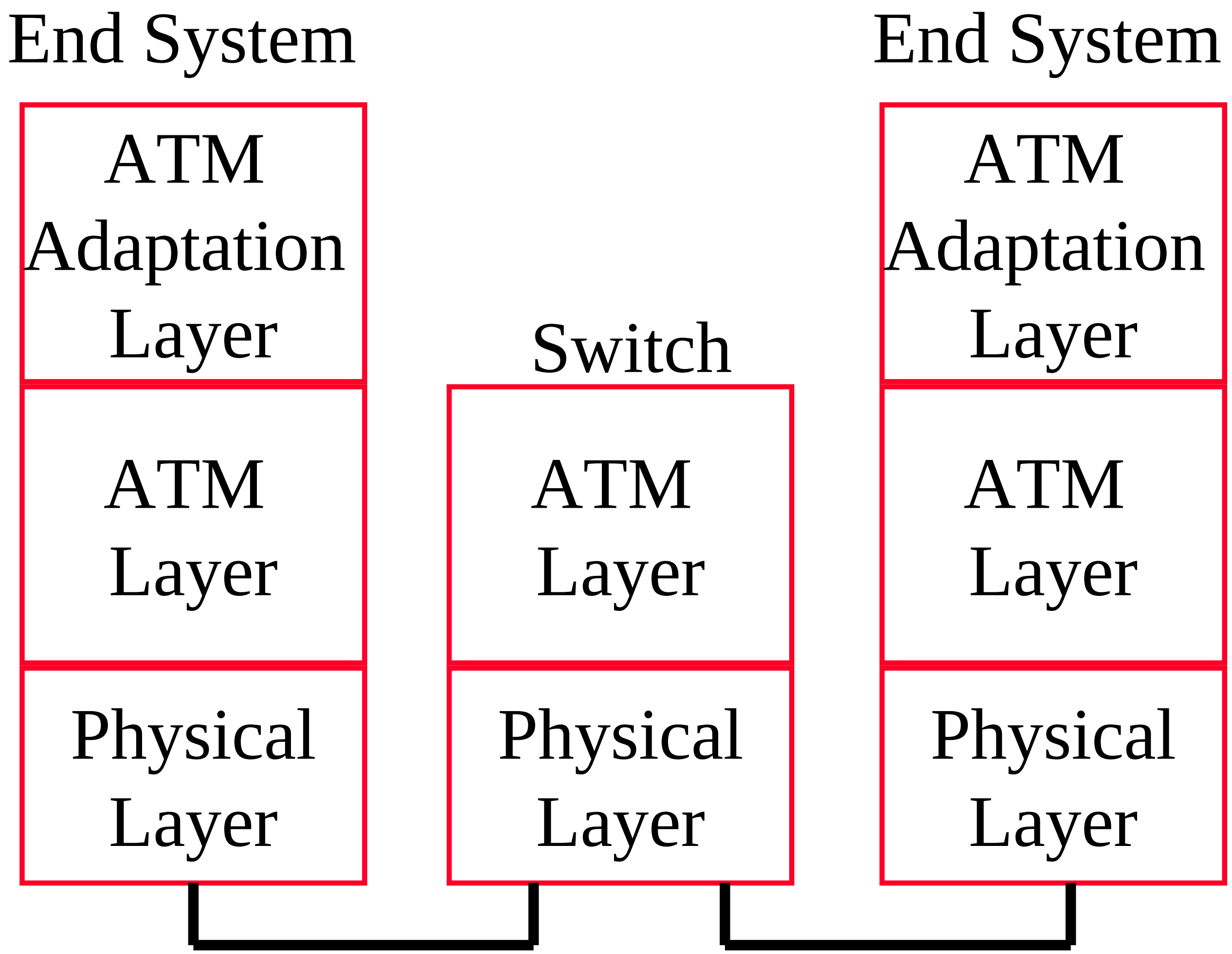
- VTOA: Protocol Stack and Services
- ATM Adaptations Layers: AAL1, AAL5
- New AAL2

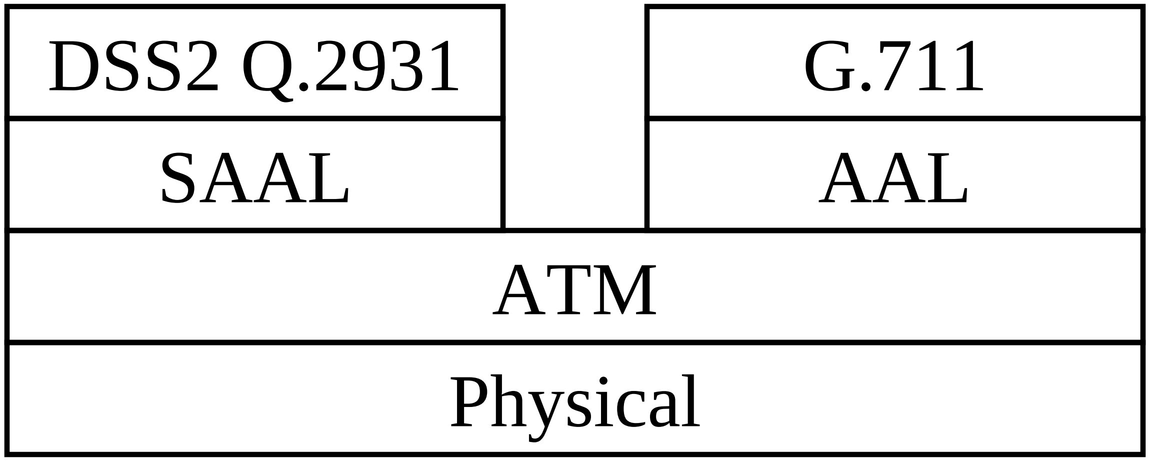
Protocol Layers



Protocol Layers

- The ATM Adaptation Layer
  - How to break messages to cells
- The ATM Layer
  - Transmission/Switching/Reception
  - Congestion Control/Buffer management
  - Cell header generation/removal at source/destination
  - Cell address translation
  - Sequential delivery

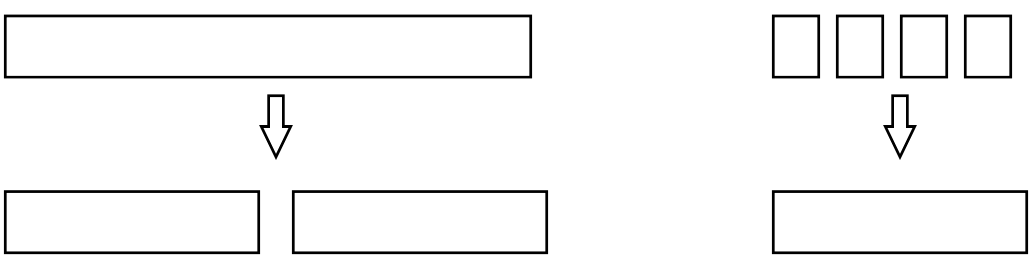
Protocol Reference Model



- AAL1 or AAL5. AAL5 required.
- One packet per cell
- 64 kbps PCM μ-law or A-law (G.711)

Delay

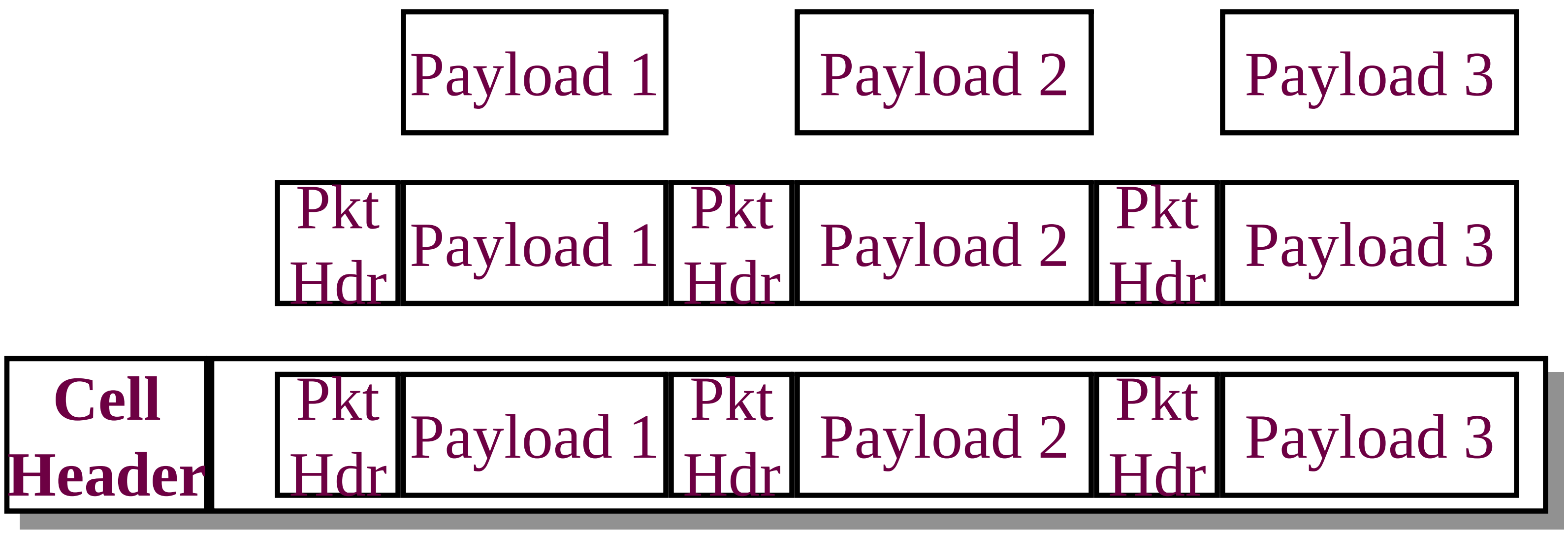
- 48 bytes at 64 kbps = 6 ms
  ⇒ Need Echo cancellers
- 48 bytes at 16 kbps = 24 ms ⇒ too long
- Wireless: 8 kbps ⇒ 48 Bytes = 48 ms
- Can't fill a cell completely
- Current AALs allow segmentation
  (long packets to multiple cells).
- Do not allow blocking (short packets in one cell)



AAL

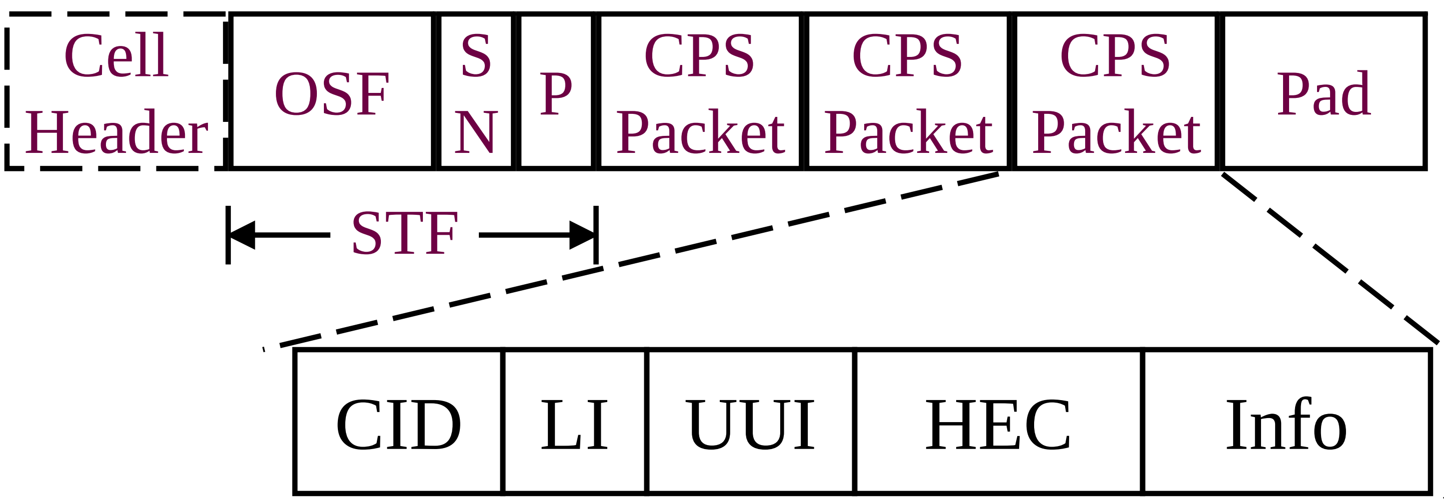
AAL2

- Ideal for low bit rate voice
- Variable/constant rate voice
- Multiple users per VC
- Compression and Silence suppression
- Idle channel suppression



Cell Format

- STF: Start field = CPS PDU header
- OSF: Offset of the first packet
- SN: Sequence number mod 2, 0 or 1
- P: Parity (odd) of start field
- Pad: Padding (0-47 bytes)



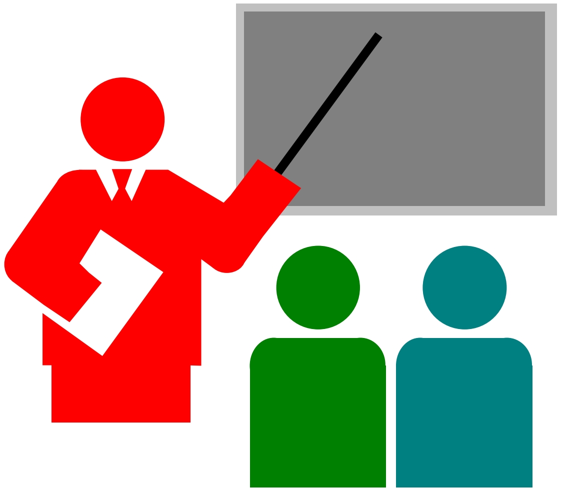
AAL2: Status

- Sept 97: I.363.2 approved
- Sept 97: Segmentation and reassembly I.366.1 frozen
- June 98: I.trunk frozen
- On-Going:
  - AAL2 negotiations procedures (ANP)
  - Operations, Administration and Maintenance (OAM)
- Future: Interworking with
  - Voice over IP
  - Voice over Frame Relay

Desired Changes to ATM

- Heterogeneous Point-to-Multipoint: Variegated VCs
- QoS Renegotiation
- Group Address
- Lightweight Signaling

Summary



- Circuit emulation services for CBR using AAL1 or AAL5.
- ATM Trunking using AAL2 is being developed. Allows low bit rate VBR, multiple users/cell

References

- For tutorials on VTOA, Signaling, and PNNI see: http://www.cis.ohio-state.edu/~jain/atm/
- ATM Forum Standards, see http:/www.cis.ohio-state.edu/~jain/atmf_ref.htm
- ATM Forum, "Voice and Telephony over ATM to the Desktop," af-vtoa-0083.000, May 1997
- ATM Forum, "Circuit Emulation Service Specification V2.0," af-vtoa-0078.000, January 1997.
- ATM Forum, "Dynamic Bandwidth Utilization in 65 kbps time-slot trunking over ATM- using CES," af-vtoa-0085.000, July 1997

- ATM Forum, "ATM Trunking using AAL1 for Narrowband Services V1.0," af-vtoa-0089.00, July 1997
- ITU-T, "B-ISDN ATM Adaptation Layer Specification: Type 2 AAL," I.363.2
- ITU-T, "B-ISDN ATM Adaptation Layer Specification: Type 1 AAL," I.363.1, Aug 96.
- ITU-T, "B-ISDN ATM Adaptation Layer Specification: Type 5 AAAL," I.363.5, Aug 96.

Final Review: 25 Facts

1. Voice over IP products and services are being rolled out
2. Ideal for computer-based communications
3. IP needs QoS for acceptable quality
4. A number of working group at IETF are working on it
5. H.323 provides interoperability
6. Gatekeeper allows managing resources on a network. Gateways translate protocols between dissimilar networks.

Final Review (Cont)

7. Traffic classes and dynamic multicast on LANs to allow multimedia
8. IEEE 802.1p allows 8 priorities
9. Distributed multicast registration protocol
10. Virtual LANs Location independent LAN Groups
11. IEEE 802.1Q allows both explicit and implicit tagging
12. 802.1Q explicit tagging allows priority signaling
13. RTP and RTCP allow provide several common functions for multimedia applications

Final Review (Cont)

14. RTSP allows controlling streaming media
15. SIP allows locating a user
16. Integrated Services: GS = rtVBR, CLS = nrt-VBR
17. Signaling protocol: RSVP
18. Differentiated Services will use the DS byte
19. QoS Routing: QOSPF

Final Review (Cont)

20. Multiprotocol Label Switching has 3-bit CoS
21. Multicast registration using IGMP
22. Multicast propagation on Mbone using Mrouted
23. Multicast routing: MOSPF
24. Circuit emulation services for CBR using AAL1 or AAL5.
25. ATM Trunking using AAL2 allows low bit rate VBR, multiple users/cell