Seminar: Multimedia Coding and Transmission

Multimedia Transmission over (IP) Networks

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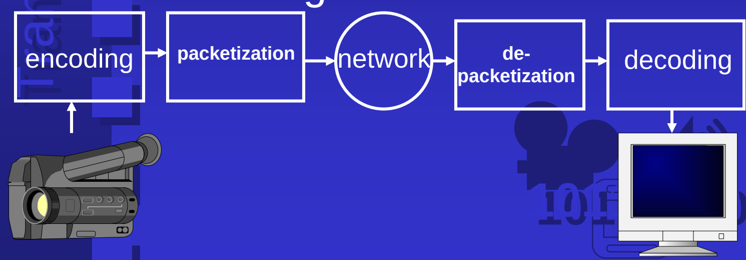
Today

- Transmission over IP networks
  - Network characteristics
  - Protocols: RTP, RTCP ++
  - Case: H.263, MPEG-2
- Transmission over other networks
  - Wireless

Transmission

- So far: audio/video *encoding/decoding*
- Today: *Transmission* of compressed multimedia over (IP) networks in real-time - streaming



Transmission networks

- Why IP focus of this talk?
  - most widespread at user level
  - future of multimedia transport
  - cheap technology
  - best known by me!
- Little (no) lower layer talk
  - Mention wireless networks
  - Not link layer: ATM, Ethernet…
  - Not physical layer: SDH, SONET...

IP networks (IN270 light)

- IP protocol suite
  - IP: addresses, checksum
  - UDP: unreliable, unordered datagram service, port numbers, used for *media*
  - TCP: reliable (retransmission), ordered (ack'ed) byte stream service, port numbers used for *control*
- Unicast: one-to-one
- Multicast: one-to-many, IPv4: 224.0.0.0 - 239.255.255.255

Network characteristics (1)

- Network delay
  - propagation (distance+medium), 3-5µs/km
  - transmission (network interface), ~20µs/hop

  fixed

  - variable queuing delay from congestion, link-layer retransmission
  - Remedy: playout delay buffer compensation
- Application delay
  - coding (processing), look-ahead
- E2e delay = network + application

Network characteristics (2)

- Packet loss
  - what?
    - either a packet *never* arrives
    - or too *late* to be of use
  - why?
    - dropped because of congestion (wired media)
    - corrupted packet checksum (wireless media)

Network characteristics (3)

- Packet reordering
  - different routes from source to destination
  - rare occurence
- Packet duplication
  - faulty hardware
  - lost ack’s at link-layer
- Connection refusal
  - reservation refused, insufficient resources (bandwidth)

Internet Service Grades

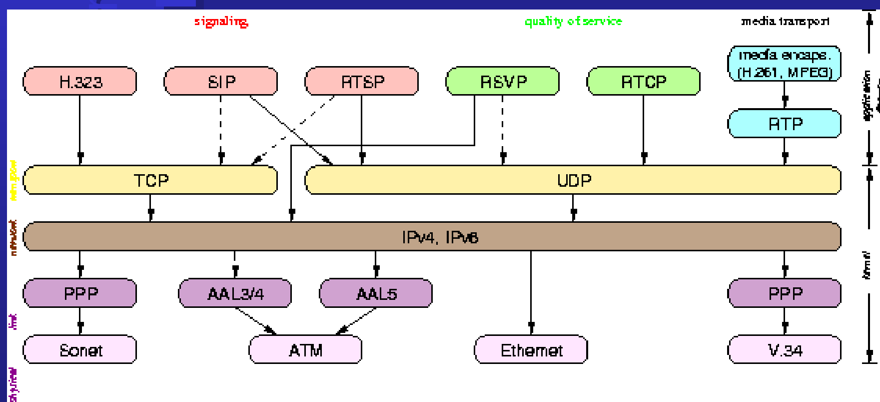
Characteristics	*UDP*	*TCP*
packet loss	**yes**	**no**
delay bound	**no**	**no**
abstraction	**packet**	**byte stream**
ordering	**none**	**in order**
duplication	**possible**	**no**
multicast	**yes**	**no**

- Always *tradeoff* between delay, reliability and throughput

How improve e2e quality?

- Error concealment techniques (during decoding)
- Layered coding
- Error resilience coding
- Variable bit rate coding
- Feedback of rate and loss information

IP multimedia protocol stack



IP continuous media support (1)

- Three types of architectural protocols:
- 1. Media transport
  - Standardized protocols such as e.g. RTP
  - Proprietary solutions such as e.g. RDT (Real)
- 2. Quality of Service (QoS)
  - Measurements e.g. RTCP
  - Set aside resources
    - flow based e.g. RSVP / IntServ
    - aggregate based e.g. DiffServ

IP continuous media support (2)

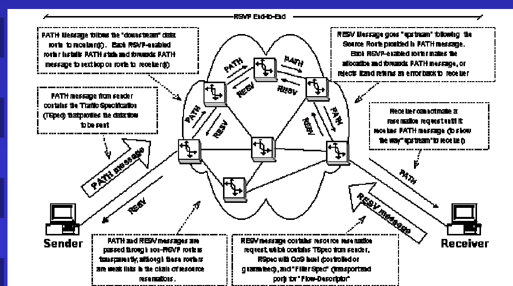
- 3. Signalling of stream control (application dependent)
  - Sender controlled session
    - Mbone, broadcast e.g. SAP
  - Receiver controlled session
    - Media on-demand e.g. RTSP
  - Bidirectional session, “meet up”
    - IP telephony, video conferences e.g. SIP/H.323
- Also
  - Content stream description e.g. SDP

Integrated services

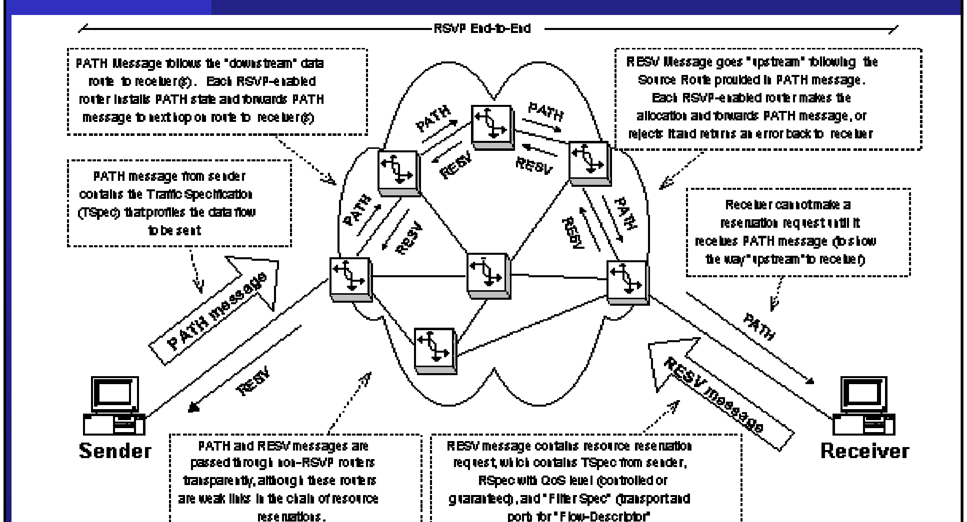
- Two new service classes
  - *Guaranteed* service
    - Absolute: mathematically computed delay and bandwidth bound
    - Traffic flow specification: TSpec + RSpec
  - *Controlled load* service
    - Relative: approximate lightly loaded best-effort network
    - Traffic flow specification: TSpec
  - How implemented?
    - E.g. Weighted Fair Queueing
  - Don’t scale!

RSVP: Resource Reservation Protocol

– Independent of, but used with IntServ
  - receiver-oriented
  - soft state in nodes, periodically update (30s)
  - m:n simplex connections



RSVP



Differentiated services

- Coarse traffic differentiation
  - Aggregate-based prioritation, e.g. all audio
  - Per-hop behaviour
  - Scalable in backbone: complexity in edges, no state
- Marking DS code points in DS field (6 bits):
  - IPv4: Type-of-Service, IPv6: Traffic Class field
- Standardized behaviour
  - *Expedited Forwarding*: `101110`
  - *Assured Forwarding Group*: `001010` to `100110`
  - Best-effort: `000000`

RTSP: Real-Time Streaming Protocol

- Control protocol for multimedia servers
- Text- based, HTTP-alike, both sides in control
- rtsp:// - URL indicate a ”presentation” or file
- Methods:
  - Setup
  - Play
  - Stop
  - Pause
  - Record
  - Describe
  - Redirect
  - etc.

Example:

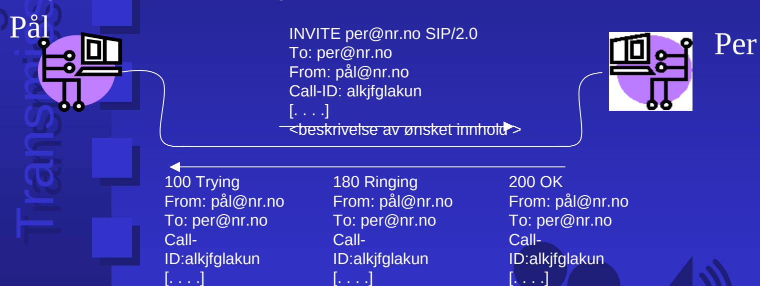
```
C->S: DESCRIBE
   rtsp://server.example.com/fizzle/foo RTSP/1.0
      CSeq: 312
      Accept: application/sdp, application/rtsl,
   application/mheg
S->C: RTSP/1.0 200 OK
      CSeq: 312
      Date: 23 Jan 1997 15:35:06 GMT
      Content-Type: application/sdp
      Content-Length: 376
      <SDP-beskrivelse av sesjonen, 376 tegn lang>
```

SAP: Session Announcement Protocol

- "TV-like" announcements
- Multicast invitations of one-to-many sessions
- Sent periodically every 5 minutes
- Limited to 4000 bits overall
- Contains info to start media tools needed to partake in session
- Of little use outside Mbone

SIP: Session Initiation Protocol

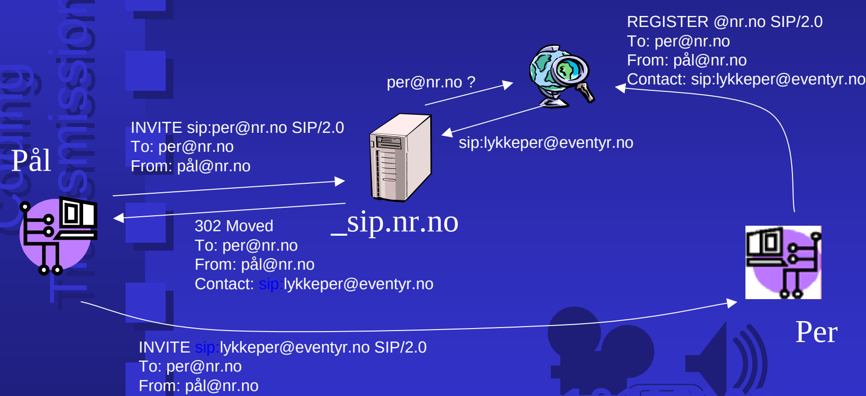
- Text-based, HTTP-alike: send message about contacting person to a program on his machine capable of "buzzing"



- Requires:
  - recepient has a server waiting for messages (SIP server)
  - sender has a client program to send them (SIP client)

Localisation of person

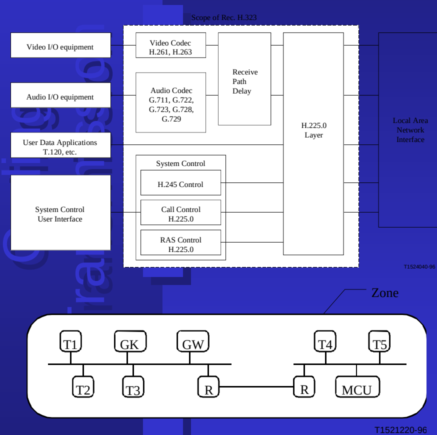
- *SIP-server* with *location service* and *registrar* enables automatic localisation of a person



SIP vs. H.323

- SIP (Internet-thinking) : Simple network as possible, smart end-terminals
  - Computers can do advanced things
  - Network moves packets from A to B
  - Service = configuration of programs on your computer(s)
- H.323 (telecom-thinking) : Smart network, simple end-terminals
  - You buy cheap terminal
  - We sell you the services you need
  - All services must be standardized

H.323



- Binary protocol suite
- Complex procedures
  - Phase A: Call setup.
  - Phase B: Initial communication and capability exchange.
  - Phase C: Establishment of audiovisual communication.
  - Phase D: Call services.
  - Phase E: Call termination

SDP: Session Description Protocol

- Text-based protocol for multimedia session descriptions
- Used by other IETF protocols (SAP, RTSP, SIP,…)

Example (from RTSP):

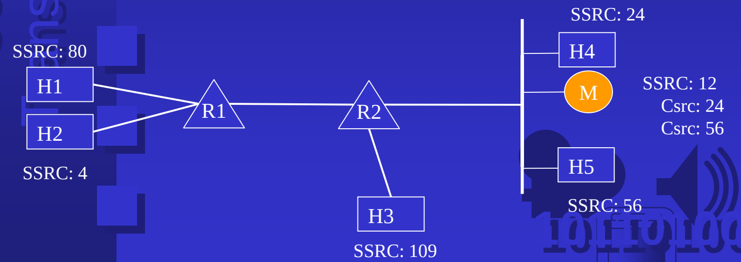
```
   v=0
   o=mhandley 2890844526 2890842807 IN IP4
126.16.64.4
   s=SDP Seminar
   i=A Seminar on the session description
protocol

u=http://www.cs.ucl.ac.uk/staff/M.Handley/s
dp.03.ps
   e=mjh@isi.edu (Mark Handley)
   c=IN IP4 224.2.17.12/127
```

```
t=2873397496 2873404696
a=recvonly
m=audio 3456 RTP/AVP 0
m=video 2232 RTP/AVP 31
m=whiteboard 32416 UDP
WB
a=orient:portrait
```

RTP: Real-time Transport Protocol

- Encapsulation protocol for real-time date over UDP (and multicast)
- Synchronize timing aspects and data source
- IETF RFC 1889



RTP header

Byte: 0 1 2 3
Bit:: 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1

ver =2	P	X	CSRC count	M	Payload type	sekvensnummer
timestamp						
SSRC-identifikator						
Evt. CSRC-identifikatorer						

P: padding
X: extension headers
M: marking
Payload Type: number for different formats, registered by IANA
Sequence Number: incremental, loss detection
Timestamp: when media *generated*, intramedia synchronization
CSRC: contributing sources in packet when multiplexed

RTCP: Real-Time Control Protocol

- Periodically exchange of information about multimedia session, i.e. quality and participants
- Neighbour port number to companion RTP-session
- Everybody sends reports to everybody they know of, but fixed overall bandwidth
- Message types:
  - SR: Sender Report (data and time)
  - RR: Receiver Report (quality)
  - SDES: Source Descriptor with CNAME describing the sender (user@machine) + evt. other data (telephone etc)
  - APP: Application specific data
  - BYE:
- Several messages in same UDP packet possible

RTCP Sender Report

Byte: 0 1 2 3

Bit:: 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1

	ver =2	P	RR-count	Payload type = 200	lengde
	SSRC-identifikator for den som sender rapporten				
Sender-info	NTP timestamp (høyeste 32 bit)				
	NTP timestamp (laveste 32 bit)				
	RTP timestamp				
	antall sendte pakker				
	antall sendte byte				
nr 1	RR-rapporter				
	…..				

RTCP Receiver Report

Byte: 0 1 2 3

Bit:: 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1

	ver =2	P	Report count	Payload type = 201	lengde
	SSRC-identifikator for den som sender rapporten				
Mottaker-Rapport nr 1	SSRC-identifikator for første rapport				
	andel tapt * 256			Kumulativt antall pakker tapt	
	Høyeste pakke mottatt (16 bit) og antall ganger overflyt på tallet (16 b)				
	Arrival jitter				
	Timestamp i siste Sender report (SR) mottatt for denne kilde				
	Tid siden siste SR i 1/65536 sekund				
nr 2	ssrc rapport nr 2				
	…..				

RTP payload

- Application level framing principle
  - Packetize at natural breakpoints to reduce effects of packet loss
- Number assignment
  - static: RTP profile documents, RFC 1890
  - dynamic: 96-127, negotiated per session
- Each payload
  - header + decoded frame

RTP supported codecs

- http://www.iana.org/assignments/rtp-parameters
- Format: Number Name Media Clock Channel Std.

- 0 PCMU A 8000 1 [RFC1890]
- 1 1016 A 8000 1 [RFC1890]
- 2 G726-32 A 8000 1 [RFC1890]
- 3 GSM A 8000 1 [RFC1890]
- 4 G723 A 8000 1 [Kumar]
- 5 DVI4 A 8000 1 [RFC1890]
- 6 DVI4 A 16000 1 [RFC1890]
- 7 LPC A 8000 1 [RFC1890]
- 8 PCMA A 8000 1 [RFC1890]
- 9 G722 A 8000 1 [RFC1890]
- 10 L16 A 44100 2 [RFC1890]
- 11 L16 A 44100 1 [RFC1890]
- 12 QCELP A 8000 1
- 14 MPA A 90000 [RFC1890,2250]
- 15 G728 A 8000 1 [RFC1890]
- 16 DVI4 A 11025 1 [DiPol]
- 17 DVI4 A 22050 1 [DiPol]
- 18 G729 A 8000 1

- dyn GSM-HR A 8000 1
- dyn GSM-EFR A 8000 1
- dyn L8 A var. var.
- dyn RED A

- dyn VDVI A var. 1

- 25 CelB V 90000 [RFC2029]
- 26 JPEG V 90000 [RFC2435]
- 28 nv V 90000 [RFC1890]
- 31 H261 V 90000 [RFC2032]
- 32 MPV V 90000 [RFC2250]
- 33 MP2T AV 90000 [RFC2250]
- 34 H263 V 90000 [Zhu]

- dyn BT656 V 90000
- dyn H263-1998 V 90000
- dyn MP1S V 90000
- dyn MP2P V 90000
- dyn BMPEG V 90000

Case: H.263

- IETF RFC 2429
- Framing at picture segments (GOB/slice), ideally one segment per packet
- Payload-specific header (16 bits):
  - Reserved (5):
  - P (1):
    - picture start
    - picture segment start
    - video sequence end
  - V (1): video redundancy coding extension header
  - PLEN (6): length of extra picture header
  - PEBIT (3): number of bits ignored in last byte of picture header

Case: MPEG

- IETF RFC 2250
- Encapsulation modes
  - Elementary streams for MPEG1/2 A/V
  - Transport streams for MPEG-2
  - System streams for MPEG-1 (no ALF)
- ES framing at
  - video sequence header
  - GOP header
  - picture header
- Payload-specific header (32 bits)

Example: Internet TV

- HiØ: MPEG-2/DVB streaming
- Program: BBC World (NRK)
- Linux platform
- http://158.36.47.165/files/dvb-test_hiof.tar.gz

Wireless networks

- Satellite and mobile communication
  - Now packet based (GPRS, UMTS)
- Wireless characteristics
  - Error-prone media; rain, fading, handoff
  - Higher BER -> more packet loss, not congestion
  - Bursty traffic
  - Low bandwidth- > robust header compression
- Always transmission errors:
  - tradeoff: channel-coding redundancy and source coding redundancy
- Active research area

Mobile multimedia

- Hype or reality?
  - best compression = wavelet coding?
  - bandwidth still too small for streaming?
  - what kind of applications?
  - will people pay?
- Example: GMN project at NR
  - streaming multimedia over GSM

News application
Interactive Mobile News

The Communicator Device

- 133MHz processor
- RAM, ROM, EPROM (16MB)
- EPOC
- No disk or external device
- GSM (9600 bps), TCP/IP stack
- Colour screen (1/4 VGA)
- Sound

References

- “Compressed Video over Networks”, Sun & Reibman (eds.), Marcel Dekker, 2001

- Big thanks to Eirik Maus for some of the slides!