Protocols for multimedia in the Internet

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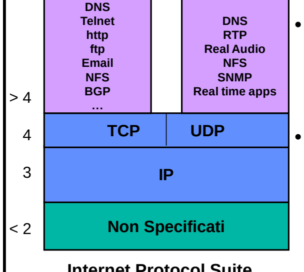
Applications and protocol stack



Internet Protocol Suite

- Which transport protocol?
- UDP is suited for:
  - Request-response (LAN)
  - Multimedia applications
  - Multicast
- TCP (reliability) is suited for:
  - File transfer
  - Terminal emulation
  - Request-response (WAN)
  - Unicast

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UDP protocol

- UDP (User Datagram Protocol) permits application to application (host to host) communication through datagram transmission
- UDP provides a layer 4 service:
  - Connectionless (out of sequence packets)
  - Unreliable (packet lost)
  - Low overhead (slim header)
    - Optional checksum
- Application identification through :
  - Source IP address, destination IP address, source UDP port, destination UDP port
  - No rate control
    - No flow control (possible receiver saturation)
    - No congestion control

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UDP packet format

0 … 15	16 … 31
UDP Source Port	UDP Destination Port
UDP Message Length	UDP Checksum (opt.)
DATA	

32 bit

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TCP protocol

- TCP (Transmission Control Protocol) is the other Internet layer 4 protocol
- Main characteristics
  - Connection-oriented
    - full-duplex
  - Reliable and in sequence delivery
    - Retransmission
  - Rate control
    - Flow controlled by receiver
    - Congestion control to avoid network saturation

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TCP packet header

0 … 15			16 … 31
Source Port Number			Destin. Port Number
Sequence Number			
Acknowledgement Number			
HLEN	Resv.	flags	Receiver window
Checksum			Urgent Pointer

32 bit

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Problems for multimedia

- Packetization
  - Voice sample of few bit
  - Single image has very large size
- How to distinguish at the receiver among different coding techniques?
- How to compensate for IP limitations?
  - Packet losses
  - Out of order delivery
  - Packet duplication
- How to notify to the source the correct reception of data?
- How to deal with multicast?

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Using TCP for multimedia?

- TCP is reliable, but
  - Retransmissions cause delays
- TCP is rate controlled to avoid receiver and network congestion, but
  - The available bit rate for the multimedia application is highly variable
- TCP does not support multicast
- TCP cannot be used for real-time multimedia
  - Non real time multimedia can be treated as file transfer

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UDP for multimedia

- UDP supports multicast, is not rate controlled and does not use retransmissions, but
  - Does not guarantees in-order delivery
  - Does not detect and deals with packet losses
  - Does not compensate for delay fluctuations
  - Does not recognize multimedia contents
- IETF proposal: add RTP protocol over UDP

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Example of multimedia application

- IP telephony: three different problems
  - H.323 / SIP: Establish multimedia connection, find IP addresses (possibly multicast), negotiate the type of coding and/or compression scheme, possibly inter-operate with the telephone network
  - RTP: Once the connection has been established, transfer audio packets
  - RTCP: Periodically send feedback information to the transmitter (and to receivers) to indicate the quality of the multimedia connection

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RTP: Real-time Transport Protocol

- Defined in RFC 1889
  - a framework to build multimedia applications
- Provides some basic mechanisms for multimedia data transfer
- It is not an independent protocol, but must be included in the application (no API defined)
- Composed by two different (although related) protocols:
  - RTP: deals with multimedia data transport (even UDP ports)
  - RTCP: provides control and monitoring services (odd UDP ports)
    - Feedback on packet delay and losses
    - Request/response to coding modifications
    - Helps in the management of the list of participants

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RTP + RTCP

- Functions available
  - Payload identification (coding)
  - Sequence number management
  - Timestamp management
  - Monitoring and performance analysis
  - Participants identification
- Functions NOT available
  - No QoS support
    - No reliability, bit rate or delay guarantees
  - No guarantee on correct and in order delivery
    - Exploits UDP checksum to detect errors

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RTP: packet losses

- UDP/IP do not guarantee zero losses
  - Packets may be lost or delivered not in order
- RTP provides sequence numbers in the RTP header to detect out of order delivery

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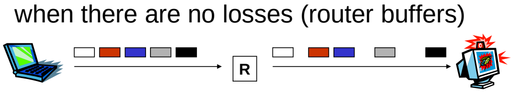
RTP: delay jitter

- When the multimedia source sends synchronous signals (e.g., voice), one packet is sent every T seconds (source clock)
- The network introduces variable delays even when there are no losses (router buffers)



- How to recover the synchronous signal at the receiver?

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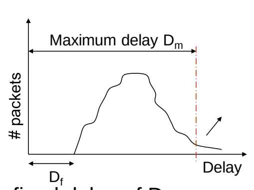
RTP: delay jitter

- Possible solution: introduce a delay at the receiver
- Use a playback buffer
  - Received packets are stored in the buffer
  - One packet every T seconds is played out



  - The buffer size emulates a fixed delay of D_m seconds
  - D_m is a compromise between low delays and low losses

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RTP: delay jitter

- If the source uses silence suppression?
  - During talk spurt: one packet every T seconds
  - During silence: no packets
- … a packet may be delayed because
  - the network has delayed it
  - it was preceded by a silence suppressed
- The sequence number is not enough
- A 'timestamp' field (conventional number which increases at the transmitter according to the source clock) is needed in the RTP header to recover information on the generation instant of a packet
  - Timestamp cannot be used as sequence number, since many packets may have the same timestamp (e.g., a video frame generates several packets with the same timestamp)

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RTP: applications

- Applications using RTP may be multicast
- If the network does not support multicast addressing and routing
  - A unicast connection among each pair of participants to the multicast session should be opened
  - The number of connections increases quadratically with respect to the number of users
- An application must specify two RTP parameters:
  - RTP profile
    - A table that associates to each type of payload (coding) a unique code
  - How RTP should use the payload
    - Information such as RTP packet size, number of samples contained in each packet, etc

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RTP: example of voice transmission

- Consider a multi-user voice conference over RTP
- Basic elements:
  - One IP multicast address
    - Users must register the IP multicast address to access the service
  - Two UDP ports
    - One even port for RTP, the next one (odd) for RTCP
  - RTP does not specify how to choose these numbers
- Voice is produced according to a proper coding technique
- Voice samples are grouped in packets
  - Sent to the multicast address
  - The number of samples in each packet depends on the RTP configuration

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RTP: example of voice transmission

- Packets size should be small to keep under control the packetization delay (should be smaller than few tens of ms)
  - Few samples in each packet
- Samples are encapsulated within an IP + UDP + RTP headers

IP header	UDP header	RTP header	Samples

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RTP: example of voice transmission

- Through RTCP, each participants sends (in multicast) statistical data
  - It is possible to analyze service performance
    - Code rate adaptation may be envisioned to adapt the transmission to the measured quality
- Since the RTCP traffic is sent in multicast to all participants (and is generated by all participants) the required bit rate may be significant and should be kept under control

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RTP: example of audio-video conference

- The standard suggests to use two independent RTP flows
  - Advantage: a user may access only one of the two services
- It is mandatory to synchronize the two flows:
  - The RTP timestamps of the two flows may be used with this goal

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RTP: terms and definitions

- An RTP **session**
  - Allows the communication among a set of users through RTP
  - Is identified by a pair of transport addresses
    - A transport address is a IP address-UDP port pair
    - One transport address is used by RTP, the other one for RTCP
    - The IP multicast address (if used) is the same for both pairs
- The **host** or **end-system** (ES) is the user system where applications based on RTP are running. Hosts generate and receive multimedia flows

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RTP: terms and definitions

- An RTP **translator** is a coding translator
  - Modify the data coding and retransmit the modified data flow over the session
  - May also operate as gateways
  - Permit the service provisioning also over non IP islands
- An RTP **mixer** is an RTP flow aggregator
  - More RTP flows in input are mixed and generate a new single RTP flow at output
    - Typically, the bit rate of the new flow is smaller that the sum of the bit rate of original flows
  - Devices more complex than translator
    - Correlated flow synchronization must be dealt with

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RTP: terms and definitions

- RTP **monitors** observe externally the control packet flow (RTCP packets)
  - Do not participate in the RTP transmission/reception process
  - Collect information on the QoS of the RTP session
  - Useful for network providers to control the network service quality

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RTP: terms and definitions

- The **SSRC** (Synchronization SouRCe) is the unique identifier of the data flow generator
  - Is a 32 bit number contained in the RTP header
  - Warning: mixers are characterized by an own SSRC
    - The output data flow from a mixer is a new flow, with its proper timestamp
  - Within an RTP session, each SSRC must be unique
- Since it may be useful or needed to recover the original source of a mixed RTP flow
  - The **CSRC** (Contributing SouRCe) are fields optionally contained in the RTP header that contain the SSRCs of the original sources of the RTP mixed flow

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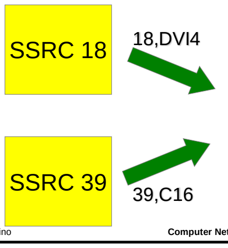
RTP: Host

- The first host creates an RTP flow with SSRC 18 and DVI4 code, the second host sends RTP data with SSRC 39 and C16 code



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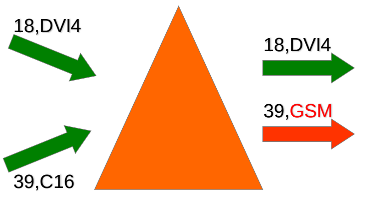
RTP: Translator

- The translator modifies the coding of the flow with SSRC 39 from C16 to GSM



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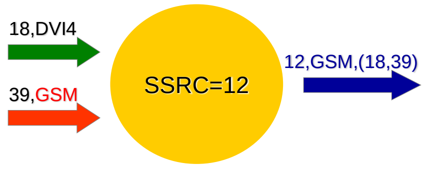
RTP: Mixer

- The RTP flow generated by the mixer has SSRC 12 and CSRC 18 e 39



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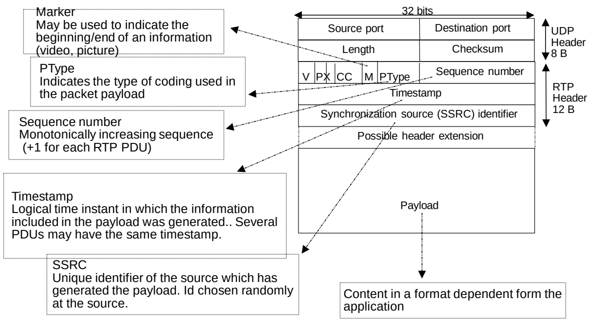
RTP packet format



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RTP header

- V (Version) (2bit)
  - Version of the RTP protocol
- P (padding) (1bit)
  - Padding contained in the payload
- X (eXtension header) (1bit)
  - Proprietary
- CC (CSRC count) (4 bit)
  - Number of CSRC field contained in the header
- Marker (1 bit)
  - Use dependent on the RTP session profile
- PT (Payload Type) (7 bit)
  - Type of coding used in the packet payload

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RTP header

- Sequence number (16 bit)
  - The initial sequence number X is chosen randomly at session startup
  - X is inserted in the sequence number field of the first generated RTP packet
  - The second packet will have sequence number X+1, the third X+2...
    - The random extraction minimizes the probability of choosing the same number previously selected in an older RTP session (which may have some packets still in the network)

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RTP header

- Timestamp (32 bit)
  - Represents the “logical” time instant in which the packet payload was generated. More packets my have the same timestamp value
  - The first timestamp value is randomly selected at RTP session startup
  - The timestamp refers to the creation time instant of the first sample contained in the packet
  - Example: if each packet of an RTP phone session contains 160 samples:
    - If packet I has timestamp X, then packet I+1 will have timestamp X+160

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RTP header

- SSRC (32 bit)
  - Identifier of the source that has created the RTP payload. Randomly chosen at the source at session startup. Algorithm defined to solve contentions.
- CSRC (32 bit)
  - Optional field
  - Identifier of the sources that have originally created the payload from which the current payload was derived through a mixing operation

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RTCP: objectives

- Quality of service and congestion control
  - RTCP packets are used as “low frequency” ACKs to signal the reception quality
  - On the basis of RTCP “report”, the server may adapt the coding to the communication status
- Identification
  - Provide identification information on RTP session participants (signalling)
- Estimate the number of participants to the multicast session
  - Needed to control the transmission bit rate required by RTCP control signals which would increase too much if the number of participants increases too much

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RTP Control Protocol (RTCP)

- Control protocol for RTP data flow
- Defines the information exchange among the source and the destinations
- Several types of RTCP packets:
  - SR (Sender Report): sent by all active sources to all participants; includes TX and RX statistics
  - RR (Receiver Report): sent by all receivers to all participants; includes RX statistics
  - SDES (Source DEScriptor): source description through a unique identifier
  - BYE: session ends or one participant leaves the session
  - APP: application-specific

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RTCP Sender Report

- SR is used to provide information on data recently sent
- A SR contains:
  - An absolute timestamp (NTP timestamp) of the data sending time
  - Relative timestamp referring to the current RT flow
  - Amount of data sent from RTP session start-up
    - Total number of RTP packets sent
    - Total number of byte sent

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RTCP Receiver Report

- RR are sent to inform senders on the quality of the RTP session as seen by receivers
- A RR is sent to each source from which a SR was received
- A RR contains:
  - Identification of the received source
  - Timestamp of the last received SR
  - Delay from the reception of the last SR
  - Highest sequence number received from the source
  - Number of lost RTP packets
  - Fraction of lost RTP packets
  - Estimate of RTP packets jitter

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RTCP SDES

- Used by sources and destinations to identify themselves
- An SDES may contain:
  - CNAME: user identifier (user@host.domain)
  - NAME: name of the person using the application
  - EMAIL
  - PHONE
  - LOC: geographical location of the user
  - TOOL: application using RTP
  - NOTES

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RTCP: report transmission speed

- Four rules must be followed to generate RTCP packets
  - The RTCP traffic should be limited to a given percentage of the data traffic (5%)
  - 25% of the session bit rate is devoted to SR packets, the remaining part to other packets
  - An RTCP packet cannot be sent earlier than 5s after the previous RTCP packet transmission
  - A variable time P should be added to the waiting time

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RTCP: report transmission speed

- P is computed as a random number uniformly chosen between 0.5 e 15, multiplied by T_{sr}
- The period for transmitting RTCP packets for the transmitter

$$T_{SR} = \frac{\text{Num\_senders}}{0.25 \times 0.05 \times \text{Session\_rate}} \times \text{Avr\_RTCP\_pkt\_size}$$