ΟΙΚΟΝΟΜΙΚΟ ΠΑΝΕΠΙΣΤΗΜΙΟ ΑΘΗΝΩΝ

ATHENS UNIVERSITY OF ECONOMICS AND BUSINESS

Multimedia Technology

Section # 17: Networks & Transport

Instructor: George Xylomenos

Department: Informatics

Contents

- Internet structure
- Network applications
- Basical protocols
- RTP
- RTCP

Internet structure

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Internet structure (1 of 6)

- Organizational / corporate networks
  - Optical fiber connection to the Internet
    - Possibly, with reserve connection
  - Central router
  - Firewall enforcing Demilitarized Zone (DMZ)
    - Externally visible servers
    - Internal access network
    - Controlled access between zones

Internet structure (2 of 6)

- Organizational / corporate networks
  - Usually, access is via Ethernet
    - Optical backbone (vertical)
    - UTP (Cat 6) access (horizontal
    - Increasingly, WiFi along with / instead of Ethernet
  - Switching rather than routing
    - Network broken down into VLANs
    - Router connects VLANs together

Internet structure (3 of 6)

- Small business / home networks
  - Copper connection to Internet
  - DSL in most countries
    - Based on existing telephony network
    - Each client has separate line
  - HFC wherever cable TV exists
    - Based on cable TV network
    - Multiple homes share a loop
  - Gradual introduction of fiber to the home

Internet structure (4 of 6)

- Small business / home networks
  - Modem connected to Internet
    - Includes router, firewall and switch
    - May split voice or TV service
    - NAT for internal clients
      - Hard to have externally visible server
  - WiFi and/or Ethernet internally
    - Usually one large LAN
    - Repeaters used to expand WiFi

Internet structure (5 of 6)

- Internet Service Providers (ISPs)
  - Hierarchical access network
    - Copper cables up to local concentrator
    - Optical fiber between concentrator and DSLAM
  - DSLAM (DSL Access Multiplexer)
    - Splits telephony and Internet signals
    - Connects to two different networks
    - Gradually, telephony turns to VoIP

Internet structure (6 of 6)

- Tier 1 ISPs
  - Optical network between points of presence
  - Usually grid rather than tree
  - POPs located in different cities
  - Regional / local ISPs connected to POPs
  - Regional / local ISPs may be connected directly
    - Using direct (peering) links
    - Avoid payment to Tier 1 ISPs

Who pays for what? (1 of 4)

- Money flows to the top!
- A client/company pays an ISP
  - For Internet or double/triple play
- The ISP pays the cable owner
  - One cable reaches each home
    - Soon, one optical fiber
  - The ISP rents our line from its owner
    - May have its own fibers between concentrators

Who pays for what? (2 of 4)

- ISPs pay the Tier 1 providers
  - May also work hierarchycally
    - Local ISP pays Regional ISP who pays Tier 1
  - Typically volume based pricing
    - But, ISPs typically offer flat rates!
    - So, ISPs try to reduce traffic
    - Peering links to other ISP
    - Use of caches in the network

Who pays for what? (3 of 4)

- Who pays for video, specifically?
  - Creates largest traffic volume
  - Forces ISPs to increase speeds
- The easy case: IPTV
  - Service provided by an ISP
  - Usually over a custom network
    - Multicasting, VLANs, STBs
  - Included in monthly fee

Who pays for what? (4 of 4)

- The hard case: streaming
  - User buys a NetFlix subscription
    - Covers royalties and NetFlix infrastructure
  - But no money flows to the ISP!
    - Of course, NetFlix pays their ISP (but not ours!)
  - But our ISP needs to invest in the network
    - Or the customer may leave
  - How can the ISP get compensated?

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Network applications

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Network applications (1 of 4)

- Networked multimedia applications
  - Sensitive to delay
    - Possibly also to delay jitter
  - Tolerant to errors
    - It is better to sacrifice some quality
    - Rather than retransmit lost data
  - Minimum transmission rate required
    - Some are elastic on that (multiple rates allowed)

Network applications (2 of 4)

- Streaming (stored) media
  - Streaming: playback in parallel with reception
  - Interactivity with user
    - Pause, play, forward, backward
    - Example: NetFlix
  - Each user is different
    - Must be server separately

Network applications (3 of 4)

- Media streaming with multiple receivers
  - Similar to live TV
    - May be live or stored content
    - The issue is whether there are many viewers
  - No interactivity
    - Example: Cosmote TV
  - All users treated in the same way

Network applications (4 of 4)

- Multimedia interaction
  - Communication between users
    - Not between server and user
  - Requires low end-to-end latency
    - Otherwise, application suffers
    - Example: Zoom
  - Users are generally heterogeneous

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Basic protocols

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Basic protocols (1 of 5)

- Network layer: IP
  - Best effort service
  - Unknown delay and reliability
    - Packets may be lost or reordered
    - Depending on traffic and routing
  - In principle, packets up to 64 KB
    - Usually, 1.5 KB to avoid fragmentation
    - Must fit in an Ethernet frame!

Basic protocols (2 of 5)

- Transport layer: UDP
  - Multiplexes transport flows over IP
    - Basically, IP with transport layer ports
    - Optional checksum (IP does not have one)
  - Anything else, left to applications
    - Often used by multimedia applications
    - Application layer congestion control
    - Must be friendly to TCP

Basic protocols (3 of 5)

- Transport layer: TCP
  - Flow, congestion and error control
  - Reliable transport
    - But, delay unpredictable due to retransmissions
  - Transmission rate depends on congestion
    - Plus, flow control (TCP window)
    - Not controlled by application
    - Fluctuates depending on TCP's policies

Basic protocols (4 of 5)

- Application layer: HTTP
  - In principle, for web pages
  - But, web pages can contain anything!
    - Images, sound, video, text
    - MIME types used for description
  - Can be used for multimedia application
  - Although it runs over TCP
    - Which is reliable but unpredictable

Basic protocols (5 of 5)

- The “curse” of HTTP
  - The web made the Internet
  - HTTP crosses firewalls by default
    - TCP allows monitoring connections
  - Nearly nothing else is allowed
    - Especially UDP that has no connections
  - So, we put everything inside HTTP!
    - Even multimedia (e.g., MPEG DASH)

RTP

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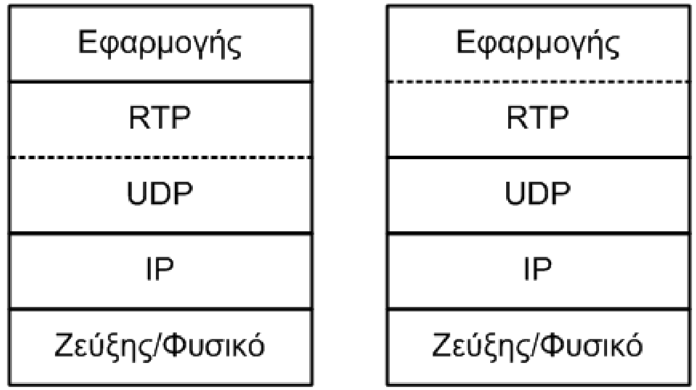
What is RTP? (1 of 2)

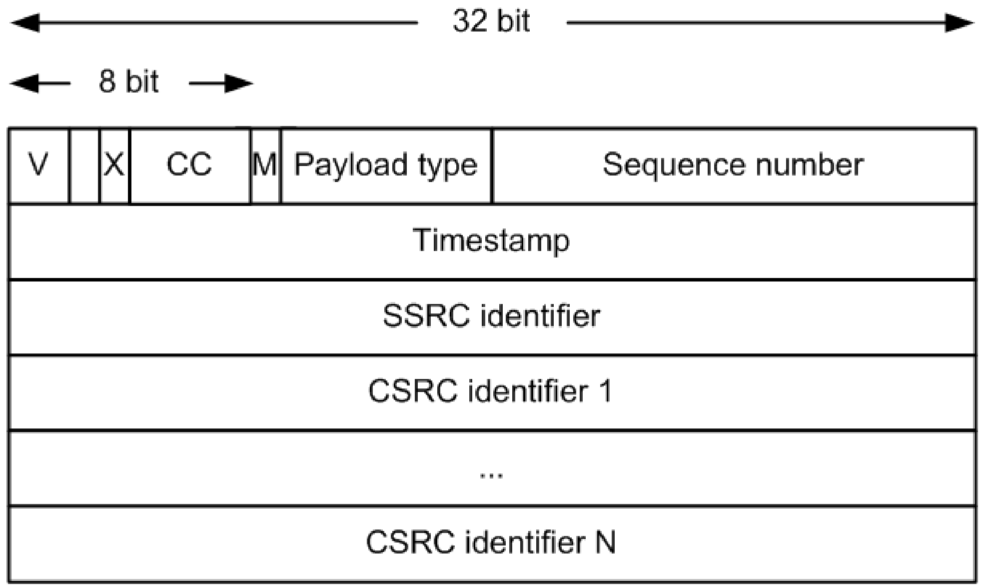


- RTP: Real-time Transport “Protocol”
  - Standardized media header
    - Allows applications to interoperate
    - Implemented by an application library
  - Media encapsulated in RTP packet
  - RTP packet encapsulated in UDP

What is RTP? (2 of 2)

- RTP is not a transport protocol
  - It is part of the application
  - The application actually does packetization
- RTP has no functionality
  - It is just a header
  - Every application uses it as it deems fit
    - It may be used for error control
    - But it is the application's job

RTP packets (1 of 4)



- Every media source creates an RTP flow
  - Flows are unidirectional packet streams
  - This is the Contributing Source (CSRC)
- RTP session
  - All RTP flows used by an application

RTP packets (2 of 4)

- Payload type: 7 bit
  - Can be changed during the session
  - Types defined for different media
    - With a specific encoding
- Sequence number: 16 bit
  - Counts packets in the stream
  - Can be used for error detection
  - Or to reorder packets at the receiver

RTP packets (3 of 4)

- Timestamp: 32 bit
  - Time of first sample in the packet
  - Based on sender’s clock
    - Sampling clock, not real time clock
  - Complements the sequence number
    - We may have gaps due to inactivity (e.g., silence)
- V: protocol version
- X: extended header

RTP packets (4 of 4)

- Synchronization source (SSRC): 32 bit
  - A mixer that combines multiple sources
  - Inserts its own timestamps
- Contributing sources (CSRC): 32 bit
  - CC shows how many there are
- M: Marker bit
  - Meaning depends on application
  - Example: marks last packet in a video frame

RTCP

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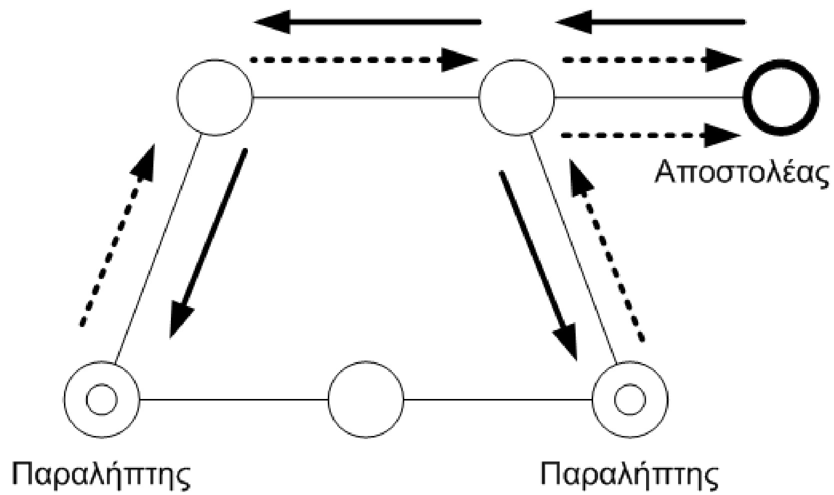
RTP and RTCP



- Real-Time Control Protocol (RTCP)
  – Monitoring of media flows
  – Also applicable to multicast
- RTCP packets sent periodically
  – In RTP, only senders transmit
  – In RTCP, both senders and receivers transmit

RTCP packets (1 of 3)

- Transmission report
  - Separate for each sender's source
    - Example: separate for audio and video
  - Identified by Synchronization Source ID
  - Timestamp / real time of last packet sent
  - Packet and bytes sent

RTCP packets (2 of 3)

- Reception report
  - Separate for each source received
  - Identified by Synchronization Source ID
  - Packet loss rate (percentage)
  - Last sequence number received
  - Delay jitter between received packets

RTCP packets (3 of 3)

- Source description (by sender)
  - Again, separate for each source
  - General information about the source
    - E-mail and name of sender, application
    - Synchronization Source ID
  - Ties user with an SSRC
    - Which is used for all other reports

Exploiting RTCP (1 of 2)

- Exploiting RTCP statistics
  - The protocol does not mandate behavior
    - Each endpoint can react as it sees fit
    - May adjust transmission rate
    - May try to diagnose problems
  - RTCP packets can contain multiple reports
    - For all the sources of a sender / receiver

Exploiting RTCP (2 of 2)

- Synchronizing packet flows
  - Timestamps: based on sampling clocks
  - No relation to real time clock
- Exploiting transmission reports
  - Timestamp and real time of last packet
    - Allows translating sampling time to real time
    - Makes possible cross-media synchronization

RTCP scaling (1 of 2)

- RTCP is susceptible to feedback implosion!
  - RTP is used by (a few) senders
  - RTCP is also used by (many) receivers
  - RTCP traffic can be higher than RTP traffic!
- Adjusting RTCP transmission rate
  - Inversely proportional to senders / receivers
  - Each participant estimates how many
    - By looking intro sender / receiver reports

RTCP scaling (2 of 2)

- Adjusting RTCP transmission rate
  - Traffic should be 5% RTCP - 95% RTP
  - And 75% for receivers - 25% for senders
    - Participants have to share these rates
- Calculating transmission period
  - L: packet size, B: total bandwidth for app

$$T_S = \frac{N_S \cdot L}{0{,}25 \cdot 0{,}05 \cdot B} \qquad T_R = \frac{N_R \cdot L}{0{,}75 \cdot 0{,}05 \cdot B}$$

End of Section # 17

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