S1

COMP28411 Computer Networks
Lecture 8

Nick Filer

Multimedia - 1

Some material from:

Kurose & Rose – Chapter 7 + Slides

Halsall – Multimedia Communications

Overview

- What is it?
- How do we get it?
  - Telephone networks
  - Data networks
  - Broadcast networks
- Overview of Delivery Methods
- Buffering
- Real-Time Constraints and Issues
- Delivery of VoIP and home media.

What is Multi-Media?

- Your ideas?
- It seems, anything you can store, transfer, experience?

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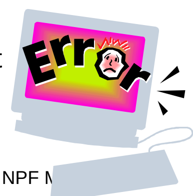
MM Issues

A3

- Animations, Movies & sounds – basic form is often analogue.
  - To mix with text & images must convert to digital!
  - And back to analogue?
- Duration of audio/video is often quite long.
- For real-time usage (conversation, interaction) – Very delay sensitive!
- Unlike web, email & some business data:
  - L t – t ome extent.
  - roErr loartent - to some extent



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MM Issues

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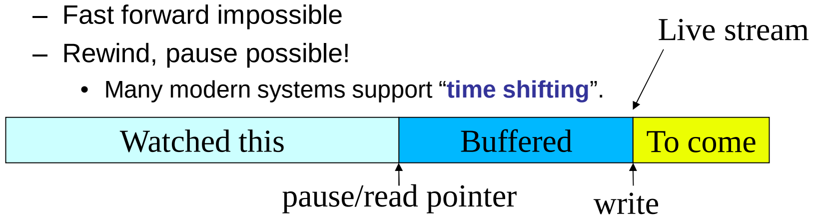
Streaming *Live* Multimedia 1:Many

- Examples:
  - Internet radio talk show
  - Live sporting event
- Streaming (as with streaming *stored* multimedia)
  - Sending + Routers + Playback buffers.
  - Playback can lag tens of seconds after transmission
  - Still have timing constraint – waits – slow links / congestion
- Interactivity
  - Fast forward impossible
  - Rewind, pause possible!
    - Many modern systems support "**time shifting**".



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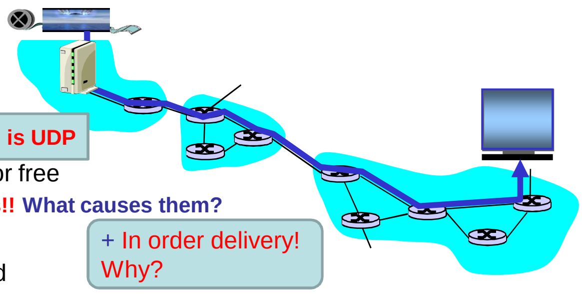
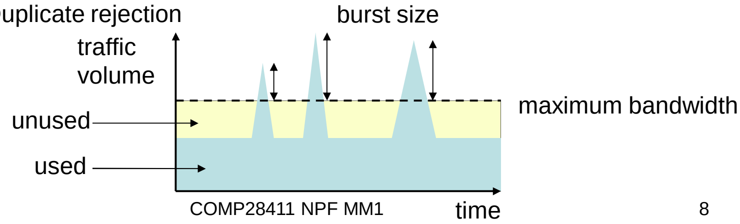
A1

Media Transfer

- Move a big file from A to Z. 1:1 transfer.
- How?
  - TCP - Alternative is UDP
    - No losses/Error free
      - But delays!! What causes them?
    - Best effort
    - Variable speed
    - Duplicate rejection

+ In order delivery! Why?





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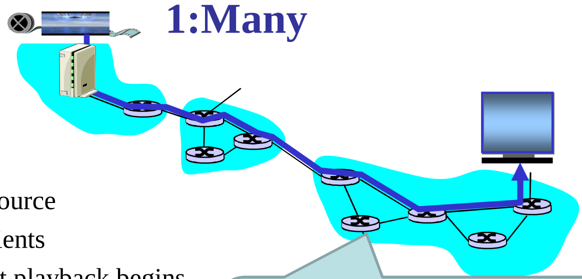
Streaming Stored Multimedia 1:Many



- Stored streaming:
  - Media stored at source
  - Transmitted to clients
  - **Streaming:** client playback begins *before* all data has arrived
- *VCR-like functionality:* client can pause, rewind, FF, push slider bar
  - 10 sec initial delay OK
  - 1-2 sec until last command effect OK
  - Timing constraint for still-to-be transmitted data:
    - **Must be in time for playback**

Could cache the data locally. Less far to send – See Content Delivery later. 1:Many via **multicast** vs **unicast**

How are multiple requests handled? Can you do multicast and stop, restart, jump back, jump forward? Can you handle the no. of unicast connections?

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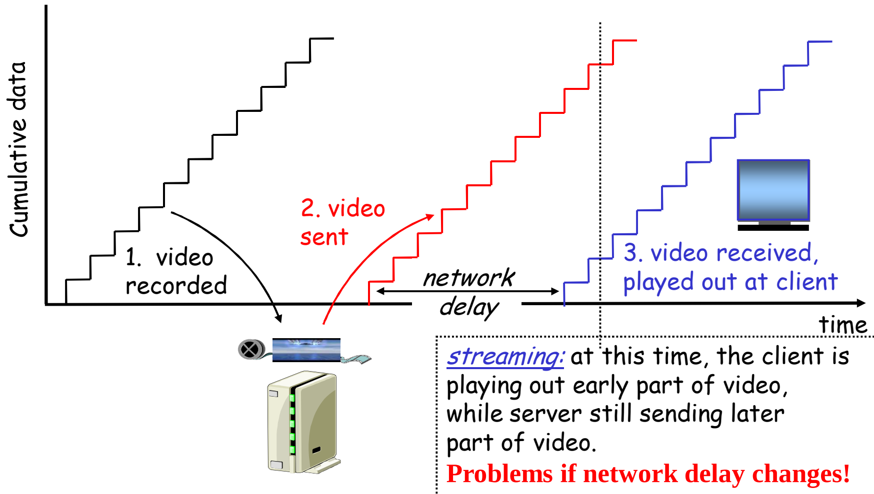
Streaming Stored Multimedia: What is it?

A5

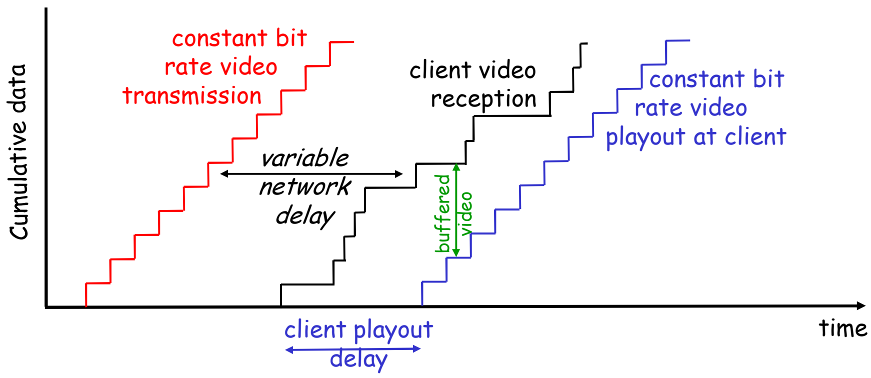


streaming: at this time, the client is playing out early part of video, while server still sending later part of video.

Problems if network delay changes!

Streaming Multimedia: Client Buffering

A4



- Client-side buffering, play out delay compensates for network-added delay & jitter (variations in delays).

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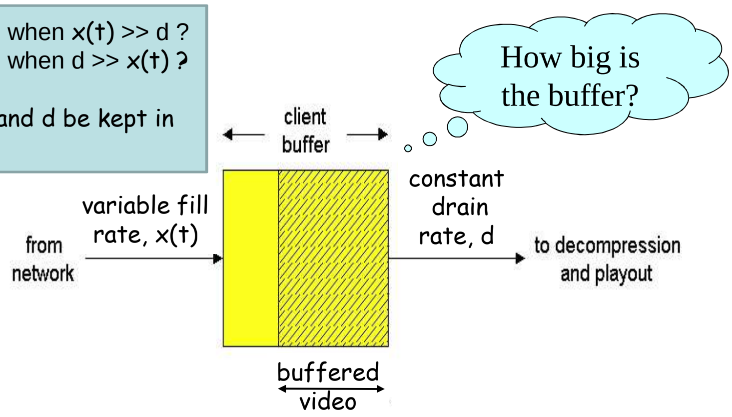
Streaming Multimedia: Client Buffering

What happens when x(t) >> d ?
What happens when d >> x(t) ?

How can x(t) and d be kept in balance?



- Client-side buffering, play out delay compensates for network-added delay and jitter (variations in delay).
- Sometimes the buffer has to be re-filled from empty.

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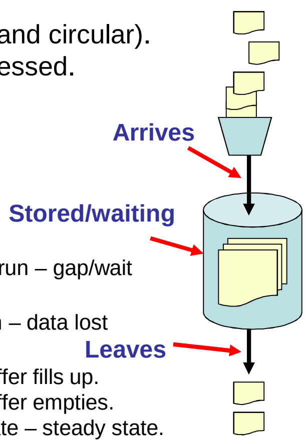
Buffers – A Reminder/Revision Slide

- Arrays/tables, stacks, queues, lists (linear and circular).
- Used to store data/work waiting to be processed.
- Data/Work:
  - **Arrives**
  - **Waits**
  - **Leaves**
    - At a constant or variable rate/speed.
  - **If** buffer is **empty**:
    - Can arrive, cannot leave – underflow/under-run – gap/wait
  - **If** buffer is **full**:
    - Can leave, cannot arrive – overflow/over-run – data lost
  - **If** buffer is part full:
    - Data/work arrives slower than it leaves – buffer fills up.
    - Data/work leaves slower than it arrives – buffer empties.
    - Data/work arrives and leaves at the same rate – steady state.



- Multi-media play out buffer should never be over full or over empty whilst playing media.
- Hence need to adjust :- arrival rate, leaving rate, buffer size + adjustment thresholds (how full/empty before make a change).

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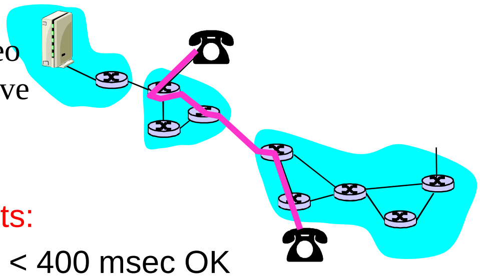
Real-Time Interactive Multimedia



- Applications: IP telephony, video conference, distributed interactive worlds.
- **Now 2 way!**
- Round-trip delay requirements:
  - Audio: < 150 msec good, < 400 msec OK
    - Includes application-level (packetization) and network delays
    - Higher delays noticeable, impair interactivity
- Session initialization
  - How do caller/called advertise their IP address, port number, encoding algorithms? ➔ SIP

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Interactive Multimedia: Internet Phone Using G711

Introduce Internet Phone by way of an example – Already seen this configuration! (last year!)

- Speaker's audio: alternating talk spurts, silent periods.
  - 64 kbps during talk spurt.
  - Packets generated only during talk spurts.
  - Typical, 20 ms chunks at 8 Kbytes/sec: 160 bytes data. Can be longer (e.g. 30ms)/shorter (e.g. 10ms)!
- Application-layer header added to each chunk.
- Chunk + header encapsulated into UDP segment.
- Application sends UDP segment into socket every (e.g.) 20 msec during talk spurt (**VBR**) or continuous (**CBR**). **V = Variable, C = Continuous, BR = Bit Rate**

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Internet Phone: Packet Loss and Delay

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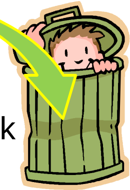

- Network loss: IP datagrams lost due to network congestion (router buffer overflow)
- Delay loss: IP datagrams arrive too late for play out at receiver
  - Delays (caused by?):
    - processing, queuing in network; end-system (sender, receiver) delays
  - Typical maximum tolerable delay: 400 ms

- Loss tolerance:
  - Depends on voice encoding.
  - How losses are concealed (speed up/down, play something in the gap from before/after, based on before & after).
  - Packet loss rates between 1% and 10% can be tolerated.
    - G711 with 5% loss has almost no degradation.

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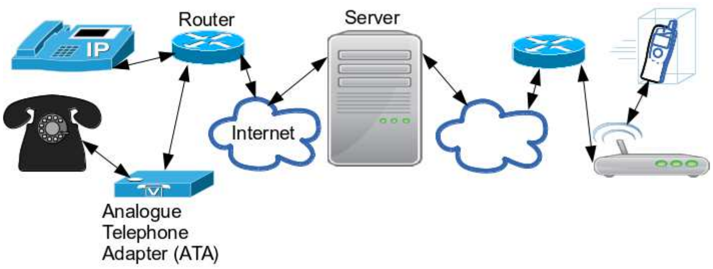
How do today's MM sessions work?

- SIP – Session Initiation Protocol
  - Inter-Asterisk-eXchange (IAX) or now IAX2 competes with SIP. IAX2 only uses 1 port, less bandwidth for control, binary not text. But it is not yet fully standard.
  - Application layer , not a service provider, other protocols provide services e.g. Real Time Protocol (RTP) to carry MM traffic.
  - Uses a client server model.



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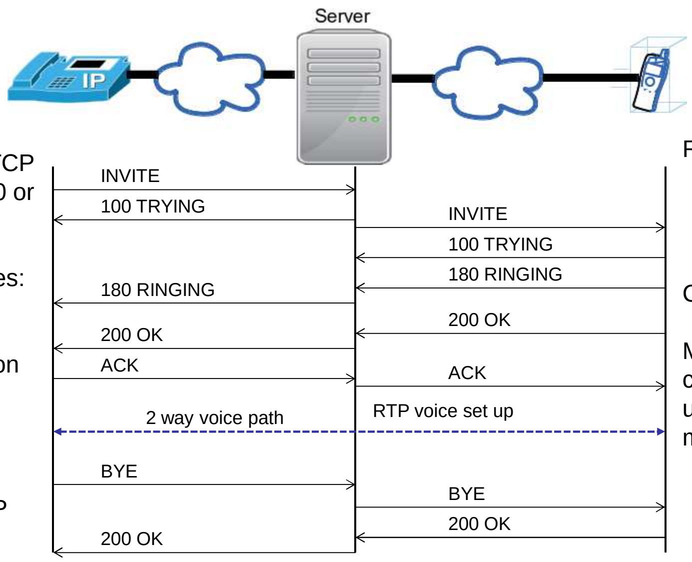
SIP Request/Response Transactions



Signalling:
- **UDP** or TCP
- Port 5060 or 5061 if 2 lines

Media types:
- Audio
- Video
- Application
- Data
- Control
- ...

Separate path to SIP signalling.

RTP:
- UDP
- Wide range of ports.

Can be TCP

Many codecs used for media

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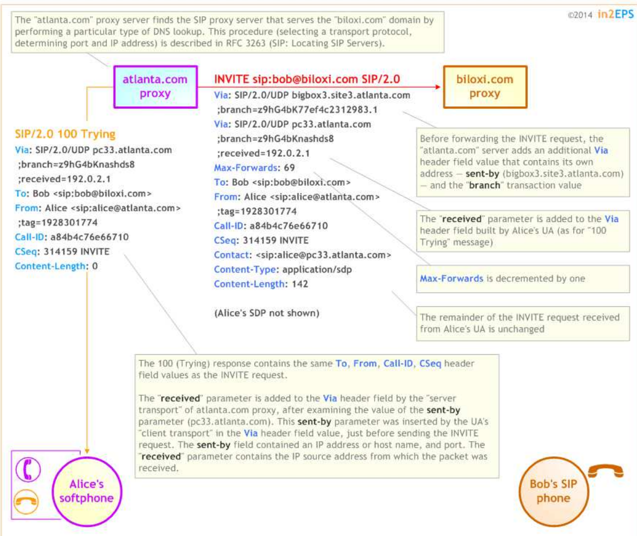
SIP is very verbose

- IAX/IAX2
  - No text parsing. Information element encoded.
  - Defined states for signalling and media.
  - Clean non ambiguous terminate on error.
  - All on 1 port!
  - No RTP, fewer bitts, great trunking.
- But:
  - SIP can send just about anything!



From: http://www.in2eps.com/fo-sip/tk-fo-sip-ex3261.html

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SIP Response Codes - 1

- 1xx – Informational
  - 100 trying
  - 180 Ringing
  - 181 Call is being forwarded
  - 182 Queued
  - 183 Session progress
- 2xx – Success
  - 200 OK
  - 202 Accepted: Used for referrals
- 3xx Redirection
  - 300 Multiple choices
  - 301 Moved permanently
  - 302 Moved temporarily
  - 305 Use proxy
  - 380 Alternative service
- 4xx – Failure
  - 400 bad request
  - 401 Unauthorized
  - 402 Payment required
  - 403 Forbidden
  - 404 Not found
  - 405 Method not allowed
  - 406 Not accepted
  - 407 Proxy authentication required
  - 408 Request timeout
  - 410 Gone
  - 413 Request entity too large
  - 414 Request URI too long
  - 415 Unsupported media type
  - 416 Unsupported URI scheme

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SIP Response Codes - 2

- 4xx – Failure (continued)
  - 420 Bad extension
  - 421 Extension required
  - 423 Interval too brief
  - 480 Temporarily unavailable
  - 481 Call/transaction does not exist
  - 482 Loop detected
  - 483 Too many hops
  - 484 Address incomplete
  - 485 Ambiguous
  - 486 Busy here
  - 487 Request terminated
  - 488 Not accepted here
  - 491 Request pending
  - 493 Undecipherable
- 5xx - Server errors
  - 500 Server internal error
  - 501 Not implemented
  - 502 Bad gateway
  - 503 Service unavailable
  - 504 Server timeout
  - 505 Version not supported
  - 513 Message too large
- 6xx Global failures
  - 600 Busy everywhere
  - 603 decline
  - 604 Does not exist anywhere
  - 606 Not acceptable

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RTP and Quality of Service (QoS)

Q

- **RTP** does **not** provide any mechanism to ensure timely data delivery or other QoS guarantees. **X**
- **RTP** encapsulation is only seen at end systems (not) by intermediate routers. **X?**
  - Routers providing best-effort service, make no special effort to ensure that RTP packets arrive at destination quickly. **X**
  - Modern router can sometimes carry out deep (internal) analysis of passing packets to adapt queuing etc. to the type of stream.
    - This is expensive in power and execution time.
    - ISPs now do it to detect e.g. adult material for filtering.

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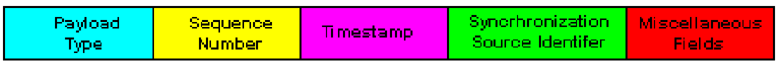
RTP Header



RTP Header

Payload Type (7 bits): Indicates type of encoding currently being used. If sender changes encoding in middle of conference, sender informs receiver via payload type field. For example,

- Payload type 0: PCM mu-law, 64 kbps
- Payload type 3, GSM, 13 kbps
- Payload type 7, LPC, 2.4 kbps
- Payload type 26, Motion JPEG
- Payload type 31. H.261
- Payload type 33, MPEG2 video

Sequence Number (16 bits): Increments by one for each RTP packet sent, and may be used to detect packet loss and to restore packet sequence. May cycle quite quickly – issue?

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RTP Header (2)

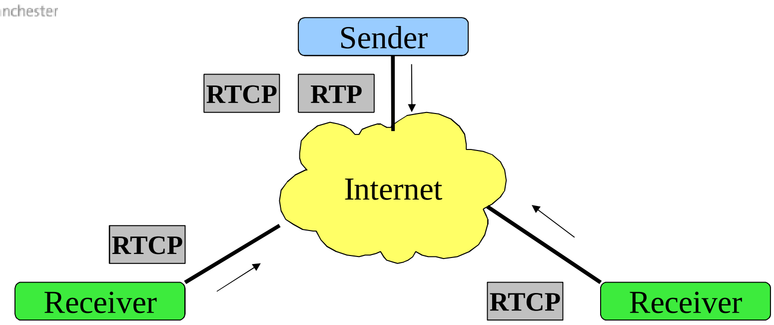
- **Timestamp field (32 bits long):** Sampling instant of first byte in this RTP data packet
  - For audio example, timestamp clock typically increments by one for each sampling period (for example, each 125 μsecs for PCM's 8 KHz sampling clock).
  - If an application generates chunks of 160 encoded samples, then timestamp increases by 160 for each RTP packet when source is active. Timestamp clock continues to increase at constant rate when source is inactive.
- **SSRC field (32 bits long):** Identifies source of the RTP stream.
  - Each source/sender in an RTP session should have a distinct SSRC.
    - Microphone + Webcam different SSRC if sent separate.
    - Same SSRC if e.g. sent as MPEG encoded sound + vision.
    - There are arguments about SSRC rules!

RTP Control Protocol (RTCP)

- Works in conjunction with RTP.
  - Each participant in RTP session periodically transmits RTCP control packets to all other participants.
  - Each RTCP packet contains sender and/or receiver reports
  - Report statistics useful to application: # packets sent, # packets lost, inter-arrival jitter, etc.
- Feedback can be used to control performance
  - Sender may modify its transmissions based on feedback

> Remember, RTSP is state based and controls **start**, **stop**, **pause**…but these commands are usually only usable with unicast streams not mutlicast.

RTCP - Continued



- Each RTP session:
  - Typically a single multicast address.
  - All RTP /RTCP packets belonging to session use the same multicast address.
- RTP and RTCP packets distinguished from each other via distinct port numbers to limit traffic.
- Each participant reduces RTCP traffic as number of conference participants increases.

RTCP Packets

Receiver report packets:

- Fraction of packets lost,
- Last sequence number,
- Average inter-arrival jitter

Sender report packets:

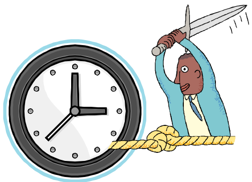
- SSRC of RTP stream.
- Current time.
- Number of packets sent.
- Number of bytes sent

Source Description Packets:

- E-mail address of sender.
- Sender's name.
- SSRC of associated RTP stream
  - Provide mapping between the SSRC and the user/host name

Synchronization of Streams

- RTCP can synchronize different media streams within an RTP session
- Consider a video-conferencing application for which each sender generates separate RTP streams for video and for audio.
- Timestamps in RTP packets tied to the video or audio sampling clocks.
  - **Not** tied to wall-clock time
- Each RTCP sender-report packet contains (for most recently generated packet in associated RTP stream):
  - Time-stamp of RTP packet
  - Wall-clock time for when packet was created.
- Receivers use association to synchronize play-out of audio and video.



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RTCP Bandwidth Scaling

- RTCP attempts to limit its traffic to 5% of session bandwidth.

Example

- Suppose one sender, sending video at 2 Mbps. Then RTCP attempts to limit its traffic to 100 Kbps.
- RTCP gives 75% of rate to receivers; remaining 25% to sender
- The 75 kbps is equally shared among receivers:
  - With R receivers, each receiver gets to send RTCP traffic at 75/R kbps.
- Sender gets to send RTCP traffic at 25 kbps.
- Participant determines RTCP packet transmission period by calculating the average RTCP packet size (across entire session) and dividing by allocated rate.

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Summary

- MultiMedia is text, still images, sound and video mixed together.
  - Sound and Video are different – e.g. delay sensitive.
- Send via:
  - Telephone - voice, music, fax….
  - Data network – everything
  - Broadcast TV – Video + Radio + some Data
  - Converged to : Broadband Multi Service networks
- Voice/Music over IP –
  - Buffering,
  - Jitter,
  - Fixed/Adaptive Play-out Delay.

Next Time
More on streaming.
Real-Time
Protocols

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Questions?

- What happens as the end to end delay between server and client continually shrinks?
- What happens as the end to end delay increases?
- Why might it be a bad idea to send live broadcast TV over the Internet?
  - Why do media companies do it anyway?
- What changes in network resources were necessary to allow HD video and sound to be transmitted?
- How is time-slip broadcast implemented e.g. so you can answer the phone and not miss any of your TV show?

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Questions? Things to thinks about.

- How would you change the equations for “adaptive play-out delay” to adapt rapidly to delay variations?
- Is RTP ever used or useful embedded in TCP packets?
- An audio stream samples at 4000Hz. Suggest a suitable timestamp clock increment gap for RTP to use for this audio stream?
- Often RTP SSRC values are chosen randomly. How might a receiver detect and resolve SSRC value collisions? – likely to need a little research!