S-38.3150

Networked Multimedia Protocols and Services

2005–2006, 2nd period

Jörg Ott jo@netlab.hut.fi SE 324

General

- Architectures and details concerning IP-based multimedia from an Internet perspective
- Lectures: Tuesday, 10 – 12, S1 and Thursday, 10 – 12, S1
- Exercise (assignments + lectures): Wednesday, 14 – 16, S1
- Prerequisites
  - S-38.(2)188
  - Interest in protocols and their technical realization
  - Substantial coding skills (no C/C++ novice)
- Suitable for master studies: 4 ECTS points
- 6 weeks: a bit of an “animal experiment” (for you and me)

Coding Assignments

- 3 Assignments
  - Building on top of one another
  - Create the structure of a communication application
  - Deal with socket i/o and related system calls
  - Interpret standards text and implement packet interpretation/generation
  - Support parameterization and some visualization (no GUIs!)
- C/C++ code
  - Last(!) resort alternative: Java
  - Do the work on the Unix machines in the department (must at least work there)
- Small groups: 2 – 3
  - Send one email per group in the following format (one line per group member)
    "Last name: First name:IDs:email address"
- Completion: usually 2 weeks, last one until 31 December 2005
  - Send email with tgz or zip archive of source, build environment
  - Present all results interactively in 10-20 minutes per group (early January)

Exam

- 13 December 2005, 13 – 16, S5
- 3 hours time
- Some 10 – 12 questions
- Mostly knowledge + understanding
- Possibly one small problem to solve
- Hints in the last lecture
- Grade based upon the exam only
  - Delivering working assignment results prerequisite
  - First time exam (another "animal experiment", again for you and me)

Material

- Most Slides will be online as PDF
  - SIP lectures require paper copies to be made
  - Should be organized (volunteer?)
- Primary literature: RFCs and Internet Drafts
  - You can't read all of them (at least not before the end of next term)
  - Will point to a few selected ones recommended for studying
  - Some are required for assignments (usually only parts!)
- Books (difficult to find!)
  - Colin Perkins: RTP: Audio and Video for the Internet
  - Gonzalo Camarillo and Miguel Garcia: good books on SIP & 3G
- Beware of many bad or outdated ones!

Relation to other Netlab Courses

- 38.(3)188: Computer Networking: prerequisite
  - Some minor overlap
- 38.(3)115: Signaling Protocols: complementary
  - Can be done before or afterwards
- Protocol Design (4^{th} period): complementary
  - Will pick up and generalize some of the protocol concepts shown here
- Special Assignment in Networking Technology
  - May be developed based upon the subject discussed here
- Theses
  - IP-based multimedia one important research subject

Contents 1: Multimedia in General

1. Traditional (well: partly almost historic) Multimedia Applications
   Packet Real-time Basics

2. Real-time Transport Protocol (RTP)
   RTP Payload Formats and Error Correction

3. Session Announcements (SAP) and Descriptions of Multimedia Sessions and Media Streams (SDP, SDPng)

4. Multimedia Streaming Applications
   Internet Media Guides
   Real Time Streaming Protocol (RTSP)

Contents 2: Session Initiation Protocol

5. Introduction: History, Architecture, Terminology
6. Basic Signaling: Session Setup, Teardown
   Registration and User Location
7. Security for SIP-based Multimedia: Media Streams and Signaling
8. Issues with NATs and Firewalls
   NAT Traversal for SIP and Media Streams (STUN, TURN, ICE)
9. SIP Service Creation
10. SIP for Presence and Instant Messaging
11. SIP für Telephony and Multimedia Conferencing
12. Real World SIP
    Configuration, Legal Requirements, SIP Equipment

Further Informationen

- Course web page
  - http://www.netlab.hut.fi/~jo/teaching/nmps/index.htlm

- Some IETF Resources
  - http://www.ietf.org/charters.html/mmusic-charter.html
  - http://www.ietf.org/charters.html/avt-charter.html
  - http://www.ietf.org/charters.html/sip-charter.html
  - http://www.ietf.org/charters.html/sipping-charter.html
  - http://www.ietf.org/charters.html/simple-charter.html
  - http://www.softarmor.com/sipwg/
  - http://www.softarmor.com/sipping/
  - http://www.softarmor.com/simple/
  - http://www.dmn.tzi.org/ietf/mmusic/
  - http://www.rtsp.org/

IP Multimedia Architecture

Packet Real-time (A/V) Basics

IP Multimedia Applications (1)

- Packet multimedia experiments since 1980s
  - A/V tools + protocols for A/V over IP
  - Conference control protocols

Internet broadcasting (Mbone)

- First IETF Audiocast (1992)
- Broadcasts of IETF WG sessions
  - audio + video + whiteboard (transparencies)
  - enables remote participation (even talks)
- Broadcasting special events
  - talks, concerts, NASA shuttle missions, ...
- Broadcasting “radio” and “television” programs
  - Various channels available today (there was more some time ago)

IP Multimedia Applications (2)

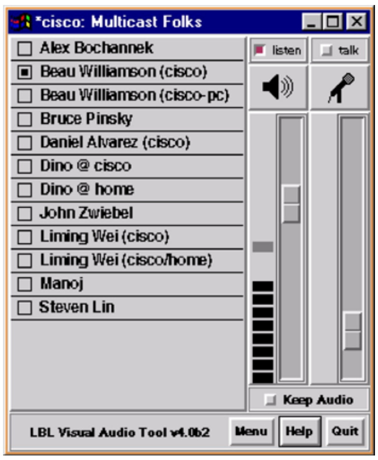
Teleconferences

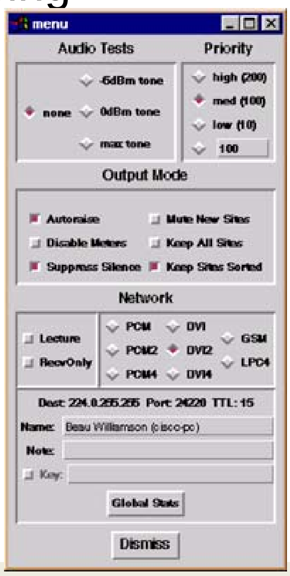
- Traditional Internet focus: large groups
- Small groups supported as well
- Audio + video + data (whiteboards, editors, ...)
- (Multimedia gaming sessions)
- Examples:
  - seminars and lectures
  - project meetings
  - work group meetings between IETFs
- Gatewaying where needed (PSTN, ISDN, cellular, …)

vic—Video Conferencing

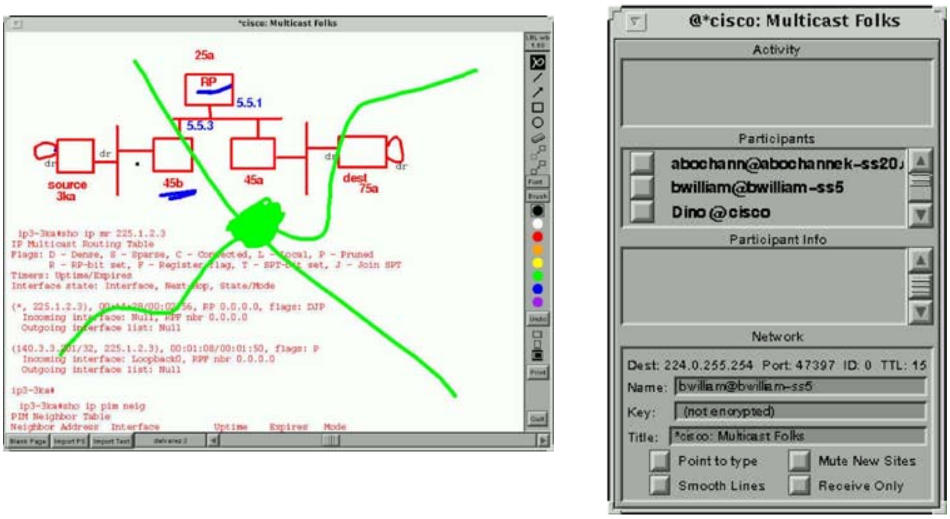
vat—Audio Conferencing

Vat is the original, now somewhat dated LBL tool.
For audio redundancy coding, use UCL's *rat* (robust audio tool).





wb—White Board



IP Multimedia Applications (3)

<u>IP Telephony</u>

- “Special case” of teleconferences
  - point-to-point + conference calls
- Gatewaying to PSTN / ISDN / GSM
  - also H.323, MGCP, MEGACO
- Include “Supplementary Services”
  - what users are used to from their touch tone phone or PBX environment and partly ISDN / GSM
- Include “Intelligent Network (IN)” services
  - To some degree trivial in an IP environment

IP Multimedia Applications (4)

Multimedia retrieval services

- "Video on demand"-style
  - including "VCR controls": pause/restart/cue/review
- Access to multimedia clips from web browsers
  - Commercial examples: RealAudio/RealVideo, IP/TV, Microsoft
- Often: Internet- / web-based access to live streams
  - "Big Brother", concerts, etc.
- Option: recording multimedia information

Common Requirements

Network infrastructure

- Multicast routing
- Real-time-capable packet forwarding
- Resource reservation

Transport protocols

- Real-time information (audio / video)
- Non-real-time information (data)

Media encoding standards

Security

Specific requirements

Control protocols

- Setup / teardown of communication relationships
- Call (and conference) control
- Remote control of devices (e.g. media sources)

Naming and addressing infrastructure

User (and service) location

Billing and accounting (and policing)

(Legal requirements)

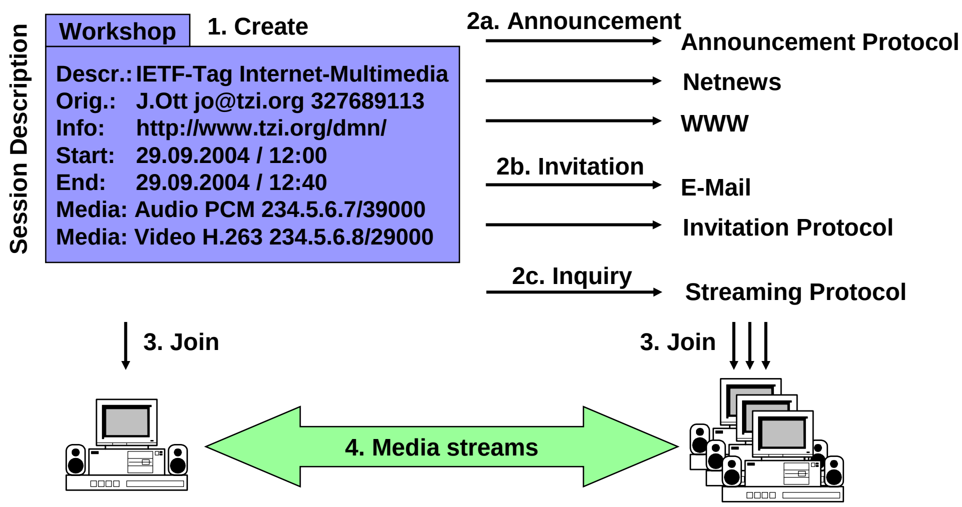
IETF Multimedia Conferencing

- Packet multimedia experiments since the 1980s
  - Audio/video tools + protocols for A/V over IP
  - Conference announcement and control protocols
- First IETF Audiocast (1992)
  - Mbone-based audio transmission from selected IETF working groups
- Since then: IETF sessions on the Mbone
  - Audio + video (+ sometimes slides)
  - Enabling remote participation (even talks)
- Other uses of Mbone conferencing
  - Broadcasting NASA missions, concerts, …
  - Lectures, seminars, project meetings, …

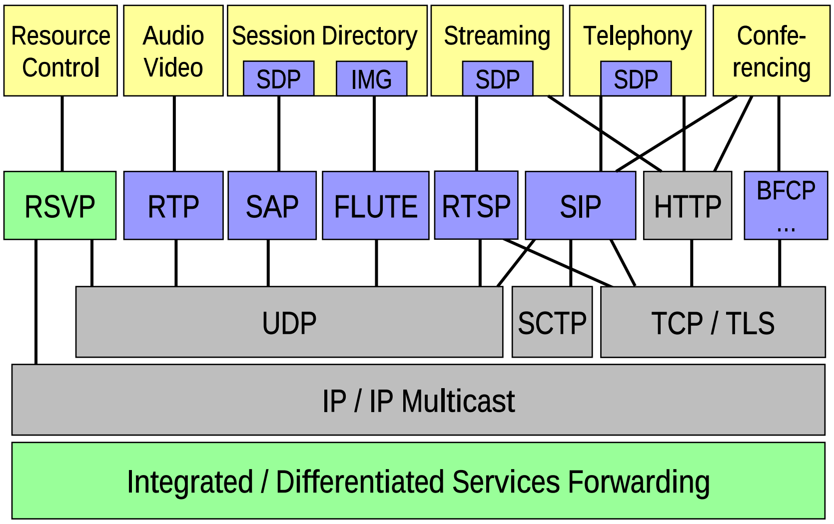
Traditional IETF Conferencing Concept

- Multicast-based
- Loosely-coupled conferences
  - no membership control
  - inexact information about participants
    - provided on a voluntary basis
  - security by encryption
- Public announcements and invitations
  - Convey session parameters, then get out of the way
    - Session Announcement Protocol (SAP), Internet Media Guides (IMG)
  - Session Initiation Protocol (SIP), Real-Time Streaming Protocol (RTSP)
- Conference control
  - Some need perceived; limited success over many years

Conference Establishment & Control



IETF Multimedia (Conferencing) Architecture

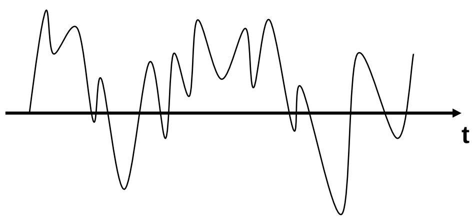


Real-time Media over Packets

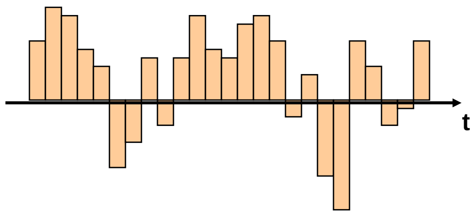
- Audio / Video are continuous media
- Packet networks transport discrete units
  - digitize media
  - compression
  - packetization
- No additional multiplex (beyond UDP/IP) needed:
  - no separate lines, bit allocations, etc.
  - transport different media in different packets
  - can give different quality of service to different media streams
  - allows different sites to receive different subsets

Real-time Media over Packets (2)

1) analog input signal



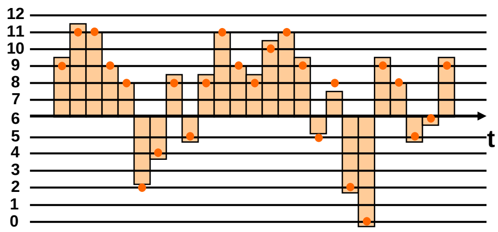
2) sampled input signal (implicit compression)



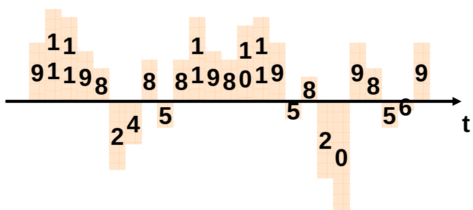
HELSINKI UNIVERSITY OF TECHNOLOGY
NETWORKING LABORATORY

Real-time Media over Packets (3)

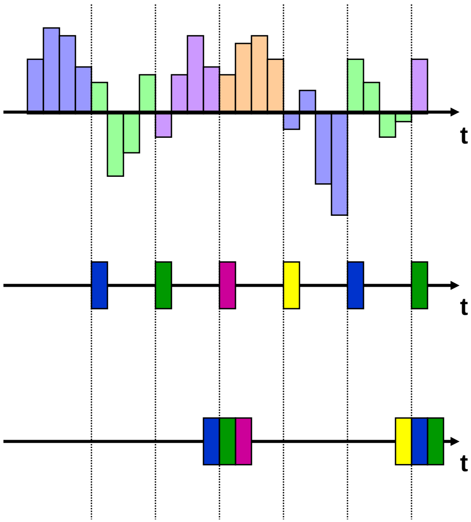
3) Quantization (another step of implicit compression)



4) Digital data stream



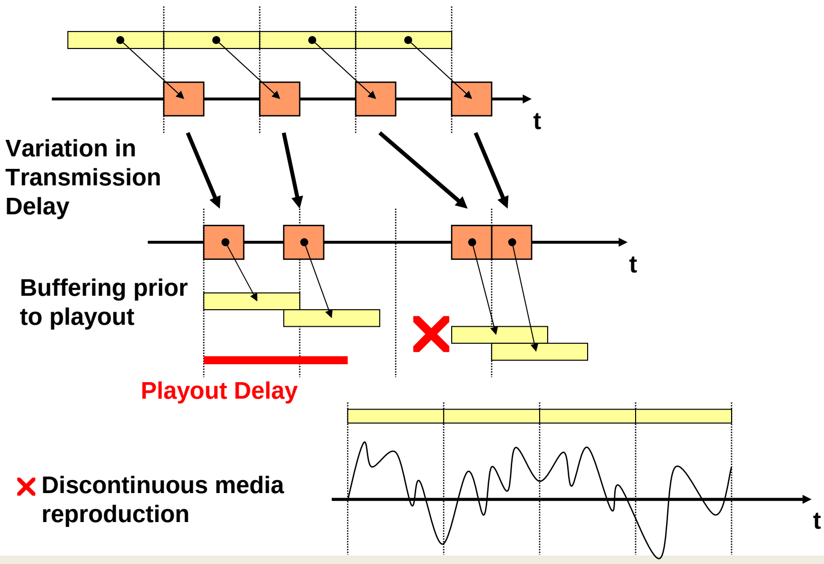
Real-time Media over Packets (4)



5) optional further compression yields small discrete frames

6) multiple frames or samples are collected to form packets

Real-time Media over Packets (5)



Real-time Media over Packets (6)

Little help needed from transport protocol:

- Retransmission may take too long (interactivity!)

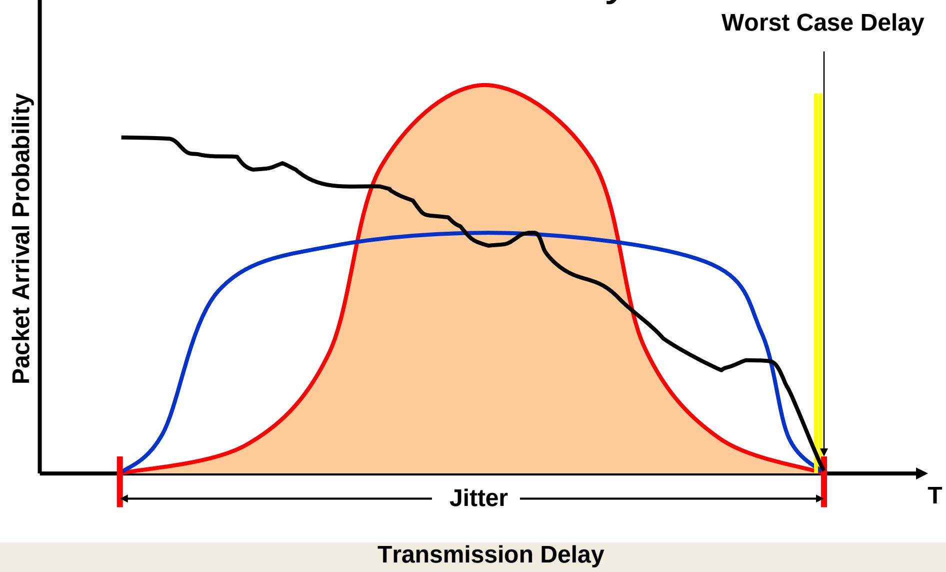
End systems must buffer before playout!

- Jitter in transmission delay due to queueing
- Packet A/V rule #1:
  - jitter is never a problem,
  - worst-case delay is!
- Need a timestamp in packet to be able to play at right time
  - intra-stream timing
  - optionally correlate for inter-stream timing (e.g. lip-sync)

Sources of Delay

- Sender
  - Capturing / digitizing delay (+ operating system)
  - Encoding / compression delay
  - Packetization delay
- Network (potentially highly variable!)
  - Link propagation delay (order of speed of light)
  - Serialization delay
  - Queuing delay
- Receiver
  - buffering delay + potential delay for repair
  - decoding / decompression delay
  - rendering / replay delay (+ operating system)

Jitter vs. Delay



Artificial Upper Bound on Delay