S-38.3152: “NMPS”

Networked Multimedia Protocols and Services

2007–2008, 1st and 2nd period

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General

- Architectures and details concerning IP-based multimedia from an Internet perspective
- Lectures: Tuesday, 8 – 10, S1 and Thursday, 10 – 12, S4
- 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++ or Java novice)
  - Time for lectures (the slides alone won’t do)

- Suitable for master studies: 5 ECTS points
This Specific Course

- **Experimental:** option to use two terms for the course

**Period 1:** 18 and 20 September
- Introduction
- Coding background
- First assignment

**Period 2:** 30 October – 13 December
- Repetition of the above
- All the rest

- **Idea:** allow for more time for the coding assignments

Coding Assignments

- 2-3 Assignments (schedule on the web to be updated)
  - 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++ or Java code supported by us
  - You can also use other languages: **on your own and at your own risk**
  - Do the work on the Unix machines in the department (must at least work there)
  - Details to follow
- **Small groups:** 2 – 3
  - Send one email per group in exactly the following format (one line per group member)
    "Last name:First name:Student ID:email address"
- **Completion:** 3 and 4 weeks, last one until 31 December 2007
  - Send email with tgz or zip archive of source, build environment
  - Present all results interactively in 10-20 minutes per group (early January)
A Note on Group Work

- Assignments organized around small groups
  - Work together: discuss, design, code, ask, understand
  - Split the load (but understand all parts)
  - Share the same assignment results

- You and your group members depend upon each other
  - So, please carry through
  - If you cannot make, let your other group members know
  - If you lose all your other group members, talk to us right away

Exam

- Thursday, 20 December 2007, 13 – 16, S5
- 3 hours time
- Some 10 – 12 questions
- Mostly knowledge + understanding
- Possibly one small problem to solve

- Hints in the last lecture (13.12.)

- Grade based upon the assignments (~30%) and the exam (~70%)
  - But: delivering working assignment results is a must
  - Need to obtain each ≥ 50% of the exam and assignment points
Material

- Lecture slides will be online as PDF
  - SIP lecture slides will only be accessible from TKK workstations

- Primary literature: RFCs and Internet Drafts
  - You can’t read all of them (at least not before the end of next term)
  - But you SHOULD read the core ones (we will point them out)
  - 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
  - Henry Sinnreich, Alan Johnston: good overview; not so much detail

- Beware of many bad or outdated ones!

Relation to other Netlab Courses

- 38.(2)188: Computer Networking: prerequisite
  - Some minor overlap

- 38.(3)115: Signaling Protocols: quite some overlap
  - Can be done before or afterwards
  - We focus on IETF-style IP-based multimedia

- Protocol Design (4th 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 of the major research themes
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), Descriptions of Multimedia
   Sessions, Media Streams (SDP, SDPng), Internet Media Guides

   Peer-to-Peer Streaming, Real Time Streaming Protocol (RTSP)
   IPTV, Speech Services Control (distributed speech synthesis)

Contents 2: Session Initiation Protocol

5. Introduction: History, Architecture, Terminology
   Basic Signaling: Session Setup, Teardown

6. Registration and User Location
   Advanced SIP signaling, media sessions

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: interfaces, application servers, endpoints

10. SIP for Presence and Instant Messaging, location information

11. SIP für Telephony, QoS, Multimedia Conferencing

12. Real World SIP: Policies, SPAM/SPIT, Configuration, Legal
    Requirements, SIP Equipment
Further Informationen

- Course web page
- Newsgroup
  - opinnot.sahko.s-38.tietoverkkotekniikka
- Some IETF Resources
  - http://www.softarmor.com/sipwg/
  - http://www.softarmor.com/sipping/
  - http://www.softarmor.com/simple/
  - http://www.softarmor.com/xcon/
  - 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
- Gatewaysing where needed (PSTN, ISDN, cellular, …)
vic—Video Conferencing

[yesterday]

vat—Audio Conferencing

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

[yesterday]
IP Multimedia Applications (3)

IP Telephony

- “Special case” of teleconferences
  - point-to-point + (centralized) conference calls

- Gateways to traditional telephony
  - PSTN / ISDN / GSM
  - Include “Intelligent Network (IN)” services
  - PBXes + supplementary services
  - also other IP telephony protocol suites: H.323, skype, …

- Expanding to cover other aspects of interpersonal interaction
  - Instant messaging + personal presence
  - Further application integration, …
Interactive Multimedia, Messaging, Presence:
SIP soft clients, skype, google talk, (mobile) phones

[and today]

IP Multimedia Applications (4)

Multimedia retrieval services

- "Video on demand"-style
  - including "VCR controls": pause/restart/cue review
  - Option: recording multimedia information

- 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.

- Broadcasting
  - IPTV, TVoDSL, ...
Multimedia streaming & IPTV

- Soft clients
- Mobile phones
- “Set-top Boxes”
- Mac Mini, Dreambox, …
- Television sets?

- Server-based streaming
  - YouTube and the like
- Peer-to-Peer Streaming
  - PPLive

Excursion: IP Multimedia Buzzwords

- Triple play
  - IP access + IP telephony + IP-based television
  - For DSL, cable, …
- Quadruple play (“we need to top this…”)
  - Adds mobility
  - Plain “marketingese”
- Internet Multimedia Subsystem (IMS)
  - Developed by 3GPP/2GPP2
  - IP-based subsystem for advanced multimedia services in UMTS networks
  - “Recent grand idea of the telcos”: use IMS in the fixed access networks, too.
  - Last attempt to take their customers hostage and prevent erosion of margins
  - There is little technical justification—it’s all about customer control and charging!
- Beware of service bundles
  - At least: freedom of choice and privacy are at risk!
Common Requirements

Network infrastructure
- Multicast routing
- Real-time-capable packet forwarding
- Resource reservation or proper provisioning

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
- (Messaging and presence)
- 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

Session Description

1. Create
   - Workshop
   - Descr.: Multimedia Protocols
   - Orig.: J.Ott jo@acm.org 327689113
   - Info: http://www.netlab.tkk.fi/~jo/
   - Start: 02.11.2006 / 10:00
   - End: 02.11.2006 / 12:00
   - Media: Audio PCM 234.5.6.7/39000
   - Media: Video H.264 234.5.6.8/29000

2a. Announcement
   - Announcement Protocol
   - Netnews
   - WWW

2b. Invitation
   - E-Mail
   - Invitation Protocol

2c. Inquiry
   - Streaming Protocol

3a. Join

4. Media streams

IETF Multimedia (Conferencing) Architecture

- Resource Control
- Audio Video
- Session Directory
- Streaming
- Telephony
- Conferencing
- RSVP
- RTP
- SAP
- FLUTE
- RTSP
- SIP
- HTTP
- BFCP
- UDP
- SCTP
- TCP / TLS
- IP / IP Multicast
- Integrated / Differentiated Services Forwarding
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)
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)

Variation in Transmission Delay

Buffering prior to playout

Playout Delay

× Discontinuous media reproduction

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)

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Jitter vs. Delay

[Diagram showing Jitter vs. Delay with curves and labels for Packet Arrival Probability and Transmission Delay]

Worst Case Delay

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Artificial Upper Bound on Delay

Transmission Delay

Worst Case Delay

Packet Arrival Probability

Reduced overall delay

“Artificial” Packet Loss

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Dealing with Delay and Jitter

- Dejittering buffer
  - Receive packets and store them
  - Determine playout point
  - Reorder (if necessary)
  - Determine packets lost
  - Related: Error/loss concealment mechanisms of the codec

- Determining playout point: non-trivial
  - Don’t want to be too early (artificial loss increases) nor too late (quality)
  - Make some initial guess
  - Permanently monitor jitter of incoming packets and buffer contents
    - Monitor late packets (“artificial loss”)
  - Voice: adapt (reduce) delay during speech pauses