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!)
  - Great overview: J. Rosenberg: "A Hitchhiker's Guide to SIP"
- 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 ($4^{th}$ 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 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

4. Multimedia Streaming Applications, Multimedia Broadcasting
   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
  - http://www.netlab.tkk.fi/opetus/s383152/2007/index.html
- Newsgroup
  - opinnot.sahko.s-38.tietoverkkotekniikka
- 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.ietf.org/charters.html/xcon-charter.html
  - 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
- Gatewaying where needed (PSTN, ISDN, cellular, …)

# vic—Video Conferencing



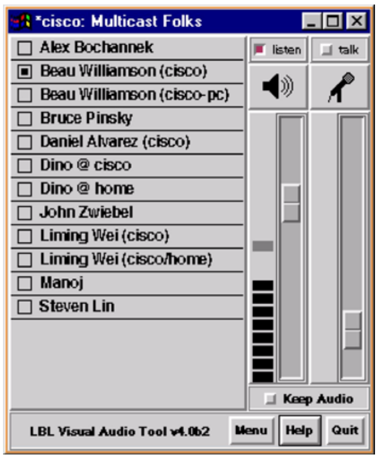
[yesterday]

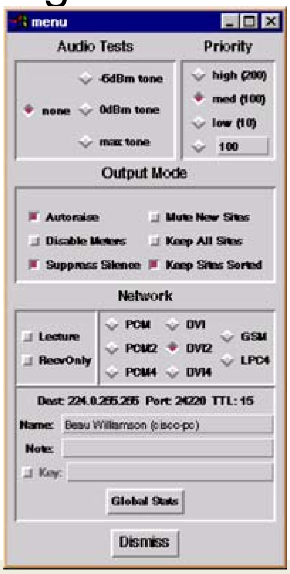





# vat—Audio Conferencing

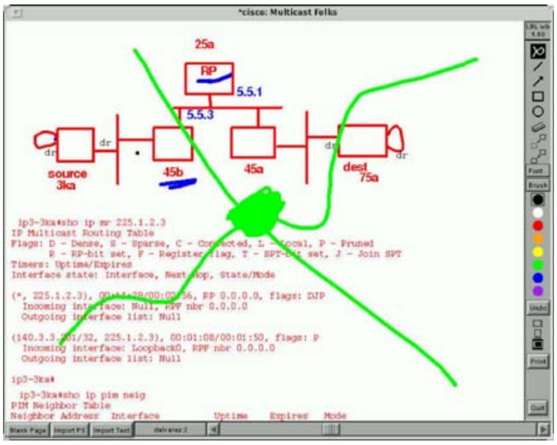
**Vat is the original, now somewhat dated LBL tool.**
**For audio redundancy coding, use UCL's *rat* (robust audio tool).**




[yesterday]

# wb—White Board





[yesterday]





# IP Multimedia Applications (3)

## IP Telephony

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

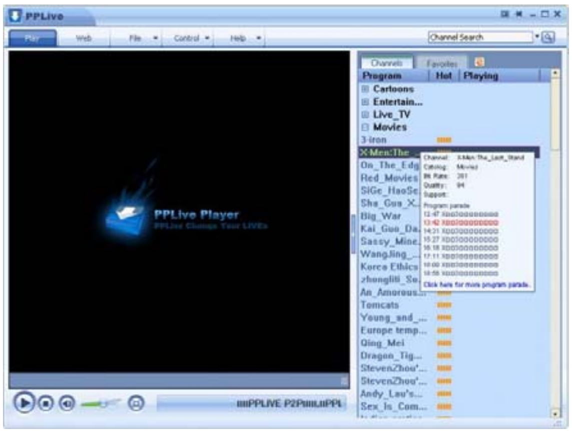
- "Video on demand"-style
  - including “VCR controls”: pause/restart/cue/review
  - Option: recording multimedia information
- Access to multimedia clips from web browsers
  - Commercial examples: RealAudio/RealVideo, IP/TV, Microsoft
- Often: Internet- / web-based access to live streams
  - “Big Brother”,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

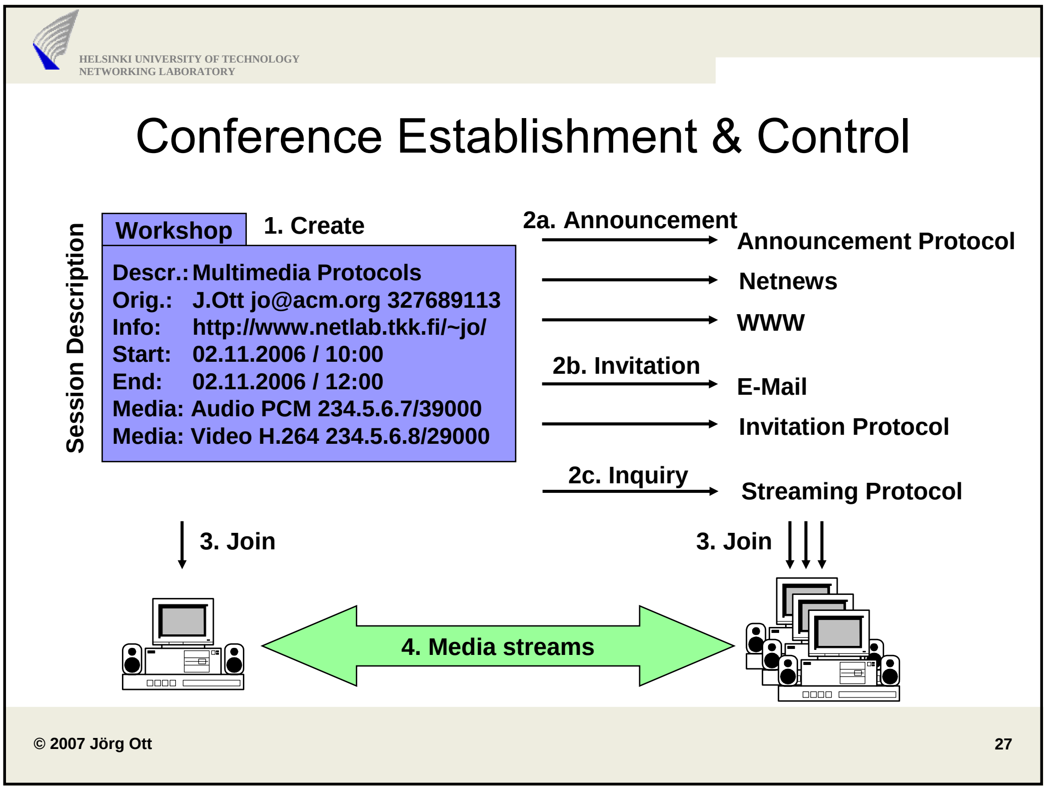
- Packet multimedia experiments since the 1980s
  - Audio/video tools + protocols for A/V over IP
  - Conference announcement and control protocols
- First IETF Audiocast (1992)
  - Mbone-based audio transmission from selected IETF working groups
- Since then: IETF sessions on the Mbone
  - Audio + video (+ 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


HELSINKI UNIVERSITY OF TECHNOLOGY
NETWORKING LABORATORY
Conference Establishment & Control
Session Description
Workshop
1. Create
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
3. Join
3. Join
4. Media streams
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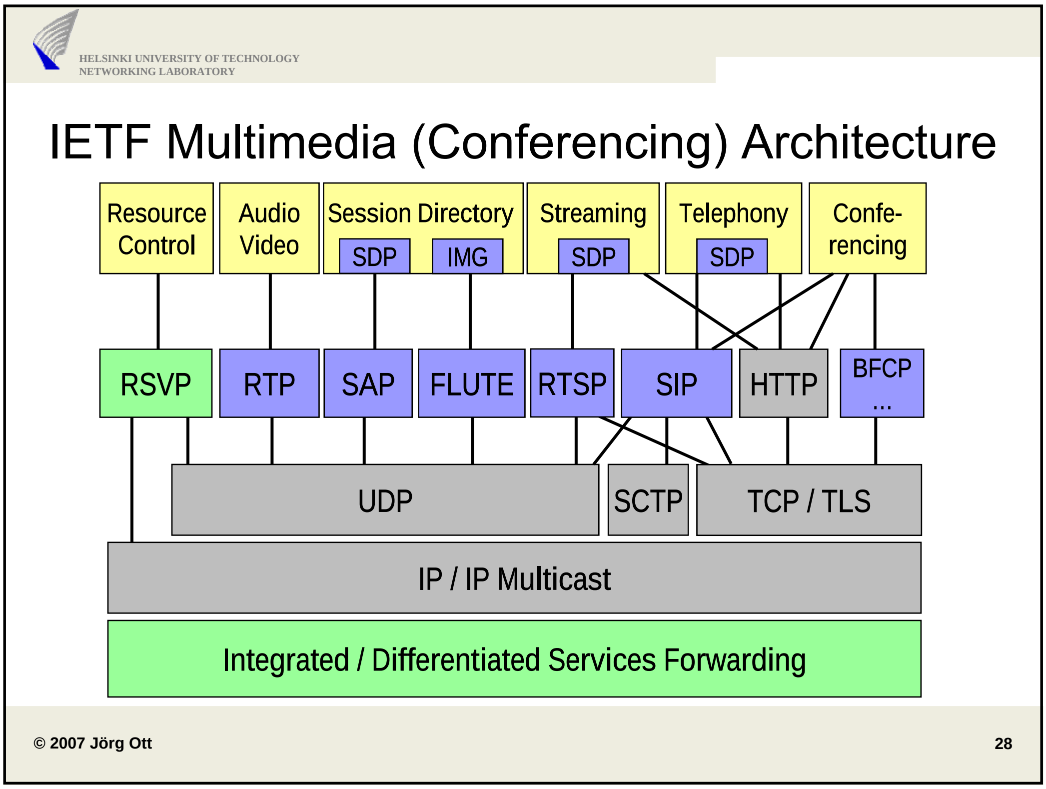
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IETF Multimedia (Conferencing) Architecture
Resource Control
Audio Video
Session Directory
SDP
IMG
Streaming
SDP
Telephony
SDP
Confe-rencing
RSVP
RTP
SAP
FLUTE
RTSP
SIP
HTTP
BFCP ...
UDP
SCTP
TCP / TLS
IP / IP Multicast
Integrated / Differentiated Services Forwarding
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# Real-time Media over Packets

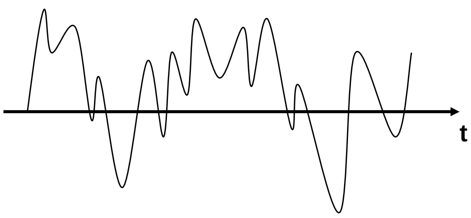
- Audio / Video are continuous media
- Packet networks transport discrete units
  - digitize media
  - compression
  - packetization
- No additional multiplex (beyond UDP/IP) needed:
  - no separate lines, bit allocations, etc.
  - transport different media in different packets
  - can give different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



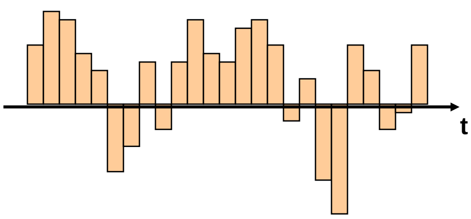



# Real-time Media over Packets (2)

**1) analog input signal**



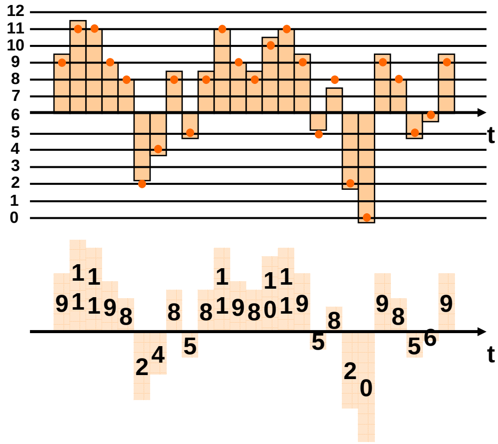

**2) sampled input signal (implicit compression)**



# Real-time Media over Packets (3)

3) Quantization
(another step of
implicit
compression)

4) Digital data stream



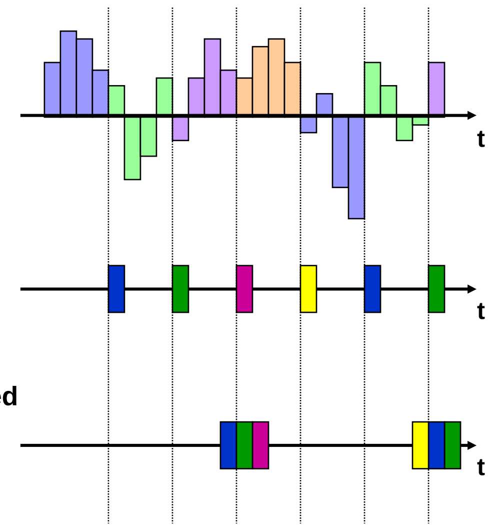





# Real-time Media over Packets (4)

5) optional further
compression yields
small discrete frames

6) multiple frames
or samples are collected
to form packets



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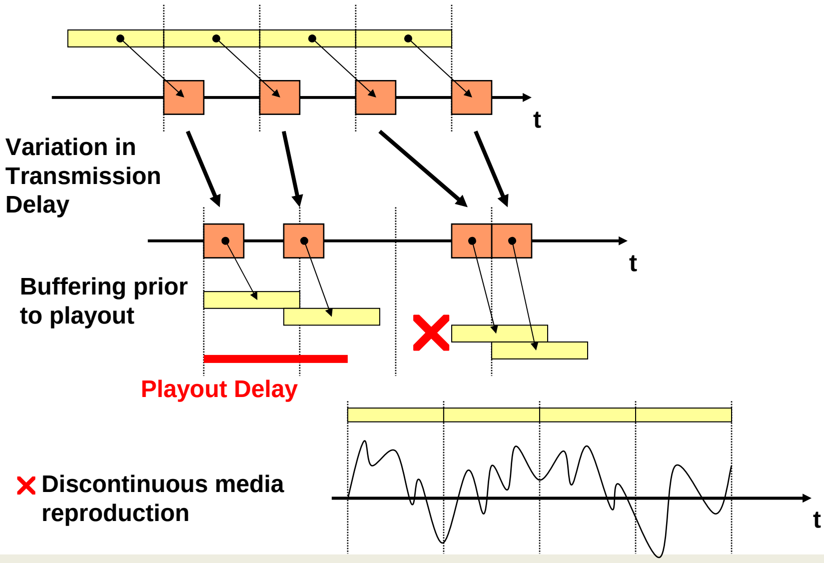
# Real-time Media over Packets (5)



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# 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)

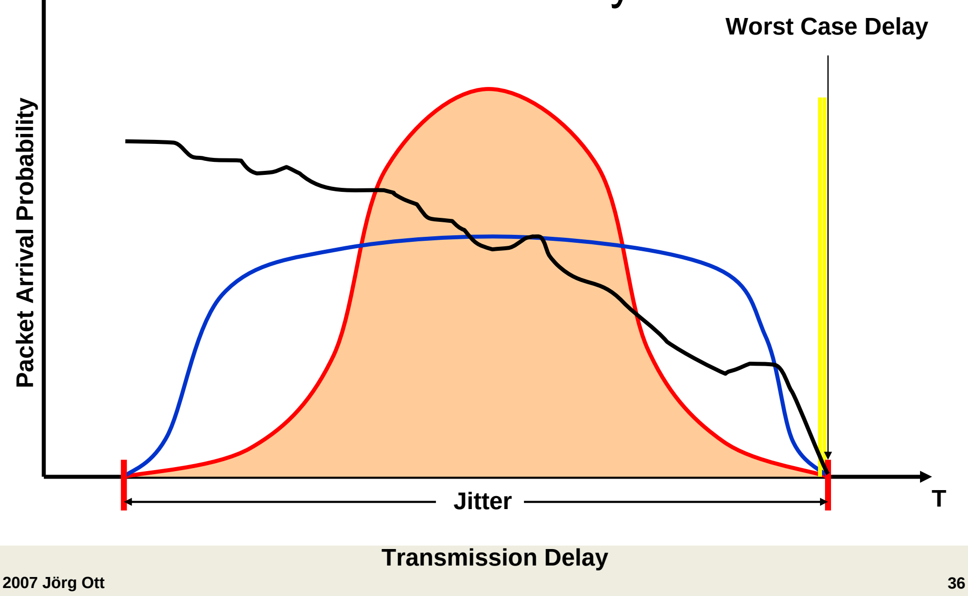
# Sources of Delay

- 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)

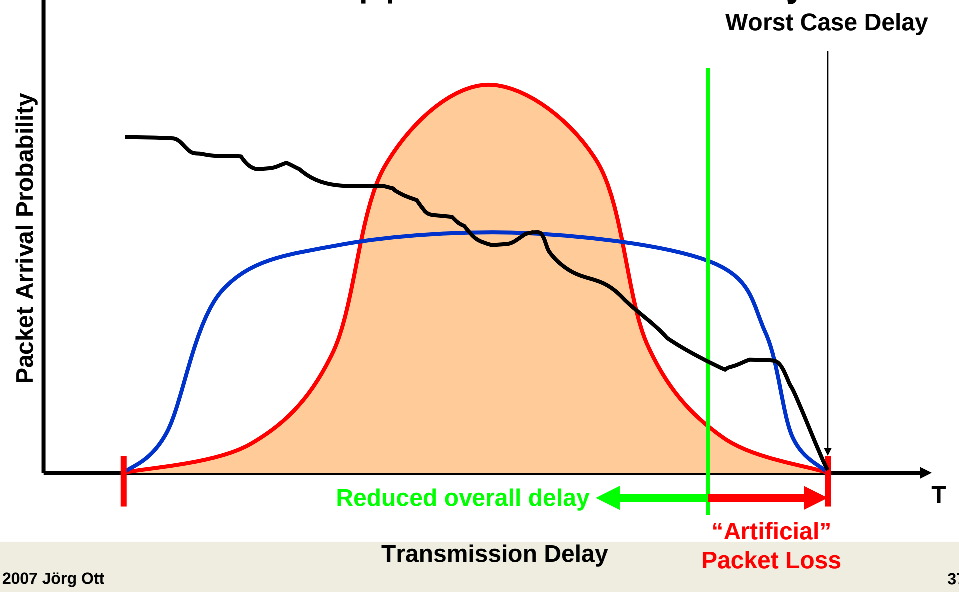




# Jitter vs. Delay



# Artificial Upper Bound on Delay








# 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