Video-mediated communication for e-health applications

Mathias Johanson
Alkit Communications AB
mathias@alkit.se
Outline

• Introduction: basic concepts and challenges
• Video compression and encoding
• Basic concepts of computer networks and data communication
• Network protocols and standards for video-mediated communication
• Multipoint video communication
• Firewall traversal
• E-health applications
The vision...

What is video-mediated communication?

"Synchronous communication between two or more persons using live video"

...using digital video signals
...in combination with other media (e.g. audio)
...using computer networks (the Internet)
Video-mediated communication over the Internet

...point to point

...or multipoint
Video communication applications

• Videoconferencing
  – "Distributed meeting room", term often used for all kinds of interpersonal video communication
• Video-on-demand
  – One-way communication, much less delay-sensitive, pre-recorded video
• Video broadcast
  – One-to-many communication (possibly with audio backchannels), less delay-sensitive, less interaction
• Videophone
  – Point-to-point interpersonal video calls
• Video chat
  – Multipoint or p2p low quality video interaction
• Telemedicine / e-Health
  – Distance consultation, remote examinations, remotely guided surgery, remote echocardiography, etc.
Motivation

• Video improves communication quality
  – Body language, gestures, facial expressions

• Enabling new applications
  – Telemedicine, teleteaching, distributed collaborative work, telerobotics, telepresence

• Reduce traveling – increase opportunities to interact – communicate more efficiently

• Use the Internet as the network for all our communication needs (i.e. a multiservice network)
Challenges

- Video is bandwidth-demanding
- Video processing is computationally expensive
- Video-mediated communication is delay-sensitive
- The Internet is best-effort; video is sensitive to packet loss
- Multipoint communication is troublesome
- NAT firewall traversal can be problematic
- Many usability pitfalls
Components of a video communication system

- Video acquisition / sampling, digitization
- Video compression and encoding
- Packetization, multiplexing, transmission
- Reception, demultiplexing, reassembly
- Decoding
- Presentation
Video compression and encoding
The bandwidth problem
(”Why do we need video compression?”)

• Uncompressed video is prohibitively broadband:
  – Example: An uncompressed PAL signal, sampled at 720x576 pixels, 25 frames-per-second, 24 bits-per-pixel requires 720x576x25x24 bps = 249 Mbps

• A typical local area network (LAN) has a bandwidth of 100 Mbps

• WAN (Internet) bandwidth is still (relatively) expensive
The bandwidth problem

<table>
<thead>
<tr>
<th>Format</th>
<th>Bitrate Mbit/s</th>
<th>Storage need GB/min film</th>
</tr>
</thead>
<tbody>
<tr>
<td>CCIR 601 768x576 RGB</td>
<td>248</td>
<td>1,9</td>
</tr>
<tr>
<td>HDTV 720i 1280x720 RGB</td>
<td>550</td>
<td>4,1</td>
</tr>
<tr>
<td>HDTV 1080i 1920x1080 RGB</td>
<td>1244</td>
<td>9,3</td>
</tr>
<tr>
<td>HDTV 1080p 1920x1080 RGB</td>
<td>2488</td>
<td>18,6</td>
</tr>
</tbody>
</table>
Video compression

• Exploits temporal and spatial redundancy to reduce bitrate (video signals are smooth and vary slowly with time)

• The bitrate can typically be reduced by more than 99% without noticeable loss of quality
Lossy and lossless compression

- **Lossy compression**
  - Original video is not recreated exactly, but with high perceptual similarity
  - Exploits properties of the human visual system to discard irrelevant data

- **Lossless**
  - Original signal is recreated exactly, after compression/decompression
  - Moderate compression performance (at best)
Video compression techniques

• Colorspace conversion
  – RGB -> YCrCb
• Component subsampling
  – 4:2:2, 4:1:1, 4:1:0
• Motion-compensated inter-frame coding
• Transform coding (intra coding)
  – Block based DCT, Wavelet, …
• Quantization
• Run-length encoding
• Entropy coding
  – Huffman coding, arithmetic coding, …
Typical video coding pipeline
Motion compensation

Macroblocks are coded differentially from a spatially translated macroblock from a previous (or subsequent) frame

The motion vector is also coded and transmitted
Inter-frame coding

- I-frame
- P-frame
- B-frame
Transform coding

\[ F(u, v) = \frac{C_u C_v}{2} \sum_{y=0}^{7} \sum_{x=0}^{7} f(x, y) \cos \left( \frac{(2x + 1)u\pi}{16} \right) \cos \left( \frac{(2y + 1)v\pi}{16} \right) \]

with:

\[ C_u = \begin{cases} \frac{1}{\sqrt{2}} & \text{if } u = 0, \\ 1 & \text{if } u > 0 \end{cases} ; \quad C_v = \begin{cases} \frac{1}{\sqrt{2}} & \text{if } v = 0, \\ 1 & \text{if } v > 0 \end{cases} \]

- Applied to 8x8 blocks
- Zig-zag scan order
DCT

Original 8x8 block

Reconstructed 8x8 block

DCT

scaling and inverse DCT

Q

run-level-coding

transmission

run-level-decoding

inverse zig-zag-scan

Mean of block: 185
{(0,3) (0,1) (1,1) (0,1) (0,1) (0,-1) (1,1) (0,1) (0,1) (0,1) (0,1) (0,1) (0,1) (0,1) (0,1) (0,1) (0,1)}

BOB

Mean of block: 285
{(0,3) (0,1) (1,1) (0,1) (0,1) (0,1) (1,1) (0,1) (0,1) (0,1) (0,1) (0,1) (0,1) (0,1) (0,1) (0,1) (0,1)}

BOB

Mean of block: 185
{(0,3) (0,1) (1,1) (0,1) (0,1) (0,1) (1,1) (0,1) (0,1) (0,1) (0,1) (0,1) (0,1) (0,1) (0,1) (0,1) (0,1)}

BOB

Mean of block: 285
{(0,3) (0,1) (1,1) (0,1) (0,1) (0,1) (1,1) (0,1) (0,1) (0,1) (0,1) (0,1) (0,1) (0,1) (0,1) (0,1) (0,1)}

BOB

Mean of block: 185
{(0,3) (0,1) (1,1) (0,1) (0,1) (0,1) (1,1) (0,1) (0,1) (0,1) (0,1) (0,1) (0,1) (0,1) (0,1) (0,1) (0,1)}

BOB

Mean of block: 285
{(0,3) (0,1) (1,1) (0,1) (0,1) (0,1) (1,1) (0,1) (0,1) (0,1) (0,1) (0,1) (0,1) (0,1) (0,1) (0,1) (0,1)}

BOB
## Video compression algorithms

<table>
<thead>
<tr>
<th>Algorithm</th>
<th>Target bandwidth</th>
<th>Typical application</th>
</tr>
</thead>
<tbody>
<tr>
<td>MPEG 2</td>
<td>2 - 50 Mbps</td>
<td>DVD, Digital TV</td>
</tr>
<tr>
<td>H.261, H.263</td>
<td>64 kbps - 2 Mbps</td>
<td>narrowband videoconferencing</td>
</tr>
<tr>
<td>MPEG 4</td>
<td>500 kbps - 50 Mbps</td>
<td>Multimedia authoring, conferencing</td>
</tr>
<tr>
<td>H.264 / MPEG 4 part 10</td>
<td>20 kbps -</td>
<td>Videoconferencing, multimedia, 3G, …</td>
</tr>
<tr>
<td>DV</td>
<td>25 Mbps</td>
<td>Digital video cameras</td>
</tr>
</tbody>
</table>
Video compression performance comparison

Fig. 1. Rate-PSNR curves of CAROUSEL sequence encoded using H.264 and MPEG-2 encoders.

Fig. 4. Rate-PSNR curves of QCIF FOREMAN sequence encoded using H.264, H.263 Baseline and H.263 CHC encoders.

\[ PSNR = 20 \log_{10} \frac{\max_{i=1}^{N} x_i}{\frac{1}{N} \sum_{i=1}^{N} (x_i - y_i)^2} \]
Aspects of video quality

• Resolution
  – How many pixels is the video signal represented by vertically and horizontally?

• Frame rate
  – How many frames per second?

• Precision / bit depth
  – How many bits are each pixel represented by?

• Compression distortion
  – How much lossy compression is applied
  – Can be quantitatively measured using PSNR
More aspects of video quality

• Good lighting conditions (right color temperature)
• Neutral background
• High quality cameras
• Camera positioning: eye contact
• Natural size
Video compression problems

• Too heavy compression introduces distortion (compression artifacts)
• Very high computational complexity (but remember Moore's law…)
• Coding delay
• Inter-frame dependencies makes video streams sensitive to packet loss
Compression artifacts

Blockiness: quantisation distortion in block-based based compression algorithms (JPEG, MPEG, H.263, etc.)

Mosquito noise, Gibbs effect: Quantisation distortion in high frequency parts of an image, due to transform coding (DCT, Wavelet)
Compression artifacts

Temporal prediction error:

Propagation error in inter-frame compression algorithms (MPEG, H.26x, …)
IS&T’s 50th Annual Conference
Basic concepts of computer networks and data communication
Connectionless vs. Connection-oriented networks

• Packet switching
  – The Internet, frame relay, GPRS, …
• Circuit switching
  – POTS, ISDN, ATM, X.25, …
IP-based packet networks

• Data is fragmented into small units called *packets*
• Packets are given a *header* containing the destination address the source address (and some other stuff)
• Packets are relayed hop by hop by *routers*. Routing is performed based on destination address only
• Packet delivery is *best effort*
Video communication in packet networks

Video camera

Video signal

digitizing, compression, packetization, transmission

IP packets

defaultrouter

Internet

depacketization, decompression, rendering

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Router

• En router är en nätverksutrustning som ser till att IP-paket skickas vidare till rätt destination, utifrån den IP-adress som finns i paketets huvud

• En router har en stor tabell där alla destinationsnätverk den känner till lagras tillsammans med nästa hopp till destinationen

• Vägval görs hopp-för-hopp (hop-by-hop) baserat på destinationsadressen
Routing

• *Dynamisk routing*
  – En routingalgoritm räknar ut hur routingtabellen skall se ut, utifrån information från andra routrar om hur deras routingtabeller ser ut. Om en länk går ner routas paketen om.

• *Statisk routing*
  – Ett statiskt manuellt konfigurerat vägval

• *Default routing*
  – Om ingen route matchar destinationen för ett paket skickas det till en *default router*
Autonomous system

"An autonomous system (AS) is a set of routers having a single routing policy, running under a single technical administration."
Internet architecture

AS 1

AS 2

AS 3

Interior gateway protocol, e.g. OSPF

Exterior gateway protocol, e.g. BGP-4

Interior gateway protocol, e.g. OSPF

Interior gateway protocol, e.g. OSPF
Routing protocols

• Interior Gateway Protocols
  – Distance Vector Protocols
    • RIP
  – Link State Protocols
    • OSPF, IS-IS

• Exterior Gateway Protocols
  – EGP, BGP-4
RIP: Routing Information Protocol

- *Distance vector*-algoritm (Bellman-Ford)
- Varje router skickar hela sin routingtabell med jämna mellanrum till alla sina grannar.
- En router räknar ut sitt avstånd till varje destination baserat på grannarnas routingtabeller
- Långsam konvergens
- Fungerar bara för mycket små nät
OSPF: Open Shortest Path First

- Link-State-algorithm
- Varje router håller reda på sina grannar och genererar ”link state”-paket, som anger vilka grannar routern har förbindelse med, och en ”kostnad” för varje länk.
- LS-paketen skickas via ”flooding” till alla andra routrar i nätverket.
- Varje router räknar ut kortaste vägen till varje destination (dvs routingtabellen) med Dijkstra’s algorithm
Border Gateway Protocol (BGP)

- Exterior Gateway Protocol
- BGP routers at the same exchange point (known as *neighbors* or *peers*) exchange route update messages of all known routes
- Route update messages contain an IP address prefix and a sequence of AS numbers identifying the Autonomous Systems the route has traversed so far
- Each router builds a routing table based on the route update messages and a policy database
Network protocols and standards for video-mediated communication
**OSI reference model**

<table>
<thead>
<tr>
<th>Layer</th>
<th>Protocols/Protocols etc.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Application layer</td>
<td>FTP, HTTP, SMTP, other application-level protocols</td>
</tr>
<tr>
<td>Transport layer</td>
<td>TCP, UDP, RTP…</td>
</tr>
<tr>
<td>Network layer</td>
<td>IP (IPX, DECnet, other obsolete protocols…)</td>
</tr>
<tr>
<td>Datalink layer</td>
<td>Ethernet, SDH, Frame relay, ATM, ISDN,</td>
</tr>
<tr>
<td>Physical layer</td>
<td>Electrical/optical specifications, cabling, etc.</td>
</tr>
</tbody>
</table>
Physical layer

• Electrical and optical signaling, cabling, etc.
Data link layer

• På datalänksnivån specificeras det som rör kommunikationen mellan två fysiskt sammankopplade enheter, t.ex. mellan en dators nätverksinterface och en nätverksväxels interface

• Korrigering av bitfel

• Om mediet för transmissionen är delat (t.ex. radiolänk) hanteras åtkomsten till mediet, kollisionshantering, etc.
Ethernet

- IEEE 802.3
- 10 Mbps, 100 Mbps, 1 Gbps, 10 Gbps
- CSMA/CD
- 48-bit unique MAC-addresses, e.g. 00:30:05:24:61:F1
- 22 bytes header (preamble 8 bytes, destination address 6 bytes, source address 6 bytes, type 2 bytes)
- 4 byte trailer (CRC)
- MTU 1500 bytes
Network layer

• På nätverksnivån hanteras den funktionalitet som har att göra med kommunikationen mellan två ändpunkter i ett nätverk
• Överbryggning mellan olika datalänkstekniker
• Adressering
• Routing
IP - Internet Protocol

• Nätverksprotokollet som används på Internet
• Adressering via unika 32-bitars adresser (t.ex. 192.36.136.15)
• Routing, dvs vägval
• “Best effort”, dvs inga garantier för att paket kommer fram
• Förbindelselöst
## IP-header

<table>
<thead>
<tr>
<th></th>
<th>0</th>
<th>8</th>
<th>16</th>
<th>24</th>
<th>32</th>
</tr>
</thead>
<tbody>
<tr>
<td>Version</td>
<td>IHL</td>
<td>Type of Service</td>
<td>Total Length</td>
<td>Identification</td>
<td>Flags</td>
</tr>
<tr>
<td>Time to live</td>
<td>Protocol</td>
<td>Header checksum</td>
<td>Source Address</td>
<td>Destination Address</td>
<td>Options</td>
</tr>
</tbody>
</table>

Data (variable length)

IHL = Internet header Length
Transport layer

• Funktionalitet för att flera kommunikationssessioner över en förbindelse
• Mekanismer för tillförlitlighet (omsändningar)
• Flödeskontroll (congestion control)
TCP - Transmission Control Protocol

- Transportprotokoll som används för tillförlitlig överföring av data. (Protokollet ser till att alla paket kommer fram förr eller senare.)
- Kvitto (acknowledgement) skickas av mottagaren för varje mottaget paket
- Omsändning av borttappade paket
- Flödeskontrollalgoritmus (congestion control) anpassar överförings-hastigheten
- Flera kommunikationssessioner multiplexeras via portnummer
- Ej lämpligt för realtidskommunikation
UDP - User Datagram Protocol

- Unreliable datagram protocol
- Multiplexing through port numbers
- Checksum for detecting and discarding packets with bit errors
## UDP-header

<table>
<thead>
<tr>
<th>Source Port</th>
<th>Destination Port</th>
</tr>
</thead>
<tbody>
<tr>
<td>Length</td>
<td>Checksum</td>
</tr>
</tbody>
</table>

Data (variable length)
RTP - Real-time Transport Protocol

- Transportprotokoll för realtidsdata (ljud, video, etc.)
- Sekvensnummer
- Tidsstämplar
- Identifierare för vilken datatyp ett paket innehåller (payload type identifier)
- Mekanism för återkoppling till sändaren av kvalitetsparametrar (jitter, paketförluster, etc.)
- Rudimentär sessionshantering (vilka som deltar i en konferenssession, etc.)
## RTP-header

<table>
<thead>
<tr>
<th>V=2</th>
<th>P</th>
<th>X</th>
<th>CC</th>
<th>M</th>
<th>PT</th>
<th>sequence number</th>
</tr>
</thead>
<tbody>
<tr>
<td>timestamp</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>synchronization source (SSRC) identifier</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>contributing source (CSRC) identifiers</td>
<td>...</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
Applikationsnivån

• Allt som har att göra med med hur applikationen hanterar data som kommuniceras
Network and transport protocols used for IP-based video communication systems

- IP (Internet protocol)
- TCP (Transmission Control Protocol)
- UDP (User Datagram Protocol)
- RTP (Real-time Transport Protocol)
- SIP (Session Initiation Protocol)
# Packetization of real-time data

<table>
<thead>
<tr>
<th></th>
<th>IP-header</th>
<th>UDP-header</th>
<th>RTP-header</th>
<th>Payload</th>
</tr>
</thead>
<tbody>
<tr>
<td>Sender and receiver</td>
<td>24 bytes</td>
<td>8 bytes</td>
<td>12 bytes</td>
<td>Variable length (typically ~1 Kb)</td>
</tr>
<tr>
<td>addresses</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Port numbers for</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>multiplexing</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Sequence numbers,</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>timestamps, payload</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>type identification</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Video data</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
SIP - Session Initiation Protocol

- Protocol for initiation and management of synchronous communication sessions
SIP message example

INVITE sip:reflex@138.203.236.129 SIP/2.0
Via: SIP/2.0/UDP 138.203.248.150:5060
From: <sip:789@cc.be>;tag=10c50710
To: <sip:reflex@138.203.236.129>
Call-ID: 054f500636a84e40@vidl007p.etb.bel.alcatel.be
CSeq: 1 INVITE
Max-Forwards: 30
Contact: <sip:789@138.203.248.122:5170>
Content-Type: application/sdp
Content-Length: 153

v=0
o=mathias 25733 14972 IN IP4 138.203.248.122
s=test
c=IN IP4 138.203.248.122
m=audio 23012 RTP/AVP 0
a=rtpmap:0 PCMU/8000
Standards for video-mediated communication

- H.320
  - ITU-T standard for narrowband videoconferencing over ISDN (first generation systems, early 90's)
- H.323
  - Retrofit of H.320 to packet networks (second generation systems, late 90's)
- H.321
  - ITU-T standard for broadband videoconferencing over ATM (never really caught on)
- IETF RFC 1889, RFC1890, RFC 2475, RFC 2543, etc.
  - Internet standards for videoconferencing over IP (third generation systems, present day)
The standardization confusion

ISO
- MPEG-1
- MPEG-2
- MPEG-4
- JPEG

ITU-T
- H.320
- H.323
- G.711
- H.261
- H.263
- H.262
- H.264
- T.81

IETF
- SIP (RFC 2543)
- RTP (RFC 3550)
- RFC 3551
- RFC 2250
- RFC 2435

Video compression
Image compression
Videoconferencing (umbrella standard)
Session management
Paketization
Multipoint and group communication
Problem description

A wants to send a $P$ bps data stream to B, C and D.
Solution 1: reflector

A wants to send a $P$ bps data stream to B, C and D

Observe however that $3xP$ bps is consumed on the reflector's egress network connection
Reflector bandwidth gain

Group communication between 5 hosts

4 incoming and 4 outgoing streams on each host’s network connection

Full unicast mesh

4 incoming and 1 outgoing stream on each host’s network connection

Single reflector configuration
Reflector example
Exempel på reflektoranvändning

{sessionsnamn = DITRA, IP-adress 130.240.9.119}

{sessionsnamn = DITRA, IP-adress 130.240.21.218}

{sessionsnamn = DITRA, IP-adress 217.209.202.38}

{sessionsnamn = HEJSAN, IP-adress 192.36.136.15}

{sessionsnamn = HEJSAN, IP-adress 129.16.31.60}

Reflektor

RTP-data (video, ljud)

Sessioner:

HEJSAN 129.16.31.60, 192.36.136.15


BYE

{OK}

{OK}

{OK}

{OK}

RTP-data (video, ljud)

Sessioner:

HEJSAN 129.16.31.60, 192.36.136.15


130.240.9.119

130.240.21.218

217.209.202.38

129.16.31.60

192.36.136.15
Solution 2: multicast

A wants to send a $P$ bps data stream to B, C and D.

Network takes care of routing the packets in the optimal way.

$P$ bps consumed
Multipoint communication, summary

- **IP multicast**
  - Reserved range of IP-addresses (224.0.0.0 to 239.255.255.255) for group communication
  - Dynamic group memberships via dedicated signaling protocol (IGMP)
  - Dedicated multicast routing protocols (e.g. DVMRP, PIM, MOSPF)
  - Not implemented everywhere

- **Reflectors (MCUs)**
  - Application level gateways that relay packets among the members of a group
Adaptivity, scalability, heterogeneity and packet loss resilience
Bandwidth allocation problem

• How can the available network bandwidth be shared between the users of the network in a fair way?
• The Internet is connectionless: rate control is end-to-end
• The available bandwidth is time varying: rate control algorithms must be adaptive
Adaptive, rate controlled video communication

• Send quality feedback information from video receiver to video sender
• Adapt video compression parameters to match available bandwidth
• Choose suitable video codec based on available resources (bandwidth, CPU power) and quality demands
Adaptive video communication
Heterogenity problem in multipoint settings

- video sender
- video receiver 1
- video receiver 2
- video receiver 3

feedback

network
Transcoders, mixers

• A transcoder, or a transcoding gateway converts between different encodings in real time
• A mixer performs some synthesis operation, combining many media streams into one
Transcoding gateway

- video sender
- Transcoding gateway
- video receiver 1
- video receiver 2
- video receiver 3
Transcoder example

- Transcoding gateways
- Internet
- Video transmitter
- Receivers
Mixer example
Difference between streaming media applications and other data communication

• Traditional Internet applications (e-mail, Web, FTP, etc) use TCP, providing congestion control and reliability through retransmissions

• Video applications use RTP/UDP, with no built-in congestion control or reliability
Congestion control in TCP

- Slow start / congestion avoidance (Van Jacobson, 1988)
- AIMD: Additive increase, multiplicative decrease
Congestion control in real-time video applications

- Congestion control implemented at application level (ALF)
- AIMD works badly: dramatic rate changes gives poor perceptual quality
- Alternative methods: manual configuration, rate adaptive algorithms (TFRC)
TCP-Friendly Rate Control (TFRC)

- Rate control algorithm that tries to approximate TCP's performance over a long time period, but with smoother rate changes
- Equation-based
- Sending rate adjusted based on loss event rate, packet size and RTT

\[
\frac{T}{R_\text{c}} = \frac{s}{3t_\text{RTO}} - \sqrt{\frac{3p}{8}}(1 - \frac{2p}{32p^2})
\]
Resilience to packet loss

- The Internet is a "best effort" network
  - Applications must be resilient to packet loss
- Real time multimedia communication is very sensitive to packet loss
- Retransmissions not viable for delay-sensitive applications like videoconferencing
- Packet loss is caused by congestion
  - Congestion control and loss resilience are related topics
- Bit errors not an issue!
Adaptive Forward Error Correction

- Use Reed-Solomon erasure codes to protect video against packet loss
- Adapt the strength of the RS codes to the experienced loss rate
  - Receiver reports current loss rate to sender
- Let congestion control algorithm determine total bandwidth (video + FEC)
Adaptive FEC

- The $k$ data packet can be recreated from any $k$ out of the $n$ transmitted packets
- Can tolerate loss rates up to $1 - \frac{k}{n}$
- $(n, k)$ reassigned for each transmitted frame to match the loss rate as measured by the receiver
Video communication in networks with firewalls
Problem description

• Firewalls typically don't allow live video streams
• Re-configuration of firewall policies can be troublesome (but is almost always possible)
Problem description, cont.

- NAT - Network Address Translation
- Computers inside a firewall are often configured with private IP addresses that aren't globally routable
  - 10.0.0.0 - 10.255.255.255
  - 172.16.0.0 - 172.31.255.255
  - 192.168.0.0 - 192.168.255.255

- A NAT unit translates between private and public addresses, remapping port numbers
Types of firewalls

• Applikation level firewalls (run on your PC)
  – i.e. Microsoft’s ”Personal Firewall”, Appgate’s ”AppGate Personal Firewall”, Symantec’s ”Norton Personal Firewall”

• Network level firewalls
  – Dedicated units filtering out unwanted traffic based on the packets' destination address, source address, port number and protocol

Example of a simple network level firewall
Microsoft’s Personal Firewall

Vissa funktioner i det här programmet har inaktiverats för att skydda datorn.

Vill du fortsätta att blockera det här programmet?

Namn: confero
Utgivare: Alkit Communications

Fortsätt blockera Häv blockering Fråga igen senare

Det här programmet har blockerats så att det inte kan ta emot anslutningar från Internet eller från nätverket. Om du känner igen programmet eller litar på utgivaren kan du häva blockeringen. När kan det vara bra att häva blockering av ett program?
Firewall configuration

- Configuration option usually called "port redirection" or "virtual server"
- Instruct firewall to allow the wanted traffic to your computer

Example of web-based firewall configuration interface
Examples of video-mediated communication in e-Health applications
Remote echocardiography over the Internet for diagnosing heart disease

A video signal showing the patient and the robot holding the ultrasound probe

A remotely controlled robot holding the ultrasound probe

The ultrasound signal

Real time data in the shape of video signals, ultrasound signal, and the remote control data are communicated over the network

A virtual 3D representation of the robot, showing its current position

The robot is remotely controlled with a joystick

The operator (the heart disease specialist) also communicates verbally with the patient

A remotely controlled robot holding the ultrasound probe

Internet
Remote echocardiography

The ultrasound probe

The probe and the bed

The robot

The remote control device for the robot
Pediatric cardiology

- Tests performed between Sunderbyn hospital in Luleå and Sahlgrenska university hospital in Gothenburg
  - Audio/video for interpersonal communication
  - Live ultrasound video for specialist support in critical situations
  - Sharing of stored ultrasound clips for collaborative analysis
Ear/nose/throat

- Tests performed between Luleå and Piteå
  - Audio/video for interpersonal communication and overview
  - High quality video for endoscopy
  - Stroboscopic examinations of movements of vocal chords

Broadband network

Luleå

Piteå
Field emergency support

- Mobile systems for emergency support
- Real-time collaboration using wireless video and audio
- Wearable computer with head-mounted display and camera
Remote medical auscultations over the Internet

- Separate audio channels for voice / auscultation
- Video + graphics support
- Electronic stethoscope
- IP/UDP/RTP
Telemedicine in surgery

High quality video from one or more operating rooms are distributed over a broadband network to one or more lecture halls for educational purposes.

The auditorium can follow multiple ongoing operations, and ask questions directly to the surgeons.

Example from Östra sjukhuset, Göteborg
Both open surgery and minimal invasive surgery
Several operations at the same time
The audience can choose which operation to follow, and ask the performing surgeon questions in real time.
Stereoscopic video communication

- Two cameras mimicking our two eyes
- Gives true depth perception through stereopsis
- Higher realism
- Beneficial when discussing and interacting with physical objects (e.g. a physical mockup in a product development project) or when navigating in a physical world (e.g. controlling a robot)