Course Presentation

Multimedia Systems
Multimedia Networking
Part II
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TCP/UDP/IP: “best-effort service”

No guarantees on delay, loss

Best effort delivery describes a network service in which the network does not provide any guarantees that data is delivered or that a user is given a guaranteed quality of service level or a certain priority. In a best effort network all users obtain best effort service, meaning that they obtain unspecified variable bit rate and delivery time, depending on the current traffic load.

But multimedia apps requires QoS and level of performance to be effective!

Today’s Internet multimedia applications use application-level techniques to mitigate (as best possible) effects of delay, loss.
Multimedia Networking, QoS

Multimedia Over Today’s Internet

How should the Internet evolve to better support multimedia?

- **Integrated services** philosophy:
  - **Fundamental changes** in Internet so that apps can reserve end-to-end bandwidth.
  - Requires new, complex software in hosts & routers.

- **Differentiated services** philosophy:
  - **Fewer changes** to Internet infrastructure, yet provide 1st and 2nd class service (traffic in classified and the traffic in the higher class is given priority).
Multimedia Networking, QoS

Integrated Services (IntServ)

- IP: best-effort delivery
  - Means that it does not guarantee the minimum of a service, such as bandwidth, to applications such as real-time audio and video.

- IntServ is a flow-based QoS model designed for IP:
  - Means that a user needs to create a flow, a kind of virtual circuit, from the source to the destination and inform all routers of the resource requirement.

Reservation protocol (RSVP) is used to install the reservation state along that path. After reservation, the application can start to send traffic over the path for which it has exclusive use of the resources.
Differentiated Services (DiffServ)

- Was introduced by the IETF (Internet Engineering Task Force) to handle the shortcomings of Integrated Services.
- DiffServ is a class-based QoS model designed for IP:
  - e.g. can be used to provide low-latency to critical network traffic such as voice or video while providing simple best-effort traffic guarantees to non-critical services such as web traffic or file transfers.
- DiffServ uses the 6-bit Differentiated Services Code Point (DSCP) field in the header of IP packets for packet classification purposes. DSCP replaces the outdated IP precedence, a 3-bit field in the Type of Service (ToS) byte of the IP header originally used to classify and prioritize types of traffic.
- All the policing and classifying is done at the boundaries between DiffServ clouds. This means that in the core of the Internet, routers can get on with doing the job of routing, and not care about the complexities of collecting payment or enforcing agreements.
IntServ vs. DiffServ

IntServ:
- Requires advance setup, reservation, and time-consuming end-to-end negotiation for each flow. As a result, IntServ is applicable to long lasting traffic (e.g. video conferencing).
- Many states must be stored in each router. As a result, IntServ works on a small-scale.

DiffServ:
- Is relatively easy to implement.
- Diffserv operation only works if the boundary hosts honour the policy agreed upon. A host may tag its own traffic with a higher precedence, even though the traffic doesn't qualify to be handled with that importance.
Real-time Interactive Multimedia and Internet Telephony as an Example
Real-time Interactive Multimedia Protocol: Why not TCP or UDP?

- **TCP/IP**
  - Not suitable for real-time.
  - **Retransmissions** can lead to high delay and cause delay jitter.
  - Does not support multicast.
  - **Congestion control** mechanism (slow start) not suitable for AV media.

- **UDP/IP**
  - No defined technique for synchronizing.
  - Streams from different servers may collide.
  - A feedback channel must be defined for quality control.
Real-time Interactive Multimedia

RTP (Real-time Transport Protocol)

- Designed for the transport of real-time data such as audio and video streams. RTP stands between UDP and the application program.

Data encapsulation example up to IP layer
RTP (Real-time Transport Protocol)

- The main contributions of RTP (Features)
  - Creates its own timestamping and sequencing mechanisms to ensure the ordering.
  - **Mixing**: Combining several streams of traffic into one stream.
  - Payload type identification
  - Multicasting
  - Delivery monitoring
- Used in Netscape LiveMedia, Microsoft Netmeeting, and Intel Videophone.
Real-time Interactive Multimedia

RTP [Supplementary Materials]

RTP packet header format

<table>
<thead>
<tr>
<th>Ver</th>
<th>P</th>
<th>X</th>
<th>Contr. count</th>
<th>M</th>
<th>Payload type</th>
<th>Sequence number</th>
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<tbody>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>Timestamp</td>
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<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>Synchronization source identifier</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>Contributor identifier</td>
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<td></td>
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<td></td>
<td></td>
<td></td>
<td></td>
<td>Contributor identifier</td>
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Table 25.1 Payload Types

<table>
<thead>
<tr>
<th>Type</th>
<th>Application</th>
<th>Type</th>
<th>Application</th>
<th>Type</th>
<th>Application</th>
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<tbody>
<tr>
<td>0</td>
<td>PCMµ Audio</td>
<td>7</td>
<td>LPC audio</td>
<td>15</td>
<td>G728 audio</td>
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<td>1</td>
<td>1016</td>
<td>8</td>
<td>PCMA audio</td>
<td>26</td>
<td>Motion JPEG</td>
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<td>2</td>
<td>G721 audio</td>
<td>9</td>
<td>G722 audio</td>
<td>31</td>
<td>H.261</td>
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<td>3</td>
<td>GSM audio</td>
<td>10–11</td>
<td>L16 audio</td>
<td>32</td>
<td>MPEG1 video</td>
</tr>
<tr>
<td>5–6</td>
<td>DV14 audio</td>
<td>14</td>
<td>MPEG audio</td>
<td>33</td>
<td>MPEG2 video</td>
</tr>
</tbody>
</table>
Real-time Interactive Multimedia

RTP [Supplementary Materials]

- **Sequence number field**
  - Incremented for each RTP packet

- **Synchronization SouRCe (SSRC) field**
  - Uniquely identifies the source in the session (e.g. in AV Conferencing, both audio and video are transmitted as two separate RTP sessions).

- **Contributing SouRCe (CSRC) and CC fields**
  - Used by a mixer to identify the contributing sources
  - Size of the list given by the CSRC Count (CC) field
Real-time Interactive Multimedia

RTCP (Real Time Control Protocol)

- **RTP** ≡ *data messages*
  - RTP allows only one type of message, one that carries data from the source to the destination.

- **RTCP** ≡ *control messages*
  - In many cases, there is a need for other messages in a session. These messages **control the flow and quality of data** and allow the recipient to send feedback to the source or sources. Real-Time Transport Control Protocol (RTCP) is a protocol designed for this purpose.

![RTCP Message types](image.png)

- Sender report 200
- Receiver report 201
- Source description message 202
- Bye message 203
- Application specific message 204
RTP and RTCP [Supplementary Materials]

- RTP uses a temporary even-numbered UDP port.
- RTCP uses an odd-numbered UDP port number that follows the port number selected for RTP.
- RTCP message types:
  - SR: Sender report, for transmission and reception statistics from participants that are active senders.
  - RR: Receiver report, for reception statistics from participants that are not active senders.
  - SDES: Source description items.
  - BYE: Indicates end of participation.
  - APP: Application-specific functions.
Multimedia Networking

Internet Telephony, Pros

- VoIP services convert voice into digital signal that travels over the internet.

- Main advantages of Internet telephony over POTS (Plain Old Telephone Service):
  - Uses packet-switching | network usage is much more efficient (voice communication is bursty and VBR encoded).
    - Save money.
    - Easier to install and upgrade, Fast setup.
    - With the technologies of multicast or multipoint communication, multi-party calls are not much more difficult than two-party calls.
Multimedia Networking

Internet Telephony, Pros

- With advanced multimedia data compression techniques, various degrees of QoS can be supported and dynamically adjusted according to the network traffic.

- Good graphics user interfaces can be developed to show available features and services, monitor call status and progress, etc.

- Integration: Keeping voice and its pertinent data on the same lines.

- Different devices can communicate with each other (PC to PC, Phone to PC, PC to Phone, Phone to Phone).

- Currently VoIP has been added to many mobile devices and cell phones
  - Seamless mobility (Location independent).
Multimedia Networking

Internet Telephony

- **RTP and RTCP**: The transport of real-time audio (and video) in Internet telephony is supported by RTP (whose control protocol is RTCP).

- **RTSP and RSVP**: Streaming media is handled by RTSP and Internet resource reservation is taken care of by RSVP.

- **Two Voice over IP (VOIP) Protocols**:
  - **H.323**
  - **SIP (Session Initiation Protocol)**

**H.323**: A standard for packet-based multimedia communication services over networks that do not provide a guaranteed QoS. It specifies the signaling protocols.

**SIP (Session Initiation Protocol)**: An application-layer control protocol in charge of the establishment and termination of sessions in Internet telephony.

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Network Protocol Structure for Internet Telephony

- **H.323 or SIP**
- **RTP, RTCP, RSVP, RTSP**
- **Transport layer (UDP, TCP)**
- **Network layer (IP, IP Multicast)**
- **Data link layer**
- **Physical layer**

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Multimedia Systems, Spring 2011, Mahdi Amiri, Multimedia Networking
Simple Session: Setting up a call to known IP address

INVITE message includes:
* Caller’s IP address and port number.
* Caller’s preferred encoding format (e.g. PCM ulaw)

OK message includes:
* Callee’s IP address and port number.
* Callee’s preferred encoding format.

Codec negotiation: Suppose callee doesn’t have caller's preferred encoder. callee will instead reply with '606 Not Acceptable Reply', listing his encoders. caller can then send new INVITE message, advertising different encoder.

Rejecting a call: Caller can reject with replies "busy", "gone", "payment required", and "forbidden".

Note: Media can be sent over RTP or some other protocol.
Internet Telephony

SIP, Vision

- **SIP long-term vision:**
  - All telephone calls, video conference calls take place over Internet.
  - People are identified by names or e-mail addresses, rather than by phone numbers.
  - You can reach callee, no matter where callee roams, no matter what IP device callee is currently using.

SIP URIs, a Uniform Resource Identifier (URI) is a string of characters used to identify a name or a resource on the Internet.
Internet Telephony

SIP Services

- Setting up a call
  - SIP provides mechanisms…
    - … for caller to let callee know she wants to establish a call.
    - … so caller and callee can agree on media type and encoding format.
  - … to end call.

- Determine current IP address of callee
  - Maps mnemonic identifier to current IP address.

- Call management
  - Add new media streams during call.
  - Change encoding during call.
  - Invite others.
  - Transfer, and hold calls.
SIP, Session Example 2

**SIP Session**: When caller only has callee’s name or e-mail address

**Registrar server**: Provides a location service which registers one or more IP addresses to a certain SIP Uniform Resource Identifier (URI). When callee starts SIP client, client sends SIP REGISTER message to callee’s registrar server.

**Proxy server**: Primarily plays the role of routing SIP messages.

Example: Caller: Alice, Callee: Bob
* Alice sends invite message to her proxy server contains address sip:bob@domain.com
* Proxy responsible for routing SIP messages to callee (possibly through multiple proxies).
* Callee sends response back through the same set of proxies.
* Proxy returns SIP response message to Alice (contains Bob’s IP address).

Note: Proxy is analogous to local DNS server.
1. Caller Sends INVITE john@home.edu
2. Proxy Uses DNS and sends request
3- 4. john@home.edu Isn't logged on request sent to location server john@work.edu located
5. return john@work.edu to proxy server
6. attempt next proxy server
7- 8. consults location server to discover John's local address john_doe@my.work.edu
9- 10. Proxy 3 forwards the invitation to the callee
11-14. John accepts the call and returns acknowledgment to caller
Internet Telephony

**H.323 vs. SIP**

- **H.323** is a complete, vertically integrated suite of protocols for multimedia conferencing: signaling, registration, admission control, transport, codecs.

- **SIP** is a single component. Works with RTP, but does not mandate it. Can be combined with other protocols, services.

- **H.323** comes from Telecommunication Standardization Sector (ITU-T), **SIP** comes from The Internet Engineering Task Force (IETF)

- **SIP** has Web flavor, whereas **H.323** has telephony flavor.

- **SIP** borrows much of its concepts from HTTP.

- **SIP** uses the KISS principle: Keep it **simple** stupid.

Find out more at: [http://www.packetizer.com/ipmc/h323_vs_sip/](http://www.packetizer.com/ipmc/h323_vs_sip/)
### H.32X Standards

<table>
<thead>
<tr>
<th>ITU-T recommendation</th>
<th>Underlying Network over which audio, video and data conferencing is provided.</th>
</tr>
</thead>
<tbody>
<tr>
<td>H.320</td>
<td>ISDN</td>
</tr>
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<td>H.321 and H.310</td>
<td>ATM</td>
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<td>H.322</td>
<td>LAN’s that provide a guaranteed QoS</td>
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<tr>
<td>H.323</td>
<td>LAN’s and Internet</td>
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<td>H.324</td>
<td>PSTN/Wireless</td>
</tr>
</tbody>
</table>

**ITU-T recommendations for Audio/Video/Data conferencing standards**
Internet Telephony

H.323 Components

- H.323 Gatekeeper
- H.323 Terminal
- H.323 Gateway
- H.323 Zone
- Multipoint Control Unit
- Public Switch Telephone Network
- PSTN
Internet Telephony

H.323 Components

Terminals

- These are the endpoints of the H.323 conference.
- Can be a software or a special device
- A multimedia PC with a H.323 compliant stack can act as a terminal.

Functions:

- Initiating and receiving notifications
- Sending and receiving data
Internet Telephony

H.323 Components

Gateway

- Responsible for connecting the telephone IP network to other types of network (Gateway is only needed whenever conferencing needs to be done between different H.32X-based clients).
Internet Telephony

H.323 Components

♦ Gatekeeper
  ♦ Supervises all telephone conversations carried out in a zone.
  ♦ Admission control (Terminals must get permission from the gatekeeper to place any call).
  ♦ Bandwidth management (Control of the available band).

♦ Multi-point Control Unit (MCU)
  ♦ This is an optional component that provides point-to-multipoint conferencing capability to an H.323 enabled network.
Internet Telephony

An H.323 enabled network with different components

[Diagram showing H.323 components including Gatekeeper, Gateway, Terminal, MCU, ATM/B-ISDN, N-ISDN, Guaranteed QoS LAN, H.321 Terminal, H.320 Terminal, H.322 Terminal, and Packet based network like Ethernet LAN]
Internet Telephony

H.323 Protocol Stack

Audio
- Audio Codecs: G.711, G.722, G.723.1, G.728, G.729
- G.726

Video
- Video Codecs: H.261, H.263

A/V control
- RTP
- RTCP

Control
- H.225 Registration, Admission, Status
- Q.931 Call Setup/Teardown
- H.245 Connection Negotiation Protocol

Data
- T.120 Real-Time Data Conferencing Protocol stack

UDP

TCP

IP
Internet Telephony

H.323 Protocol Stack

- **RTP and RTCP**: RTP and its associated control protocol, RTCP, are employed for timely and orderly delivery of packetized audio/video streams.

- **The H.225 RAS (Registration, Admission and Status)** is mainly used by H.323 end-points (terminals and gateways) to discover a gatekeeper, register/un-register with the gatekeeper, requesting call admission and bandwidth allocation and clearing a call. The gatekeeper can also use this protocol for inquiring on an end-point and for communicating with other peer gateways.

- **The Q.931 signaling protocol** is used for call setup and teardown between two H.323 end-points and is a lightweight version of the Q.931 protocol defined for PSTN/ISDN.

- **The H.245 media control protocol** is used for negotiating media processing capabilities such as audio/video codec to be used for each media type between two terminals and determining Master-Slave relationships.

- **T.120**: Real-time data conferencing capability is required for activities such as application sharing, whiteboard sharing, file transfer, fax transmission, and instant messaging. Recommendation T.120 provides this optional capability to H.323. T.120 is a real-time data communication protocol designed specifically for conferencing needs. Like H.323, Recommendation T.120 is an umbrella for a set of standards that enable the real-time sharing of specific applications data among several clients across different networks.
Internet Telephony

VoIP Terminals

- Modem Adapter
  - Vonage.

- Computer Adapter
  - Magic Jack.

- Software
  - Skype.
Internet Telephony

Example VoIP Softwares

- **Skype**
  - **Skype** runs on a closed proprietary network, though the network (but not the official Skype client software) also supports SIP clients.

- **Blink**
  - Open source **SIP** client.

- **Ekiga**
  - Open source, Supports both the **SIP** and **H.323**.

- **Yahoo! Messenger**
  - Closed **Proprietary**, **SIP**.

Internet Telephony, Cons

- Quality (Packet switching network related cons):
  - Latency, Jitter, Loss, etc.
- Security concerns (Unsecure):
  - Privacy.
  - Ensure correct billing.
  - Protection from Denial of Service (DoS).
  - Protection from packet manipulation.
  - Solution: Encrypting traffic, using firewalls.
Multimedia Networking

Internet Telephony, Cons

- Reliability (Not as reliable)
  - Phone Network = wire.
  - Data Network = data switches, higher probability of failure.
- Power dependent
  - Not suitable for emergency calls
- For many organizations switching to IP phones is not cost effective.
- Censorship
Thank You

Next Session: Critical Reading Presentations

FIND OUT MORE AT...

1. http://ce.sharif.edu/~m_amiri/