Outline

• Classic view: different types of wires
  – Copper: telephone, modem, xDSL
  – Cable: TV driven
  – Fiber: future proofing
  – (Wireless: satellite and terrestrial)

• Triple play: internet, voice, and TV

Many Types of Access

Modems

• Modem offers a bit stream.
  – Aggressive signal processing has dramatically increased the available throughput - beats the Nyquist limit!

• SLIP: Serial Line IP.
  – Protocol to sent IP packets with minimum framing
  – Lacks authentication, error detection, non-IP support, ...

• PPP: Point-to-Point Packets.
  – Better framing, error control, and testing support
  – Can negotiate choice of higher layer protocols, IP address
  – Can support unreliable and reliable transmission
Integrated Services Digital Network (ISDN)

- ISDN integrates voice and data services.
- Provides a set of bit pipes that can be used for voice, data, and signaling.
  - Implemented by using time multiplexing
- Basic rate ISDN offers 64Kbps data bit pipes and one 16 Kbps signaling channel.

Digital Subscriber Line

- Squeeze more bandwidth out of the telephone line using advanced signal processing.
- Asymmetric digital subscriber line (ADSL).
  - More "download" bandwidth, e.g., video on demand or web surfing
  - Initially: T1 incoming path, 64 Kbps outgoing path – now much higher bandwidths
- (Symmetric) digital subscriber line (DSL).
  - Same bandwidth both ways, e.g., 768 Kbps

DSL: Physical Layer Matters

- Telephone wiring was designed to carry a telephone signal
  - Carry analog voice signal in 0 – 4 KHz
  - 1-pair of voice-grade shielded twisted pair (UTP)
  - Ends of wiring were conditioned for optimizing low frequencies – cuts off higher frequencies
- Changes needed for higher frequencies
  - Change conditioning at the end points
  - Better coding and modulations
  - Bandwidth depends on the distance
  - Can upgrade to better wiring, e.g., 2-pair DG UTP
  - Data uses higher frequencies on the wire

DSL Speeds

<table>
<thead>
<tr>
<th>Distance (Feet)</th>
<th>Performance (kbps)</th>
</tr>
</thead>
<tbody>
<tr>
<td>0</td>
<td>10000</td>
</tr>
<tr>
<td>1000</td>
<td>9000</td>
</tr>
<tr>
<td>2000</td>
<td>8000</td>
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<td>3000</td>
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</tr>
</tbody>
</table>

(Asymmetric Digital Subscriber Line (ADSL))

- VDSL2 Downstream
- ADSL2+ Downstream
- ADSL Downstream
Cable Modem

- Use cable infrastructure for data service.
  - Inherently has more bandwidth
- The last mile is a shared infrastructure that was designed for broadcasting.
  - Meaning: the bandwidth is shared by users
  - Example: 27 Mbs shared incoming path; 768 Kbs common outgoing path

Fiber – “FTTX”

- Traditionally, fiber only used in network “core”
  - More expensive technology – higher capacity
- Reach of fiber has expanded over time, i.e., fiber reaches closer to the consumer
  - Fiber to the cabinet
  - Fiber to the curb
  - Fiber to the home
  - Options include “active” and “passive”
- Trend applies to all copper technologies
  - Cable, twisted pair

Cable versus Fiber

- Cable Modem Network
  - Simplex 6 MHz downstream channels
  - Simplex 200 KHz to 6+ MHz upstream channels
  - All traffic traverses the Headend

Comparison

- Modems use “worst case” technology.
  - Has to fit within any voice channel so encoding suboptimal
  - Wires can be very long (end-to-end)
- ISDN can be more aggressive – dated quickly
- DSL is highly optimized for the transmission medium
  - But there are some constraints on distance
- Cable modem uses a transmission medium that has inherently a higher bandwidth, but the network architecture will limit throughput.
  - Designed for broadcasting, not for point-point connections
- Fiber has high capacity but is a big investment
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  - Motivation
  - Media encoding and compression
  - Streaming protocols: SIP and RTP
  - IP distribution: IP multicast

Change in User Demand: Triple Play

- Cable companies already offer TV and data service – natural add phone service
  - Very low bandwidth
  - Plays well in market – convenient and cost effective
  - Services emerged as a fourth component
- What about telephone companies
  - How do you deliver TV service over a phone wire?
  - Necessary to be competitive
- How about incumbents?
  - They start with zero wires

Delivering “Triple Play” to Consumers

- Can be viewed at the 1990’s dream of integrated services networks: voice, video, and data
- IPTV delivery over DSL is optimized to work around last mile bottleneck
  - Rest of the lecture
- Cable and fiber have plenty of bandwidth – channels can be dedicated to specific uses
  - Take “TV” channels and use for Internet service and voice
  - Demand for large numbers of channels is starting to stress even cable capacity: deliver less unpopular channels “on demand”, DSL style (switched video)

Steps in Encoding and Decoding

- Process has three steps on each side
- Digitize: represent information in bits
  - Sample, quantize, (eg. PCM)
- Compress: reduce number of bits
  - Audio: GSM, G.729, G.723.3, MP3, …
  - Video: MPEG 1/2/4, H.261, …
- Send over the network
- Reverse process on receive side: uncompress, convert, play – need to match!
Audio Encoding

- Traditional telephone quality encoding: 8KHz samples of 8 bits each – 64 Kbps
- CD quality encoding: 44.1 KHz of 16 bits – 1.41 Mbs uncompressed
- MP3 compression similar to MPEG
  - Frequency ranges that are divided in blocks

<table>
<thead>
<tr>
<th>Layer</th>
<th>Range</th>
<th>Ratio</th>
</tr>
</thead>
<tbody>
<tr>
<td>Layer 1</td>
<td>384 kbps</td>
<td>4</td>
</tr>
<tr>
<td>Layer 2</td>
<td>192 kbps</td>
<td>8</td>
</tr>
<tr>
<td>Layer 3</td>
<td>128 kbps</td>
<td>12</td>
</tr>
</tbody>
</table>

Video Encoding

- Captured frames ("raw" video) are very large:
  - 320 x 240 x 24-bit color = 230,400 bytes/frame
  - 15 frames/second = 3,456,000 bytes/second
  - 10 seconds takes around 30 Mbytes! (no audio)
- Commonly-used encoding “tricks”:
  - Per-frame versus inter-frame encoding
    - Leverage fact that successive frames tend to be similar
    - Example: MPEG uses I, P, and B frames (video lecture)
    - Impacts latency
  - Layered encoding
    - Quality improves as you decode more layers

Representative Video Bit Rates

- 1.2 Gbps Uncompressed HDTV
- 19.4 Mbps ATSC (≈ HDTV quality)
- 8 - 9 Mbps MPEG4 (≈ HDTV quality)
- 90 Mbps Uncompressed NTSC (SDTV)
- 3 - 6 Mbps MPEG2 (≈ SDTV quality)
- 1.5 Mbps MPEG4 (≈ SDTV quality)
- 1.5 Mbps MPEG1 (≈ VHS < SDTV quality)
- Note: ATSC, MPEG2, & MPEG4 support a wide variety of formats (SDTV ↔ HDTV)

Codecs

- G.711
- G.726
- G.729
- G.723.1
- Audio CD
- MP3
- AAC
- Alcatel Propr. Codec

Source: Alcatel
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Different Classes of Streaming

- Multimedia streaming covers audio and video
- Playback: you play back stored content
  - Full content is available up front
  - Flexibility in when content is transmitted
- Broadcast: transmission of live content
  - Content generated on the fly but flexibility in playback delay – seconds in practice
  - VCR/DVR functionality adds a twist
- Interactive: voice and video conferencing
  - Latency is very critical
  - Impacts entire system: encoding, protocols, end-system design, ...

Transport Protocol Properties

- Reliability.
  - Some lost data may be acceptable – depends on encoding and user expectations
  - Timeouts typically result in unacceptable delay – there may not be enough time to retransmit data
- Congestion control.
  - Nature of the flow fundamentally limits its bandwidth
  - Reduction of rate in response to congestion should reduce data size, not transmit rate for samples
    - E.g. change frame rate or frame size
- Flow control: natural pacing.
  - Samples should be paced at the rate of the data
  - Too slow --> underflow and missed deadlines
  - Too fast --> buffer overflow and lost data

Real Time Transport Protocol (RTP)

- Multimedia senders append header fields before passing to transport layer
  - Format, sequence numbers, timestamps, ...
- RTP logically extends UDP: application layer between UDP and application
- RTP does not guarantee timely data delivery.
  - Simply helps applications with formatting and the collection of session information
  - Guarantees can only be provided at lower level
- The protocol has two parts: Real-time Transport Protocol (carry data) and Real-Time Control protocol (monitor quality, participant info, ...)
RTP Packet Format

- Source/Payload type
  - Different formats assigned different codes
  - Eg. GSM -> 3, MPEG Audio -> 14
- Sequence numbers
- Time stamps
- Synchronization source ID
- Miscellaneous fields, e.g., feedback

Real Time Control Protocol (RTCP)

- RTCP packets transmitted by each participant in RTP session to all others using multicast
- Distinct port number from RTP
- Reports on:
  - Loss rate
  - Inter-arrival jitter
  - Identity of receivers
  - Delay, (indirectly)
- Control bandwidth sharing
  - Needed for scalability, e.g., 5% of data bandwidth

SIP Introduction

- SIP is an application level signaling protocol for initiating, managing and terminating sessions in the Internet
  - Registrations, invitations, acceptations, and disconnections
  - Sessions may include text, voice, video
  - Can use unicast or multicast communication
  - Client-server model: request-reply transaction
- Common headers in plain text, similar to MIME/HTTP
  - request/response line (e.g., INVITE a@b.com SIP/2.0)
  - message headers (identification, routing, etc)
  - message body, e.g., session description protocol

SIP Entities

- User agent (UA)
  - UA client (UAC) and UA server (UAS)
- Proxy server
  - relay calls; chaining; forking
- Redirect server
  - redirect calls
- Registrar server
  - UA registration (UA whereabouts)
SIP Session Establishment and Call Termination

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SIP Call Redirection

IPTV - Internet Protocol Television

• Deliver TV broadcast to users using Internet topology
  – Also, services, data, and voice, i.e. triple play
• Design only delivers channels of interest to the home
  – Contrast with broadcast nature of cable design
  – Reduce load on the last mile link
• Uses IP multicast and RTP for distribution of broadcast TV
  – More on this later
• Other traffic is unicast, using different higher level protocols
  – Internet service, voice, VoD, interactive services
**TV over IP using DSL**

- DSL has limited bandwidth (few to few 10s of Mbps)
  - Depends on distance, technology
- Must limit the number of channels, other traffic
  - Read your contract carefully

**Multicast – Efficient Data Distribution**

- **IP Multicast Service Model (rfc1112)**
  - Each group identified by a single IP address
  - Groups may be of any size
  - Members of groups may be located anywhere in the Internet and can join and leave at will
  - Group membership not known explicitly
  - Analogy:
    - Each multicast address is like a radio frequency, on which anyone can transmit, and to which anyone can tune-in.
  - IP multicast not a good fit for Internet
    - Security and economic concerns, limited applicability
  - But perfect fit for IPTV!
    - IPTV membership is known, access controlled end-to-end

- **IP Multicast Addresses**
  - Class D IP addresses
    - 224.0.0.0 – 239.255.255.255
    - 1110 Group ID
  - How to allocated these addresses?
    - Well-known multicast addresses, assigned by IANA
    - Transient multicast addresses, assigned and reclaimed dynamically
    - Addresses centrally assigned in IPTV networks
Source-based Trees

Router
Source
Receiver

IP Multicast Service

• Sending – source sends one packet
• Group management – two new operations
  – Join-IP-Multicast-Group(group-address, interface)
  – Leave-IP-Multicast-Group(group-address, interface)
  – Uses Internet Group Management Protocol (IGMP)
• Receiving – same as regular unicast
• Making it work:
  – Routers must keep separate multicast forwarding table
  – Routers must replicate packets: list of egress ports
  – Routing protocol “connects” receiver to the “tree”

Multicast Routing Protocols

• Many routing protocols proposed for IP multicast
  – Flood and prune: broadcast pruned to multicast
    • “Dense mode” - not useful in Internet
  – Nodes send message towards some rendez-vous point
    • Connect to the tree on first router that is part of the tree
• IPTV network requirement are a good fit for multicast
  – Private network: full control over networks, endpoints
  – Single source and all destinations are known – routing is easy!
  – Switching channels: join new MC group and leave the old group – lots of optimizations needed to make it fast!

Controlling Quality of Service

• IPTV and voice have very high QoS requirements
  – Need to compete with traditional telephone and cable TV delivery based on circuits
  – Bandwidth requirements are known
• Data and interactive TV services have traditional web-like requirements
  – Optimize response time, e.g., browsing
  – Maintain bandwidth, e.g., Netflix
  – Bursty bandwidth requirements
• QoS control is based on careful bandwidth allocation across “flows” and enforcing bandwidth limits using traffic shapers
Credit

- Lecture includes slides from several sources:
  - home.ubalt.edu/abento/427/VOIPLASTMILE/VOIPLASTMILE.ppt
  - http://www.item.ntnu.no/fag/ttm7/Lectures/5_Convergence_IP_TV.ppt
  - www.okstate.edu/elec-engr/scheets/ecen5553/fall12/TCM2930W.PPT

The Last Mile

- One more lecture: IP forever?

- Final: similar to midterm
  - Closed book
  - Focus is on second half of the semester
  - First half limited to topics touched on in the midterm