Outline

- Classic view: different types of wires
  - Copper: telephone, modem, xDSL
  - Cable: TV driven
  - Fiber: future proofing
  - (Wireless: satellite and terrestrial)
- Media encoding and streaming
- Triple play: IPTV, cable, fiber to the home
Many a type 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 send 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 64Kbs data channels and one 16 Kbs 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 Kbs outgoing path – now much higher bandwidths
- (Symmetric) digital subscriber line (DSL).
  - Same bandwidth both ways, e.g., 768 Kbs
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 unshielded 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

<table>
<thead>
<tr>
<th>ADSL2+</th>
<th>Video/ voice/ data/copper</th>
<th>16 – 25 Mb/s Down</th>
<th>video capabilities. Dist 1.5 km/BW</th>
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<tr>
<td>G.992.5</td>
<td></td>
<td>Up to 800 Kb up</td>
<td>Higher-optional</td>
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<tr>
<td>ADSL2</td>
<td>Video/ voice/ data/copper</td>
<td>8 Mb/s down</td>
<td>video capab. Dist + 200m ADSL</td>
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<tr>
<td>With and without splitter – G.992.3 / 4</td>
<td>800 Kb/s up</td>
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<tr>
<td>ADSL</td>
<td>Voice/data/copper</td>
<td>Up to 8 Mb/s down</td>
<td>Full use existing copper. Web brows/Voice</td>
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<tr>
<td>With and without splitter – G.992.1 / 2</td>
<td>Up to 1.5 Mb/s up</td>
<td>2.7- 5.4 km</td>
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<tr>
<td>VDSL</td>
<td>Video/ voice/ data (V/V/D) over copper</td>
<td>Up to 52 Mb/s down asymm</td>
<td>Broadcast video, VoD, internet TV, 1.5 km - 300m</td>
</tr>
<tr>
<td>Symmetric Asymmetric G.993.1</td>
<td>Up to 26 Mb/s Symm.</td>
<td></td>
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<tr>
<td>Symmetric G.991.2</td>
<td></td>
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<tr>
<td>HDSL</td>
<td>Voice data and video 1/2/3 pairs</td>
<td>784 Kbit/s to 2320 Kbit/s</td>
<td>V/V/D Services mainly for business appl.</td>
</tr>
<tr>
<td>Symmetric G.991.1</td>
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</table>
Example of DSL Speeds

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
Outline

- Classic view: different types of wires
- Media encoding and streaming
  - Big Picture
  - Voice and video compression
  - Protocols: SIP and RTP
- Triple play: IPTV, cable, fiber to the home

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, …
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>
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<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>
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</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 (Hi ↓ Lo Quality)

- 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

Source: Alcatel

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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**

- **REGISTER**: register user agents
- **INVITE**: initiate calls
- **ACK**: confirm responses
- **BYE**: terminate or transfer calls
- **Other methods**:
  - CANCEL, OPTIONS, INFO, COMET, PRACK, SUBSCRIBE, NOTIFY, REFER
- **SIP response**: HTTP-like (e.g., SIP/2.0 200 OK)

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**SIP Call Redirection**

- **INVITE**: bob@acme.com
- **302**: Moved temporarily
  - contact: 3873972@gw.telco.com
- **ACK**
- **INVITE**: 3873972@gw.telco.com
- **200OK**
- **ACK**

From the RADVISION whitepaper on SIP
Outline

• Classic view: different types of wires
• Media encoding and streaming
• Triple play: IPTV, cable, fiber to the home
  • Multicast, traffic management
  • Trends for cable, fiber

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 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)

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
IPTV Components

- Streaming server: encoding of live content
  - Also, encryption (DRM), protocol support, e.g., for DVR, PIP, …
- IP network with multicast support
- DSLAM - Digital Subscriber Line Access Multiplexer
  - Mux/demux from/to multiple subscribers
- CPE - Customer Premises Equipment
  - Receives IP streams from DSLAM and distributes them throughout the home
- STB - Set Top Box
  - Decryption, decoding, D/A, channel control, …

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
- Members of groups can join and leave at will
- Senders need not be members
- 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.
  - For IPTV membership is known and access is controlled end-to-end
IP Multicast Addresses

- Class D IP addresses
  - 224.0.0.0 – 239.255.255.255

<table>
<thead>
<tr>
<th>1 1 1 0</th>
<th>Group ID</th>
</tr>
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</table>

- How to allocated these addresses?
  - Well-known multicast addresses, assigned by IANA
  - Transient multicast addresses, assigned and reclaimed dynamically
  - Addresses can be centrally assigned in IPTV networks

Source-based Trees

- Router
- Source
- Receiver
IP Multicast Service

- Sending – source sends one packet
- Receiving – two new operations
  - Join-IP-Multicast-Group(group-address, interface)
  - Leave-IP-Multicast-Group(group-address, interface)
  - Receive multicast packets for joined groups via normal IP-Receive operation
- Making it work:
  - Routers must replicate packets: list of egress ports
  - Receivers must be able to join a group – Internet Group Management Protocol (IGMP)
  - Routing protocol “connects” receiver to the “tree”

TV over IP using FTTH

- Pushing fiber to the home dramatically increases bandwidth, e.g., channels
  - Easy to justify for, e.g., apartment buildings
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 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