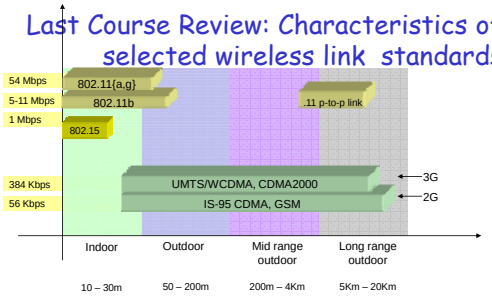
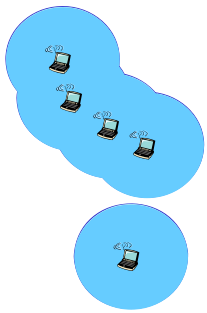
Last Course Review: Characteristics of selected wireless link standards



Elements of a wireless network



Ad hoc mode
- no base stations
- nodes can only transmit to other nodes within link coverage
- nodes organize themselves into a network: route among themselves

Wireless Link Characteristics

Differences from wired link ….

- decreased signal strength: radio signal attenuates as it propagates through matter (path loss)
- interference from other sources: standardized wireless network frequencies (e.g., 2.4 GHz) shared by other devices (e.g., phone); devices (motors) interfere as well
- multipath propagation: radio signal reflects off objects ground, arriving ad destination at slightly different times

…. make communication across (even a point to point) wireless link much more "difficult"

IEEE 802.11 Wireless LAN

- 802.11b
  - 2.4-5 GHz unlicensed radio spectrum
  - up to 11 Mbps
  - direct sequence spread spectrum (DSSS) in physical layer
    - all hosts use same chipping code
  - widely deployed, using base stations
- 802.11a
  - 5-6 GHz range
  - up to 54 Mbps
- 802.11g
  - 2.4-5 GHz range
  - up to 54 Mbps
- All use CSMA/CA for multiple access
- All have base-station and ad-hoc network versions

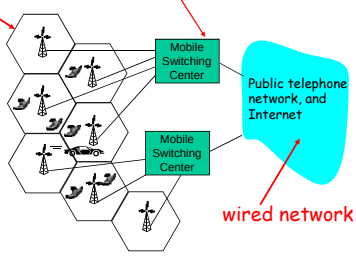
802.11: Channels, association

- 802.11b: 2.4GHz-2.485GHz spectrum divided into 11 channels at different frequencies
  - AP admin chooses frequency for AP
  - interference possible: channel can be same as that chosen by neighboring AP!
- host: must *associate* with an AP
  - scans channels, listening for *beacon frames* containing AP's name (SSID) and MAC address
  - selects AP to associate with
  - may perform authentication [Chapter 8]
  - will typically run DHCP to get IP address in AP's subnet

Components of cellular network architecture

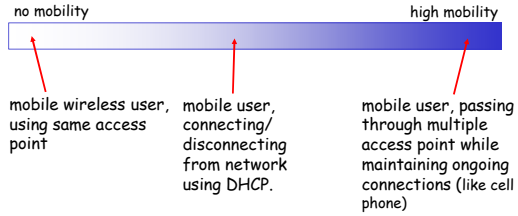
MSC
- connects cells to wide area net
- manages call setup (more later!)
- handles mobility (more later!)

cell
- covers geographical region
- *base station* (BS) analogous to 802.11 AP
- *mobile users* attach to network through BS
- *air-interface:* physical and link layer protocol between mobile and BS

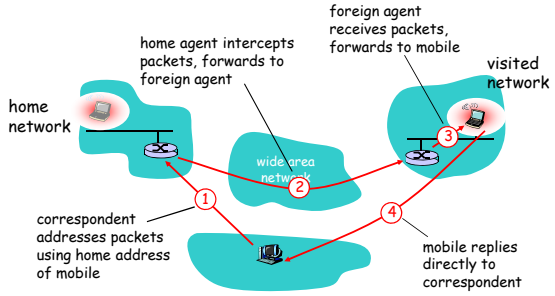


What is mobility?

- spectrum of mobility, from the *network* perspective:

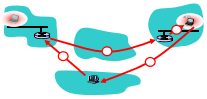


Mobility via Indirect Routing

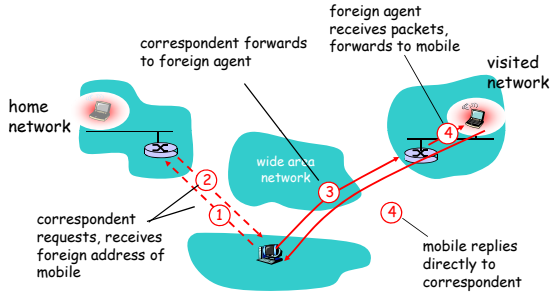


Indirect Routing: comments

- Mobile uses two addresses:
  - permanent address: used by correspondent (hence mobile location is *transparent* to correspondent)
  - care-of-address: used by home agent to forward datagrams to mobile
- foreign agent functions may be done by mobile itself
- triangle routing: correspondent-home-network-mobile
  - inefficient when correspondent, mobile are in same network

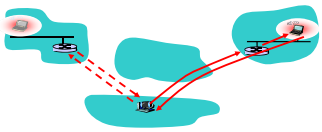


Mobility via Direct Routing

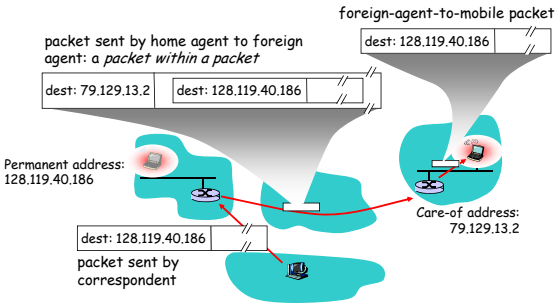


Mobility via Direct Routing: comments

- overcome triangle routing problem
- non-transparent to correspondent: correspondent must get care-of-address from home agent
  - what if mobile changes visited network?

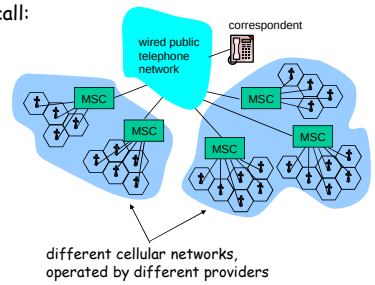


Mobile IP: indirect routing



Components of cellular network architecture

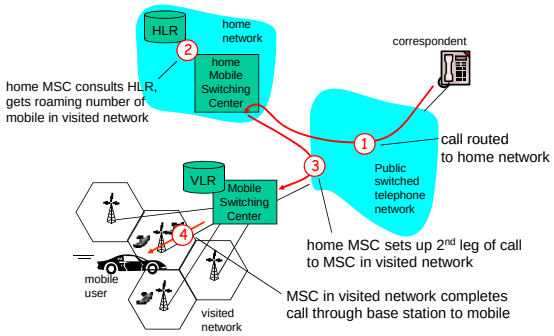
recall:



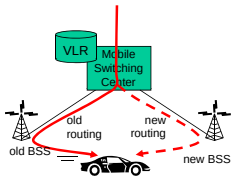
Handling mobility in cellular networks

- *home network:* network of cellular provider you subscribe to (e.g., Sprint PCS, Verizon)
  - *home location register (HLR):* database in home network containing permanent cell phone #, profile information (services, preferences, billing), information about current location (could be in another network)
- *visited network:* network in which mobile currently resides
  - *visitor location register (VLR):* database with entry for each user currently in network
  - could be home network

GSM: indirect routing to mobile

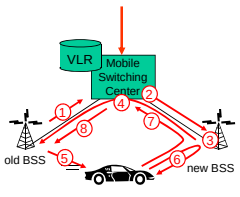


GSM: handoff with common MSC



- Handoff goal: route call via new base station (without interruption)
- reasons for handoff:
  - stronger signal to/from new BSS (continuing connectivity, less battery drain)
  - load balance: free up channel in current BSS
  - GSM doesn't mandate why to perform handoff (policy), only how (mechanism)
- handoff initiated by old BSS

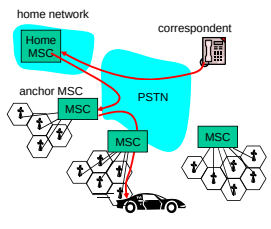
GSM: handoff with common MSC



1. old BSS informs MSC of impending handoff, provides list of 1+ new BSSs
2. MSC sets up path (allocates resources) to new BSS
3. new BSS allocates radio channel for use by mobile
4. new BSS signals MSC, old BSS: ready
5. old BSS tells mobile: perform handoff to new BSS
6. mobile, new BSS signal to activate new channel
7. mobile signals via new BSS to MSC: handoff complete. MSC reroutes call

8 MSC-old-BSS resources released

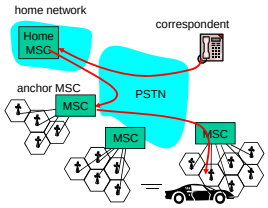
GSM: handoff between MSCs



(a) before handoff

- *anchor MSC:* first MSC visited during cal
  - call remains routed through anchor MSC
- new MSCs add on to end of MSC chain as mobile moves to new MSC
- IS-41 allows optional path minimization step to shorten multi-MSC chain

GSM: handoff between MSCs



(b) after handoff

- *anchor MSC:* first MSC visited during cal
  - call remains routed through anchor MSC
- new MSCs add on to end of MSC chain as mobile moves to new MSC
- IS-41 allows optional path minimization step to shorten multi-MSC chain

Mobility: GSM versus Mobile IP

GSM element	Comment on GSM element	Mobile IP element
Home system	Network to which the mobile user's permanent phone number belongs	Home network
Gateway Mobile Switching Center, or "home MSC". Home Location Register (HLR)	Home MSC: point of contact to obtain routable address of mobile user. HLR: database in home system containing permanent phone number, profile information, current location of mobile user, subscription information	Home agent
Visited System	Network other than home system where mobile user is currently residing	Visited network
Visited Mobile services Switching Center. Visitor Location Record (VLR)	Visited MSC: responsible for setting up calls to/from mobile nodes in cells associated with MSC. VLR: temporary database entry in visited system, containing subscription information for each visiting mobile user	Foreign agent
Mobile Station Roaming Number (MSRN), or "roaming number"	Routable address for telephone call segment between home MSC and visited MSC, visible to neither the mobile nor the correspondent.	Care-of-address

Wireless, mobility: impact on higher layer protocols

- logically, impact *should* be minimal …
  - best effort service model remains unchanged
  - TCP and UDP can (and do) run over wireless, mobile
- … but performance-wise:
  - packet loss/delay due to bit-errors (discarded packets, delays for link-layer retransmissions), and handoff
  - TCP interprets loss as congestion, will decrease congestion window un-necessarily
  - delay impairments for real-time traffic
  - limited bandwidth of wireless links

Chapter 6 Summary

Wireless

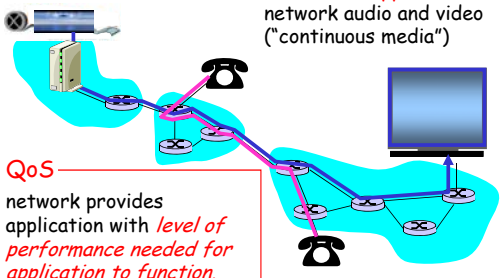
- wireless links:
  - capacity, distance
  - channel impairments
  - CDMA
- IEEE 802.11 ("wi-fi")
  - CSMA/CA reflects wireless channel characteristics
- cellular access
  - architecture
  - standards (e.g., GSM, CDMA-2000, UMTS)

Mobility

- principles: addressing, routing to mobile users
  - home, visited networks
  - direct, indirect routing
  - care-of-addresses
- case studies
  - mobile IP
  - mobility in GSM
- impact on higher-layer protocols

Multimedia, Quality of Service: What is it?

Multimedia applications: network audio and video ("continuous media")



QoS

network provides application with *level of performance needed for application to function.*

Chapter 7: Goals

Principles

- Classify multimedia applications
- Identify the network services the apps need
- Making the best of best effort service
- Mechanisms for providing QoS

Protocols and Architectures

- Specific protocols for best-effort
- Architectures for QoS

Chapter 7 outline

- 7.1 Multimedia Networking Applications
- 7.2 Streaming stored audio and video
- 7.3 Real-time Multimedia: Internet Phone study
- 7.4 Protocols for Real-Time Interactive Applications
  - RTP,RTCP,SIP
- 7.5 Distributing Multimedia: content distribution networks
- 7.6 Beyond Best Effort
- 7.7 Scheduling and Policing Mechanisms
- 7.8 Integrated Services and Differentiated Services
- 7.9 RSVP

MM Networking Applications

Classes of MM applications:

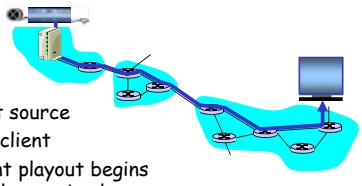
1) Streaming stored audio and video
2) Streaming live audio and video
3) Real-time interactive audio and video

Jitter is the variability of packet delays within the same packet stream

Fundamental characteristics:

- Typically **delay sensitive**
  - end-to-end delay
  - delay jitter
- But **loss tolerant**: infrequent losses cause minor glitches
- Antithesis of data, which are loss intolerant but delay tolerant.

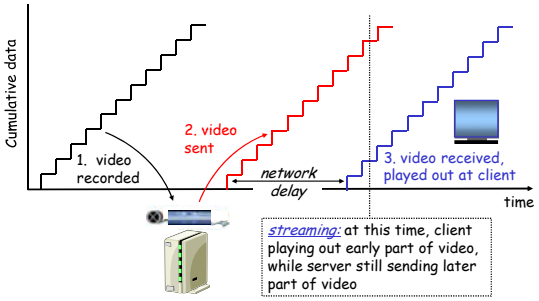
Streaming Stored Multimedia



Streaming:

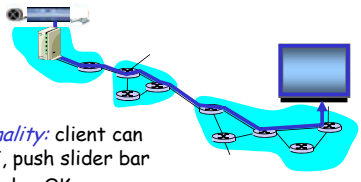
- media stored at source
- transmitted to client
- streaming: client playout begins *before* all data has arrived
  - timing constraint for still-to-be transmitted data: in time for playout

Streaming Stored Multimedia: What is it?



streaming: at this time, client playing out early part of video, while server still sending later part of video

Streaming Stored Multimedia: Interactivity



- *VCR-like functionality:* client can pause, rewind, FF, push slider bar
  - 10 sec initial delay OK
  - 1-2 sec until command effect OK
  - RTSP often used (more later)
- timing constraint for still-to-be transmitted data: in time for playout

Streaming Live Multimedia

Examples:

- Internet radio talk show
- Live sporting event

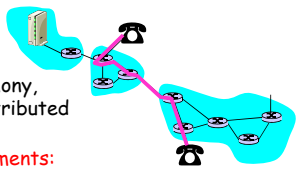
Streaming

- playback buffer
- playback can lag tens of seconds after transmission
- still have timing constraint

Interactivity

- fast forward impossible
- rewind, pause possible!

Interactive, Real-Time Multimedia



- applications: IP telephony, video conference, distributed interactive worlds
- end-end delay requirements:
  - audio: < 150 msec good, < 400 msec OK
    - includes application-level (packetization) and network delays
    - higher delays noticeable, impair interactivity
- session initialization
  - how does callee advertise its IP address, port number, encoding algorithms?

Multimedia Over Today's Internet

TCP/UDP/IP: "best-effort service"

- *no* guarantees on delay, loss

But you said 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

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

Laissez-faire

- no major changes
- more bandwidth when needed
- content distribution, application-layer multicast
  - application layer

Differentiated services philosophy:

- Fewer changes to Internet infrastructure, yet provide 1st and 2nd class service.

What's your opinion?

A few words about audio compression

- Analog signal sampled at constant rate
  - telephone: 8,000 samples/sec
  - CD music: 44,100 samples/sec
- Each sample quantized, i.e., rounded
  - e.g., 2^8=256 possible quantized values
- Each quantized value represented by bits
  - 8 bits for 256 values
- Example: 8,000 samples/sec, 256 quantized values --> 64,000 bps
- Receiver converts it back to analog signal:
  - some quality reduction

Example rates

- CD: 1.411 Mbps
- MP3: 96, 128, 160 kbps
- Internet telephony: 5.3 - 13 kbps

A few words about video compression

- Video is sequence of images displayed at constant rate
  - e.g. 24 images/sec
- Digital image is array of pixels
- Each pixel represented by bits
- Redundancy
  - spatial
  - temporal

Examples:

- MPEG 1 (CD-ROM) 1.5 Mbps
- MPEG2 (DVD) 3-6 Mbps
- MPEG4 (often used in Internet, < 1 Mbps)

Research:

- Layered (scalable) video
  - adapt layers to available bandwidth

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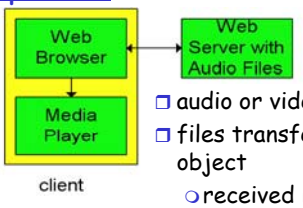
Streaming Stored Multimedia

Application-level streaming techniques for making the best out of best effort service:

- client side buffering
- use of UDP versus TCP
- multiple encodings of multimedia

Media Player
- jitter removal
- decompression
- error concealment
- graphical user interface w/ controls for interactivity

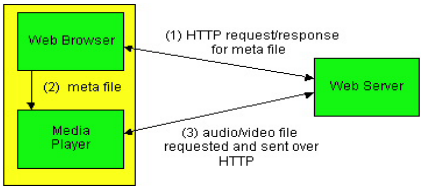
Internet multimedia: simplest approach



- audio or video stored in file
- files transferred as HTTP object
  - received in entirety at client
  - then passed to player

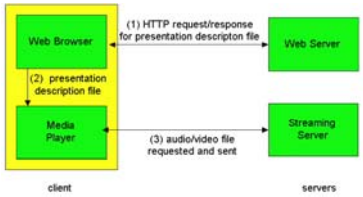
audio, video not streamed:
- no, "pipelining," long delays until playout!

Internet multimedia: streaming approach



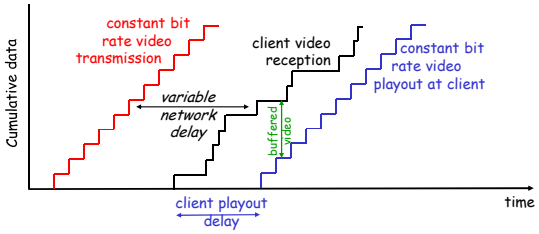
- browser GETs **metafile**
- browser launches player, passing metafile
- player contacts server
- server streams audio/video to player

Streaming from a streaming server



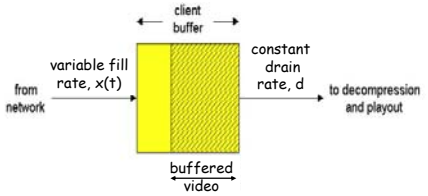
- This architecture allows for non-HTTP protocol between server and media player
- Can also use UDP instead of TCP.

Streaming Multimedia: Client Buffering



- Client-side buffering, playout delay compensate for network-added delay, delay jitter

Streaming Multimedia: Client Buffering



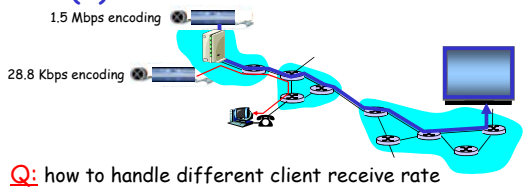
- Client-side buffering, playout delay compensate for network-added delay, delay jitter

Streaming Multimedia: UDP or TCP?

UDP
- server sends at rate appropriate for client (oblivious to network congestion !)
  - often send rate = encoding rate = constant rate
  - then, fill rate = constant rate - packet loss
- short playout delay (2-5 seconds) to compensate for network delay jitter
- error recover: time permitting

TCP
- send at maximum possible rate under TCP
- fill rate fluctuates due to TCP congestion control
- larger playout delay: smooth TCP delivery rate
- HTTP/TCP passes more easily through firewalls

Streaming Multimedia: client rate(s)



Q: how to handle different client receive rate capabilities?
- 28.8 Kbps dialup
- 100Mbps Ethernet

A: server stores, transmits multiple copies of video, encoded at different rates

User Control of Streaming Media: RTSP

HTTP
- Does not target multimedia content
- No commands for fast forward, etc.

RTSP: RFC 2326
- Client-server application layer protocol.
- For user to control display: rewind, fast forward, pause, resume, repositioning, etc…

What it doesn't do:
- does not define how audio/video is encapsulated for streaming over network
- does not restrict how streamed media is transported; it can be transported over UDP or TCP
- does not specify how the media player buffers audio/video

RTSP: out of band control

FTP uses an "out-of-band" control channel:
- A file is transferred over one TCP connection.
- Control information (directory changes, file deletion, file renaming, etc.) is sent over a separate TCP connection.
- The "out-of-band" and "in-band" channels use different port numbers.

RTSP messages are also sent out-of-band:
- RTSP control messages use different port numbers than the media stream: out-of-band.
  - Port 554
- The media stream is considered "in-band".

RTSP Example

Scenario:
- metafile communicated to web browser
- browser launches player
- player sets up an RTSP control connection, data connection to streaming server

Metafile Example

```
<title>Twister</title>
<session>
    <group language=en lipsync>
        <switch>
            <track type=audio
                e="PCMU/8000/1"
                src = "rtsp://audio.example.com/twister/audio.en/lofi">
            <track type=audio
                e="DVI4/16000/2" pt="90 DVI4/8000/1"
                src="rtsp://audio.example.com/twister/audio.en/hifi">
        </switch>
        <track type="video/jpeg"
                src="rtsp://video.example.com/twister/video">
    </group>
</session>
```