CS 422 (Park) **Midterm** **Mar. 21 (Wed.), 2007**

Remarks: Keep the answers compact, yet precise and to-the-point. Long-winded answers that do not address the key points are of limited value. Binary answers that give little indication of understanding are no good either. Time is not meant to be plentiful. Make sure not to get bogged down on a single problem.

PROBLEM 1 (48 pts)

(a) Suppose ARQ, in particular, sliding window, is used to transmit packets reliably over a network. What are the pros/cons of using positive ACK versus negative ACK? If the network is a simple point-to-point link comprised of optical fiber, which method is likely to be more suited? What if the link is wireless? What is good about using cumulative positive ACK?

(b) Suppose in a new wireless LAN standard, call it 802.11-purdue, a single carrier wave at 1 GHz frequency is used to transmit bits. Assume that one bit is carried by one full cycle (i.e., period) of a sine wave, with 4 amplitude levels to denote 00, 01, 10, and 11. If TDM is used to share bandwidth among 10 users (i.e., channels), each time slot being 10 bits long, what is the throughput (bps) achieved per user? Does the per user throughput depend on the size of the time slot? Throughput may be increased 100-fold by using the 100 GHz carrier frequency. Assuming that equipment cost does not change, why would this technology not be suited for wireless indoor communication?

(c) Suppose that stop-and-wait is used to reliably send packets from Purdue to Oxford in England with round-trip time (RTT) around 100 msec. If the packet size is 1000 bytes, what is the reliable throughput (bps) of the stop-and-wait protocol assuming no packets are dropped? How much time will it take to transfer a 100,000 byte file? The real throughput of stop-and-wait will be lower due to the need to allocate some of the 1000 bytes for the packet header such as sender and receiver addresses. In the case of stop-and-wait, what must be part of the packet header so that it functions correctly when packet drops do occur? How many bits does this header field take up? The physical bandwidth (bps) of the wires connecting Purdue to Oxford have not been considered when computing stop-and-wait's throughput. When is ignoring the impact of bandwidth a big deal, when is it not?

(d) One of the differences between Ethernet and wireless LAN (sometimes called "wireless Ethernet") is that in the former CD (collision detection) is employed whereas in the latter it is not. Why is this the case? Ethernet may be viewed as employing stop-and-wait with negative ACK. In what sense is this correct?

PROBLEM 2 (32 pts)

(a) In free or open space, it is assumed that signal strength decreases rapidly (i.e., quadratically) with distance. To provide wireless coverage over a given area (e.g., Purdue campus), one may deploy a few high-power base stations (APs in the case of WLANs) or many low-power base stations. What are the pros/cons of the two deployment strategies? To affect bandwidth sharing among multiple wireless devices associated with the same base station, we may use contention-free protocols such as TDM, FDM, TDMA, and CDMA, or contention-based protocols such as CSMA. What are the advantages of employing CSMA? What are its disadvantages? Describe a scenario where CSMA is most suited, and one where it is not.

(b) What is a switched Ethernet and why is it fundamentally different from a classical (i.e., bus-based) Ethernet? Why is CSMA/CD still used in switched Ethernet? In what setting is CSMA/CD turned off? Does it make sense to speak of a "switched wireless LAN" with all the performance benefits of a switched Ethernet? In the U.S., an infrastructure mode WLAN composed of a single AP (access point) and multiple wireless laptops associated with the AP—collectively forming a basic service set or BSS—uses 11 channels (i.e., carrier frequencies) in the 2.4 GHz range. Does this mean that infrastructure mode WLAN uses FDM to share bandwidth among the laptops in a single BSS? We know for a fact that CSMA is the protocol used in WLANs to mediate bandwidth, not FDM. Explain what's going on.

PROBLEM 3 (20 pts)

Suppose you are setting up an Internet voice service company, call it Epyks, that is to compete against Skype. Your main selling point is providing super-CD quality voice, capturing audio frequencies in the range 0–30,000 Hz. Describe how Nyquist's sampling criterion enters into play and how Shannon's compression (lower) bound enters into the picture after voice has been digitized. If 8 bits are used to represent the magnitude of a voice sample, what is the bit rate of an uncompressed voice session? After applying Shannon compression, what will determine how much compression is achieved? In the digitized voice context, what corresponds to "symbols" in the Shannon compression framework? In the worst case where there is no compression gain, will your new high-quality voice service be useful for home users with cable broadband access (say bandwidth 1-10 Mbps)? What about dial-up customers with 56 Kbps access speeds? Or your 2.5G cell phone with 256,000 Kbps data rate service? How many simultaneous Epyks voice calls can be supported over Purdue's 1 Gbps Internet link that goes through Indy? Note that the above calculations ignore packet header overhead which can significantly increase the final bit rate. What are some header fields that must be included in a voice application, what are common header fields that need not be? Keep in mind that humans perform their own ARQ by inquiring "can you repeat what you just said?" during times of bad reception.

BONUS PROBLEM (12 pts)

The ad hoc (also called peer-to-peer) mode of 802.11 WLANs is used in military settings to provide wireless connectivity among a group of soldiers without requiring an access point through which all communication is forwarded. What are some commercial contexts where ad hoc WLANs might be useful? From a common sense and/or technical perspective, what may be barriers to wide-spread adoption of ad hoc wireless LANs?