indicating the new IP address. Also, if Bob remains at the same device for an extended period of time, the device will send refresh register messages, indicating that the most recently sent IP address is still valid. (In the example above, refresh messages need to be sent every 3600 seconds to maintain the address at the registrar server.) It is worth noting that the registrar is analogous to a DNS authoritative name server: The DNS server translates fixed host names to fixed IP addresses; the SIP registrar translates fixed human identifiers (for example, bob@domain.com) to dynamic IP addresses. Often SIP registrars and SIP proxies are run on the same host. Now let's examine how Alice's SIP proxy server obtains Bob's current IP address. From the preceding discussion we see that the proxy server simply needs to forward Alice's INVITE message to Bob's registrar/proxy. The registrar/proxy could then forward the message to Bob's current SIP device. Finally, Bob, having now received Alice's INVITE message, could send an SIP response to Alice. As an example, consider Figure 9.10, in which jim@umass.edu, currently working on 217.123.56.89, wants to initiate a Voice-over-IP (VoIP) session with keith@upenn.edu, currently working on 197.87.54.21. The following steps are taken: Figure 9.10 Session initiation, involving SIP proxies and registrars (1) Jim sends an INVITE message to the umass SIP proxy. (2) The proxy does a DNS lookup on the SIP registrar upenn.edu (not shown in diagram) and then forwards the message to the registrar server. (3) Because keith@upenn.edu is no longer registered at the upenn registrar, the upenn registrar sends a redirect response, indicating that it should try keith@nyu.edu. (4) The umass proxy sends an INVITE message to the NYU SIP registrar. (5) The NYU registrar knows the IP address of keith@upenn.edu and forwards the INVITE message to the host 197.87.54.21, which is running Keith's SIP client. (6–8) An SIP response is sent back through registrars/proxies to the SIP client on 217.123.56.89. (9) Media is sent directly between the two clients. (There is also an SIP acknowledgment message, which is not shown.) Our discussion of SIP has focused on call initiation for voice calls. SIP, being a signaling protocol for initiating and ending calls in general, can be used for video conference calls as well as for text-based sessions. In fact, SIP has become a fundamental component in many instant messaging applications. Readers desiring to learn more about SIP are encouraged to visit Henning Schulzrinne's SIP Web site [Schulzrinne-SIP 2016]. In particular, on this site you will find open source software for SIP clients and servers [SIP Software 2016]. 9.5 Network Support for Multimedia In Sections 9.2 through 9.4, we learned how application-level mechanisms such as client buffering, prefetching, adapting media quality to available bandwidth, adaptive playout, and loss mitigation techniques can be used by multimedia applications to improve a multimedia application's performance. We also learned how content distribution networks and P2P overlay networks can be used to provide a system-level approach for delivering multimedia content. These techniques and approaches are all designed to be used in today's best-effort Internet. Indeed, they are in use today precisely because the Internet provides only a single, best-effort class of service. But as designers of computer networks, we can't help but ask whether the network (rather than the applications or application-level infrastructure alone) might provide mechanisms to support multimedia content delivery. As we'll see shortly, the answer is, of course, "yes"! But we'll also see that a number of these new network-level mechanisms have yet to be widely deployed. This may be due to their complexity and to the fact that application-level techniques together with best-effort service and properly dimensioned network resources (for example, bandwidth) can indeed provide a "good-enough" (even if not-always-perfect) end-to-end multimedia delivery service. Table 9.4 summarizes three broad approaches towards providing network-level support for multimedia applications. Making the best of best-effort service. The application-level mechanisms and infrastructure that we studied in Sections 9.2 through 9.4 can be successfully used in a well-dimensioned network where packet loss and excessive end-to-end delay rarely occur. When demand increases are forecasted, the ISPs deploy additional bandwidth and switching capacity to continue to ensure satisfactory delay and packet-loss performance [Huang 2005]. We'll discuss such network dimensioning further in Section 9.5.1. Differentiated service. Since the early days of the Internet, it's been envisioned that different types of traffic (for example, as indicated in the Type-of-Service field in the IP4v packet header) could be provided with different classes of service, rather than a single one-size-fits-all best-effort service. With differentiated service, one type of traffic might be given strict priority over another class of traffic when both types of traffic are queued at a router. For example, packets belonging to a realtime conversational application might be given priority over other packets due to their stringent delay constraints. Introducing differentiated service into the network will require new mechanisms for packet marking (indicating a packet's class of service), packet scheduling, and more. We'll cover differentiated service, and new network mechanisms needed to implement this service, in Sections 9.5.2 and 9.5.3. Table 9.4 Three network-level approaches to supporting multimedia applications Approach Granularity Guarantee Mechanisms Complexity Deployment to date Making the best of besteffort service all traffic treated equally none, or soft application-layer support, CDNs, overlays, networklevel resource provisioning minimal everywhere Differentiated service different classes of traffic treated differently none, or soft packet marking, policing, scheduling medium some Perconnection Quality-ofService (QoS) Guarantees each sourcedestination flows treated differently soft or hard, once flow is admitted packet marking, policing, scheduling; call admission and signaling light little Per-connection Quality-of-Service (QoS) Guarantees. With per-connection QoS guarantees, each instance of an application explicitly reserves end-to-end bandwidth and thus has a guaranteed end-to-end performance. A hard guarantee means the application will receive its requested quality of service (QoS) with certainty. A soft guarantee means the application will receive its requested quality of service with high probability. For example, if a user wants to make a VoIP call from Host A to Host B, the user's VoIP application reserves bandwidth explicitly in each link along a route between the two hosts. But permitting applications to make reservations and requiring the network to honor the reservations requires some big changes. First, we need a protocol that, on behalf of the applications, reserves link bandwidth on the paths from the senders to their receivers. Second, we'll need new scheduling policies in the router queues so that per-connection bandwidth reservations can be honored. Finally, in order to make a reservation, the applications must give the network a description of the traffic that they intend to send into the network and the network will need to police each application's traffic to make sure that it abides by that description. These mechanisms, when combined, require new and complex software in hosts and routers. Because per-connection QoS guaranteed service has not seen significant deployment, we'll cover these mechanisms only briefly in Section 9.5.4. 9.5.1 Dimensioning Best-Effort Networks Fundamentally, the difficulty in supporting multimedia applications arises from their stringent performance requirements—low end-to-end packet delay, delay jitter, and loss—and the fact that packet delay, delay jitter, and loss occur whenever the network becomes congested. A first approach to improving the quality of multimedia applications—an approach that can often be used to solve just about any problem where resources are constrained—is simply to "throw money at the problem" and thus simply avoid resource contention. In the case of networked multimedia, this means providing enough link capacity throughout the network so that network congestion, and its consequent packet delay and loss, never (or only very rarely) occurs. With enough link capacity, packets could zip through today's Internet without queuing delay or loss. From many perspectives this is an ideal situation—multimedia applications would perform perfectly, users would be happy, and this could all be achieved with no changes to Internet's best-effort architecture. The question, of course, is how much capacity is “enough” to achieve this nirvana, and whether the costs of providing “enough” bandwidth are practical from a business standpoint to the ISPs. The question of how much capacity to provide at network links in a given topology to achieve a given level of performance is often known as bandwidth provisioning. The even more complicated problem of how to design a network topology (where to place routers, how to interconnect routers with links, and what capacity to assign to links) to achieve a given level of end-to-end performance is a network design problem often referred to as network dimensioning. Both bandwidth provisioning and network dimensioning are complex topics, well beyond the scope of this textbook. We note here, however, that the following issues must be addressed in order to predict application-level performance between two network end points, and thus provision enough capacity to meet an application’s performance requirements. Models of traffic demand between network end points. Models may need to be specified at both the call level (for example, users “arriving” to the network and starting up end-to-end applications) and at the packet level (for example, packets being generated by ongoing applications). Note that workload may change over time. Well-defined performance requirements. For example, a performance requirement for supporting delay-sensitive traffic, such as a conversational multimedia application, might be that the probability that the end-to-end delay of the packet is greater than a maximum tolerable delay be less than some small value [Fraleigh 2003]. Models to predict end-to-end performance for a given workload model, and techniques to find a minimal cost bandwidth allocation that will result in all user requirements being met. Here, researchers are busy developing performance models that can quantify performance for a given workload, and optimization techniques to find minimal-cost bandwidth allocations meeting performance requirements. Given that today’s best-effort Internet could (from a technology standpoint) support multimedia traffic at an appropriate performance level if it were dimensioned to do so, the natural question is why today’s Internet doesn’t do so. The answers are primarily economic and organizational. From an economic standpoint, would users be willing to pay their ISPs enough for the ISPs to install sufficient bandwidth to support multimedia applications over a best-effort Internet? The organizational issues are perhaps even more daunting. Note that an end-to-end path between two multimedia end points will pass through the networks of multiple ISPs. From an organizational standpoint, would these ISPs be willing to cooperate (perhaps with revenue sharing) to ensure that the end-to-end path is properly dimensioned to support multimedia applications? For a perspective on these economic and organizational issues, see [Davies 2005]. For a perspective on provisioning tier-1 backbone networks to support delay-sensitive traffic, see [Fraleigh 2003]. 9.5.2 Providing Multiple Classes of Service Perhaps the simplest enhancement to the one-size-fits-all best-effort service in today’s Internet is to divide traffic into classes, and provide different levels of service to these different classes of traffic. For example, an ISP might well want to provide a higher class of service to delay-sensitive Voice-over-IP or teleconferencing traffic (and charge more for this service!) than to elastic traffic such as e-mail or HTTP. Alternatively, an ISP may simply want to provide a higher quality of service to customers willing to pay more for this improved service. A number of residential wired-access ISPs and cellular wireless-access ISPs have adopted such tiered levels of service—with platinum-service subscribers receiving better performance than gold- or silver-service subscribers. We’re all familiar with different classes of service from our everyday lives—first-class airline passengers get better service than business-class passengers, who in turn get better service than those of us who fly economy class; VIPs are provided immediate entry to events while everyone else waits in line; elders are revered in some countries and provided seats of honor and the finest food at a table. It’s important to note that such differential service is provided among aggregates of traffic, that is, among classes of traffic, not among individual connections. For example, all first-class passengers are handled the same (with no first-class passenger receiving any better treatment than any other first-class passenger), just as all VoIP packets would receive the same treatment within the network, independent of the particular end-to-end connection to which they belong. As we will see, by dealing with a small number of traffic aggregates, rather than a large number of individual connections, the new network mechanisms required to provide better-than-best service can be kept relatively simple. The early Internet designers clearly had this notion of multiple classes of service in mind. Recall the type-of-service (ToS) field in the IPv4 header discussed in Chapter 4. IEN123 [ISI 1979] describes the ToS field also present in an ancestor of the IPv4 datagram as follows: “The Type of Service [field] provides an indication of the abstract parameters of the quality of service desired. These parameters are to be used to guide the selection of the actual service parameters when transmitting a datagram through a particular network. Several networks offer service precedence, which somehow treats high precedence traffic as more important that other traffic.” More than four decades ago, the vision of providing different levels of service to different classes of traffic was clear! However, it’s taken us an equally long period of time to realize this vision. Motivating Scenarios Let’s begin our discussion of network mechanisms for providing multiple classes of service with a few motivating scenarios. Figure 9.11 shows a simple network scenario in which two application packet flows originate on Hosts H1 and H2 on one LAN and are destined for Hosts H3 and H4 on another LAN. The routers on the two LANs are connected by a 1.5 Mbps link. Let’s assume the LAN speeds are significantly higher than 1.5 Mbps, and focus on the output queue of router R1; it is here that packet delay and packet loss will occur if the aggregate sending rate of H1 and H2 exceeds 1.5 Mbps. Let’s further suppose that a 1 Mbps audio application (for example, a CD-quality audio call) shares the Figure 9.11 Competing audio and HTTP applications 1.5 Mbps link between R1 and R2 with an HTTP Web-browsing application that is downloading a Web page from H2 to H4. In the best-effort Internet, the audio and HTTP packets are mixed in the output queue at R1 and (typically) transmitted in a first-in-first-out (FIFO) order. In this scenario, a burst of packets from the Web server could potentially fill up the queue, causing IP audio packets to be excessively delayed or lost due to buffer overflow at R1. How should we solve this potential problem? Given that the HTTP Webbrowsing application does not have time constraints, our intuition might be to give strict priority to audio packets at R1. Under a strict priority scheduling discipline, an audio packet in the R1 output buffer would always be transmitted before any HTTP packet in the R1 output buffer. The link from R1 to R2 would look like a dedicated link of 1.5 Mbps to the audio traffic, with HTTP traffic using the R1-to-R2 link only when no audio traffic is queued. In order for R1 to distinguish between the audio and HTTP packets in its queue, each packet must be marked as belonging to one of these two classes of traffic. This was the original goal of the type-of-service (ToS) field in IPv4. As obvious as this might seem, this then is our first insight into mechanisms needed to provide multiple classes of traffic: Insight 1: Packet marking allows a router to distinguish among packets belonging to different classes of traffic. Note that although our example considers a competing multimedia and elastic flow, the same insight applies to the case that platinum, gold, and silver classes of service are implemented—a packetmarking mechanism is still needed to indicate that class of service to which a packet belongs. Now suppose that the router is configured to give priority to packets marked as belonging to the 1 Mbps audio application. Since the outgoing link speed is 1.5 Mbps, even though the HTTP packets receive lower priority, they can still, on average, receive 0.5 Mbps of transmission service. But what happens if the audio application starts sending packets at a rate of 1.5 Mbps or higher (either maliciously or due to an error in the application)? In this case, the HTTP packets will starve, that is, they will not receive any service on the R1-to-R2 link. Similar problems would occur if multiple applications (for example, multiple audio calls), all with the same class of service as the audio application, were sharing the link's bandwidth; they too could collectively starve the FTP session. Ideally, one wants a degree of isolation among classes of traffic so that one class of traffic can be protected from the other. This protection could be implemented at different places in the network—at each and every router, at first entry to the network, or at inter-domain network boundaries. This then is our second insight: Insight 2: It is desirable to provide a degree of traffic isolation among classes so that one class is not adversely affected by another class of traffic that misbehaves. We'll examine several specific mechanisms for providing such isolation among traffic classes. We note here that two broad approaches can be taken. First, it is possible to perform traffic policing, as shown in Figure 9.12. If a traffic class or flow must meet certain criteria (for example, that the audio flow not exceed a peak rate of 1 Mbps), then a policing mechanism can be put into place to ensure that these criteria are indeed observed. If the policed application misbehaves, the policing mechanism will take some action (for example, drop or delay packets that are in violation of the criteria) so that the traffic actually entering the network conforms to the criteria. The leaky bucket mechanism that we'll examine shortly is perhaps the most widely used policing mechanism. In Figure 9.12, the packet classification and marking mechanism (Insight 1) and the policing mechanism (Insight 2) are both implemented together at the network's edge, either in the end system or at an edge router. A complementary approach for providing isolation among traffic classes is for the link-level packetscheduling mechanism to explicitly allocate a fixed amount of link bandwidth to each class. For example, the audio class could be allocated 1 Mbps at R1, and the HTTP class could be allocated 0.5 Mbps. In this case, the audio and Figure 9.12 Policing (and marking) the audio and HTTP traffic classes Figure 9.13 Logical isolation of audio and HTTP traffic classes HTTP flows see a logical link with capacity 1.0 and 0.5 Mbps, respectively, as shown in Figure 9.13. With strict enforcement of the link-level allocation of bandwidth, a class can use only the amount of bandwidth that has been allocated; in particular, it cannot utilize bandwidth that is not currently being used by others. For example, if the audio flow goes silent (for example, if the speaker pauses and generates no audio packets), the HTTP flow would still not be able to transmit more than 0.5 Mbps over the R1-to-R2 link, even though the audio flow's 1 Mbps bandwidth allocation is not being used at that moment. Since bandwidth is a "use-it-or-lose-it" resource, there is no reason to prevent HTTP traffic from using bandwidth not used by the audio traffic. We'd like to use bandwidth as efficiently as possible, never wasting it when it could be otherwise used. This gives rise to our third insight: Insight 3: While providing isolation among classes or flows, it is desirable to use resources (for example, link bandwidth and buffers) as efficiently as possible. Recall from our discussion in Sections 1.3 and 4.2 that packets belonging to various network flows are multiplexed and queued for transmission at the output buffers associated with a link. The manner in which queued packets are selected for transmission on the link is known as the link-scheduling discipline, and was discussed in detail in Section 4.2. Recall that in Section 4.2 three link-scheduling disciplines were discussed, namely, FIFO, priority queuing, and Weighted Fair Queuing (WFQ). We'll see soon see that WFQ will play a particularly important role for isolating the traffic classes. The Leaky Bucket One of our earlier insights was that policing, the regulation of the rate at which a class or flow (we will assume the unit of policing is a flow in our discussion below) is allowed to inject packets into the network, is an important QoS mechanism. But what aspects of a flow's packet rate should be policed? We can identify three important policing criteria, each differing from the other according to the time scale over which the packet flow is policed: Average rate. The network may wish to limit the long-term average rate (packets per time interval) at which a flow's packets can be sent into the network. A crucial issue here is the interval of time over which the average rate will be policed. A flow whose average rate is limited to 100 packets per second is more constrained than a source that is limited to 6,000 packets per minute, even though both have the same average rate over a long enough interval of time. For example, the latter constraint would allow a flow to send 1,000 packets in a given second-long interval of time, while the former constraint would disallow this sending behavior. Peak rate. While the average-rate constraint limits the amount of traffic that can be sent into the network over a relatively long period of time, a peak-rate constraint limits the maximum number of packets that can be sent over a shorter period of time. Using our example above, the network may police a flow at an average rate of 6,000 packets per minute, while limiting the flow's peak rate to 1,500 packets per second. Burst size. The network may also wish to limit the maximum number of packets (the "burst" of packets) that can be sent into the network over an extremely short interval of time. In the limit, as the interval length approaches zero, the burst size limits the number of packets that can be instantaneously sent into the network. Even though it is physically impossible to instantaneously send multiple packets into the network (after all, every link has a physical transmission rate that cannot be exceeded!), the abstraction of a maximum burst size is a useful one. The leaky bucket mechanism is an abstraction that can be used to characterize these policing limits. As shown in Figure 9.14, a leaky bucket consists of a bucket that can hold up to b tokens. Tokens are added to this bucket as follows. New tokens, which may potentially be added to the bucket, are always being generated at a rate of r tokens per second. (We assume here for simplicity that the unit of time is a second.) If the bucket is filled with less than b tokens when a token is generated, the newly generated token is added to the bucket; otherwise the newly generated token is ignored, and the token bucket remains full with b tokens. Let us now consider how the leaky bucket can be used to police a packet flow. Suppose that before a packet is transmitted into the network, it must first remove a token from the token bucket. If the token bucket is empty, the packet must wait for Figure 9.14 The leaky bucket policer a token. (An alternative is for the packet to be dropped, although we will not consider that option here.) Let us now consider how this behavior polices a traffic flow. Because there can be at most b tokens in the bucket, the maximum burst size for a leaky-bucket-policed flow is b packets. Furthermore, because the token generation rate is r, the maximum number of packets that can enter the network of any interval of time of length t is Thus, the token-generation rate, r, serves to limit the long-term average rate at which packets can enter the network. It is also possible to use leaky buckets (specifically, two leaky buckets in series) to police a flow's peak rate in addition to the long-term average rate; see the homework problems at the end of this chapter. Leaky Bucket Weighted Fair Queuing Provable Maximum Delay in a Queue Let's close our discussion on policing by showing how the leaky bucket and WFQ can be combined to provide a bound on the delay through a router's queue. (Readers who have forgotten about WFQ are encouraged to review WFQ, which is covered in Section 4.2.) Let's consider a router's output link that multiplexes n flows, each policed by a leaky bucket with parameters b and using WFQ scheduling. We use the term flow here loosely to refer to the set of packets that are not distinguished from each other by the scheduler. In practice, a flow might be comprised of traffic from a single end-toend connection or a collection of many such connections, see Figure 9.15. Recall from our discussion of WFQ that each flow, i, is guaranteed to receive a share of the link bandwidth equal to at least where R is the transmission rt+b. + = i ri,i=1,...,n, R·wi/(∑ wj), Figure 9.15 n multiplexed leaky bucket flows with WFQ scheduling rate of the link in packets/sec. What then is the maximum delay that a packet will experience while waiting for service in the WFQ (that is, after passing through the leaky bucket)? Let us focus on flow 1. Suppose that flow 1's token bucket is initially full. A burst of b packets then arrives to the leaky bucket policer for flow 1. These packets remove all of the tokens (without wait) from the leaky bucket and then join the WFQ waiting area for flow 1. Since these b packets are served at a rate of at least packet/sec, the last of these packets will then have a maximum delay, d , until its transmission is completed, where The rationale behind this formula is that if there are b packets in the queue and packets are being serviced (removed) from the queue at a rate of at least packets per second, then the amount of time until the last bit of the last packet is transmitted cannot be more than . A homework problem asks you to prove that as long as then d is indeed the maximum delay that any packet in flow 1 will ever experience in the WFQ queue. 9.5.3 Diffserv Having seen the motivation, insights, and specific mechanisms for providing multiple classes of service, let's wrap up our study of approaches toward proving multiple classes of service with an example—the Internet Diffserv architecture [RFC 2475; Kilkki 1999]. Diffserv provides service differentiation—that is, the ability to handle different classes of traffic in different ways within the Internet in a scalable manner. 1 1 $R \cdot w_i/(\sum w_j)$ max $d_{max}=b_1 R \cdot w_1/\sum w_j$ 1 $R \cdot w_1/(\sum w_j)$ $b_1/(R \cdot w_1/(\sum w_j))$ $r_1 r$. $t=t_f$ $t=0$ $t=T$. $Q=0$, $Q>0$ $HT/2 \geq Q$. $H>2r$ $Q=HT/2$. $H>2r$. $H>2r$. $t=t_f$ rE. i $u=0.1$. r_1-t_1 r_2-t_2 r_4-t_4, r_3-t_3, r_2-t_2, r_1-t_1. i i n $d_1=r_1-t_1$. a. Suppose that we would like for all n. Give a recursive formula for d in terms of and t . b. Describe why for Internet telephony, the delay estimate described in Section 9.3 is more appropriate than the delay estimate outlined in part (a). P10. Compare the procedure described in Section 9.3 for estimating average delay with the procedure in Section 3.5 for estimating round-trip time. What do the procedures have in common? How are they different? P11. Consider the figure below (which is similar to Figure 9.3). A sender begins sending packetized audio periodically at The first packet arrives at the receiver at a. What are the delays (from sender to receiver, ignoring any playout delays) of packets 2 through 8? Note that each vertical and horizontal line segment in the figure has a length of 1, 2, or 3 time units. b. If audio playout begins as soon as the first packet arrives at the receiver at which of the first eight packets sent will not arrive in time for playout? c. If audio playout begins at which of the first eight packets sent will not arrive in time for playout? d. What is the minimum playout delay at the receiver that results in all of the first eight packets arriving in time for their playout? P12. Consider again the figure in P11, showing packet audio transmission and reception times. a. Compute the estimated delay for packets 2 through 8, using the formula for d from Section 9.3.2 . Use a value of . $d_n=(r_1-t_1+r_2-t_2+\cdots+r_n-t_n)/n$ n d_{n-1}, r_n, n $t=1$. $t=8$. $t=8$, $t=9$, i $u=0.1$ b. Compute the estimated deviation of the delay from the estimated average for packets 2 through 8, using the formula for v from Section 9.3.2 . Use a value of . P13. Recall the two FEC schemes for VoIP described in Section 9.3 . Suppose the first scheme generates a redundant chunk for every four original chunks. Suppose the second scheme uses a low-bit rate encoding whose transmission rate is 25 percent of the transmission rate of the nominal stream. a. How much additional bandwidth does each scheme require? How much playback delay does each scheme add? b. How do the two schemes perform if the first packet is lost in every group of five packets? Which scheme will have better audio quality? c. How do the two schemes perform if the first packet is lost in every group of two packets? Which scheme will have better audio quality? P14. a. Consider an audio conference call in Skype with participants. Suppose each participant generates a constant stream of rate r bps. How many bits per second will the call initiator need to send? How many bits per second will each of the other participants need to send? What is the total send rate, aggregated over all participants? b. Repeat part (a) for a Skype video conference call using a central server. c. Repeat part (b), but now for when each peer sends a copy of its video stream to each of the other peers. P15. a. Suppose we send into the Internet two IP datagrams, each carrying a different UDP segment. The first datagram has source IP address A1, destination IP address B, source port P1, and destination port T. The second datagram has source IP address A2, destination IP address B, source port P2, and destination port T. Suppose that A1 is different from A2 and that P1 is different from P2. Assuming that both datagrams reach their final destination, will the two UDP datagrams be received by the same socket? Why or why not? b. Suppose Alice, Bob, and Claire want to have an audio conference call using SIP and RTP. For Alice to send and receive RTP packets to and from Bob and Claire, is only one UDP socket sufficient (in addition to the socket needed for the SIP messages)? If yes, then how does Alice's SIP client distinguish between the RTP packets received from Bob and Claire? P16. True or false: a. If stored video is streamed directly from a Web server to a media player, then the application is using TCP as the underlying transport protocol. i u=0.1 N>2 N−1 N−1 b. When using RTP, it is possible for a sender to change encoding in the middle of a session. c. All applications that use RTP must use port 87. d. If an RTP session has a separate audio and video stream for each sender, then the audio and video streams use the same SSRC. e. In differentiated services, while per-hop behavior defines differences in performance among classes, it does not mandate any particular mechanism for achieving these performances. f. Suppose Alice wants to establish an SIP session with Bob. In her INVITE message she includes the line: m=audio 48753 RTP/AVP 3 (AVP 3 denotes GSM audio). Alice has therefore indicated in this message that she wishes to send GSM audio. g. Referring to the preceding statement, Alice has indicated in her INVITE message that she will send audio to port 48753. h. SIP messages are typically sent between SIP entities using a default SIP port number. i. In order to maintain registration, SIP clients must periodically send REGISTER messages. j. SIP mandates that all SIP clients support G.711 audio encoding. P17. Consider the figure below, which shows a leaky bucket policer being fed by a stream of packets. The token buffer can hold at most two tokens, and is initially full at New tokens arrive at a rate of one token per slot. The output link speed is such that if two packets obtain tokens at the beginning of a time slot, they can both go to the output link in the same slot. The timing details of the system are as follows: A. Packets (if any) arrive at the beginning of the slot. Thus in the figure, packets 1, 2, and 3 arrive in slot 0. If there are already packets in the queue, then the arriving packets join the end of the queue. Packets proceed towards the front of the queue in a FIFO manner. B. After the arrivals have been added to the queue, if there are any queued packets, one or two of those packets (depending on the number of available tokens) will each remove a token from the token buffer and go to the output link during that slot. Thus, packets 1 and t=0. Programming Assignment In this lab, you will implement a streaming video server and client. The client will use the real-time streaming protocol (RTSP) to control the actions of the server. The server will use the real-time protocol (RTP) to packetize the video for transport over UDP. You will be given Python code that partially implements RTSP and RTP at the client and server. Your job will be to complete both the client and server code. When you are finished, you will have created a client-server application that does the following: 2 each remove a token from the buffer (since there are initially two tokens) and go to the output link during slot 0. C. A new token is added to the token buffer if it is not full, since the token generation rate is r = 1 token/slot. D. Time then advances to the next time slot, and these steps repeat. Answer the following questions: a. For each time slot, identify the packets that are in the queue and the number of tokens in the bucket, immediately after the arrivals have been processed (step 1 above) but before any of the packets have passed through the queue and removed a token. Thus, for the time slot in the example above, packets 1, 2, and 3 are in the queue, and there are two tokens in the buffer. b. For each time slot indicate which packets appear on the output after the token(s) have been removed from the queue. Thus, for the time slot in the example above, packets 1 and 2 appear on the output link from the leaky buffer during slot 0. P18. Repeat P17 but assume that Assume again that the bucket is initially full. P19. Consider P18 and suppose now that and that as before. Will your answer to the question above change? P20. Consider the leaky bucket policer that polices the average rate and burst size of a packet flow. We now want to police the peak rate, p, as well. Show how the output of this leaky bucket policer can be fed into a second leaky bucket policer so that the two leaky buckets in series police the average rate, peak rate, and burst size. Be sure to give the bucket size and token generation rate for the second policer. P21. A packet flow is said to conform to a leaky bucket specification (r, b) with burst size b and average rate r if the number of packets that arrive to the leaky bucket is less than packets in every interval of time of length t for all t. Will a packet flow that conforms to a leaky bucket specification (r, b) ever have to wait at a leaky bucket policer with parameters r and b? Justify your answer. P22. Show that as long as then d is indeed the maximum delay that any packet in flow 1 will ever experience in the WFQ queue. t=0 t=0 r=2. r=3 b=2 rt+b r1

Programming Assignment In this lab, you will implement a streaming video server and client. The client will use the real-time streaming protocol (RTSP) to control the actions of the server. The server will use the real-time protocol (RTP) to packetize the video for transport over UDP. You will be given Python code that partially implements RTSP and RTP at the client and server. Your job will be to complete both the client and server code. When you are finished, you will have created a client-server application that does the following:

The client sends SETUP, PLAY, PAUSE, and TEARDOWN RTSP commands, and the server responds to the commands. When the server is in the playing state, it periodically grabs a stored JPEG frame, packetizes the frame with RTP, and sends the RTP packet into a UDP socket. The client receives the RTP packets, removes the JPEG frames, decompresses the frames, and renders the frames on the client's monitor. The code you will be given implements the RTSP protocol in the server and the RTP depacketization in the client. The code also takes care of displaying the transmitted video. You will need to implement RTSP in the client and RTP server. This programming assignment will significantly enhance the student's understanding of RTP, RTSP, and streaming video. It is highly recommended. The assignment also suggests a number of optional exercises, including implementing the RTSP DESCRIBE command at both client and server. You can find full details of the assignment, as well as an overview of the RTSP protocol, at the Web site www.pearsonhighered.com/cs-resources. AN INTERVIEW WITH . . . Henning Schulzrinne Henning Schulzrinne is a professor, chair of the Department of Computer Science, and head of the Internet Real-Time Laboratory at Columbia University. He is the co-author of RTP, RTSP, SIP, and GIST—key protocols for audio and video communications over the Internet. Henning received his BS in electrical and industrial engineering at TU Darmstadt in Germany, his MS in electrical and computer engineering at the University of Cincinnati, and his PhD in electrical engineering at the University of Massachusetts, Amherst. What made you decide to specialize in multimedia networking? This happened almost by accident. As a PhD student, I got involved with DARTnet, an experimental network spanning the United States with T1 lines. DARTnet was used as a proving ground for multicast and Internet real-time tools. That led me to write my first audio tool, NeVoT. Through some of the DARTnet participants, I became involved in the IETF, in the then-nascent Audio Video Transport working group. This group later ended up standardizing RTP. What was your first job in the computer industry? What did it entail? My first job in the computer industry was soldering together an Altair computer kit when I was a high school student in Livermore, California. Back in Germany, I started a little consulting company that devised an address management program for a travel agency—storing data on cassette tapes for our TRS-80 and using an IBM Selectric typewriter with a home-brew hardware interface as a printer. My first real job was with AT&T Bell Laboratories, developing a network emulator for constructing experimental networks in a lab environment. What are the goals of the Internet Real-Time Lab? Our goal is to provide components and building blocks for the

Internet as the single future communications infrastructure. This includes developing new protocols, such as GIST (for network-layer signaling) and LoST (for finding resources by location), or enhancing protocols that we have worked on earlier, such as SIP, through work on rich presence, peer-to-peer systems, next-generation emergency calling, and service creation tools. Recently, we have also looked extensively at wireless systems for VoIP, as 802.11b and 802.11n networks and maybe WiMax networks are likely to become important last-mile technologies for telephony. We are also trying to greatly improve the ability of users to diagnose faults in the complicated tangle of providers and equipment, using a peer-to-peer fault diagnosis system called DYSWIS (Do You See What I See). We try to do practically relevant work, by building prototypes and open source systems, by measuring performance of real systems, and by contributing to IETF standards. What is your vision for the future of multimedia networking? We are now in a transition phase; just a few years shy of when IP will be the universal platform for multimedia services, from IPTV to VoIP. We expect radio, telephone, and TV to be available even during snowstorms and earthquakes, so when the Internet takes over the role of these dedicated networks, users will expect the same level of reliability. We will have to learn to design network technologies for an ecosystem of competing carriers, service and content providers, serving lots of technically untrained users and defending them against a small, but destructive, set of malicious and criminal users. Changing protocols is becoming increasingly hard. They are also becoming more complex, as they need to take into account competing business interests, security, privacy, and the lack of transparency of networks caused by firewalls and network address translators. Since multimedia networking is becoming the foundation for almost all of consumer entertainment, there will be an emphasis on managing very large networks, at low cost. Users will expect ease of use, such as finding the same content on all of their devices. Why does SIP have a promising future? As the current wireless network upgrade to 3G networks proceeds, there is the hope of a single multimedia signaling mechanism spanning all types of networks, from cable modems, to corporate telephone networks and public wireless networks. Together with software radios, this will make it possible in the future that a single device can be used on a home network, as a cordless BlueTooth phone, in a corporate network via 802.11 and in the wide area via 3G networks. Even before we have such a single universal wireless device, the personal mobility mechanisms make it possible to hide the differences between networks. One identifier becomes the universal means of reaching a person, rather than remembering or passing around half a dozen technology- or location-specific telephone numbers. SIP also breaks apart the provision of voice (bit) transport from voice services. It now becomes technically possible to break apart the local telephone monopoly, where one company provides neutral bit transport, while others provide IP “dial tone” and the classical telephone services, such as gateways, call forwarding, and caller ID. Beyond multimedia signaling, SIP offers a new service that has been missing in the Internet: event notification. We have approximated such services with HTTP kludges and e-mail, but this was never very satisfactory. Since events are a common abstraction for distributed systems, this may simplify the construction of new services. Do you have any advice for students entering the networking field? Networking bridges disciplines. It draws from electrical engineering, all aspects of computer science, operations research, statistics, economics, and other disciplines. Thus, networking researchers have to be familiar with subjects well beyond protocols and routing algorithms. Given that networks are becoming such an important part of everyday life, students wanting to make a difference in the field should think of the new resource constraints in networks: human time and effort, rather than just bandwidth or storage. Work in networking research can be immensely satisfying since it is about allowing people to communicate and exchange ideas, one of the essentials of being human. The Internet has become the third major global infrastructure, next to the transportation system and energy distribution. Almost no part of the economy can work without high-performance networks, so there should be plenty of opportunities for the foreseeable future.

References A note on URLs. In the references below, we have provided URLs for Web pages, Web-only documents, and other material that has not been published in a conference or journal (when we have been able to locate a URL for such material). We have not provided URLs for conference and journal publications, as these documents can usually be located via a search engine, from the conference Web site (e.g., papers in all ACM SIGCOMM conferences and workshops can be located via http://www.acm.org/ sigcomm), or via a digital library subscription. While all URLs provided below were valid (and tested) in Jan. 2016, URLs can become out of date. Please consult the online version of this book (www.pearsonhighered .com/cs-resources) for an up-to-date bibliography. A note on Internet Request for Comments (RFCs): Copies of Internet RFCs are available at many sites. The RFC Editor of the Internet Society (the body that oversees the RFCs) maintains the site, http://www.rfc-editor.org. This site allows you to search for a specific RFC by title, number, or authors, and will show updates to any RFCs listed. Internet RFCs can be updated or obsoleted by later RFCs. Our favorite site for getting RFCs is the original source—http://www.rfc-editor.org. [3GPP 2016] Third Generation Partnership Project homepage, http://www.3gpp.org/ [Abramson 1970] N. Abramson, "The Aloha System—Another Alternative for Computer Communications," Proc. 1970 Fall Joint Computer Conference, AFIPS Conference, p. 37, 1970. [Abramson 1985] N. Abramson, "Development of the Alohanet," IEEE Transactions on Information Theory, Vol. IT-31, No. 3 (Mar. 1985), pp. 119–123. [Abramson 2009] N. Abramson, "The Alohanet—Surfing for Wireless Data," IEEE Communications Magazine, Vol. 47, No. 12, pp. 21–25. [Adhikari 2011a] V. K. Adhikari, S. Jain, Y. Chen, Z. L. Zhang, "Vivisecting YouTube: An Active Measurement Study," Technical Report, University of Minnesota, 2011. [Adhikari 2012] V. K. Adhikari, Y. Gao, F. Hao, M. Varvello, V. Hilt, M. Steiner, Z. L. Zhang, "Unreeling Netflix: Understanding and Improving Multi-CDN Movie Delivery," Technical Report, University of Minnesota, 2012. [Afanasyev 2010] A. Afanasyev, N. Tilley, P. Reiher, L. Kleinrock, "Host-to-Host Congestion Control for TCP," IEEE Communications Surveys & Tutorials, Vol. 12, No. 3, pp. 304–342. [Agarwal 2009] S. Agarwal, J. Lorch, "Matchmaking for Online Games and Other Latency-sensitive P2P Systems," Proc. 2009 ACM SIGCOMM. [Ager 2012] B. Ager, N. Chatzis, A. Feldmann, N. Sarrar, S. Uhlig, W. Willinger, "Anatomy of a Large European ISP," Sigcomm, 2012. [Ahn 1995] J. S. Ahn, P. B. Danzig, Z. Liu, and Y. Yan, "Experience with TCP Vegas: Emulation and Experiment," Proc. 1995 ACM SIGCOMM (Boston, MA, Aug. 1995), pp. 185–195. [Akamai 2016] Akamai homepage, http://www.akamai.com [Akella 2003] A. Akella, S. Seshan, A. Shaikh, "An Empirical Evaluation of Wide-Area Internet Bottlenecks," Proc. 2003 ACM Internet Measurement Conference (Miami, FL, Nov. 2003). [Akhshabi 2011] S. Akhshabi, A. C. Begen, C. Dovrolis, "An Experimental Evaluation of Rate-Adaptation Algorithms in Adaptive Streaming over HTTP," Proc. 2011 ACM Multimedia Systems Conf. [Akyildiz 2010] I. Akyildiz, D. Gutierrex-Estevez, E. Reyes, "The Evolution to 4G Cellular Systems, LTE Advanced," Physical Communication, Elsevier, 3 (2010), 217–244. [Albitz 1993] P. Albitz and C. Liu, DNS and BIND, O'Reilly & Associates, Petaluma, CA, 1993. [Al-Fares 2008] M. Al-Fares, A. Loukissas, A. Vahdat, "A Scalable, Commodity Data Center Network Architecture," Proc. 2008 ACM SIGCOMM. [Amazon 2014] J. Hamilton, "AWS: Innovation at Scale, YouTube video, https://www.youtube.com/watch?v=JIQETrFC_SQ [Anderson 1995] J. B. Andersen, T. S. Rappaport, S. Yoshida, "Propagation Measurements and Models for Wireless Communications Channels," IEEE Communications Magazine, (Jan. 1995),

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