audio file. To send a message from a source end system to a destination end system, the source breaks long messages into smaller chunks of data known as **packets**. Between source and destination, each packet travels through communication links and **packet switches** (for which there are two predominant types, **routers** and **link-layer switches**). Packets are transmitted over each communication link at a rate equal to the *full* transmission rate of the link. So, if a source end system or a packet switch is sending a packet of L bits over a link with transmission rate R bits/sec, then the time to transmit the packet is L/R seconds.

speed of light—which will be discussed in Section 1.4.) The source begins to transmit at time 0; at time L/R seconds, the source has transmitted the entire packet, and the entire packet has been received and stored at the router (since there is no propagation delay). At time L/R seconds, since the router has just received the entire packet, it can begin to transmit the packet onto the outbound link towards the destination; at time $2L/R$, the router has transmitted the entire packet, and the entire packet has been received by the destination. Thus, the total delay is $2L/R$. If the switch instead forwarded bits as soon as they arrive (without first receiving the entire packet), then the total delay would be L/R since bits are not held up at the router. But, as we will discuss in Section 1.4, routers need to receive, store, and

gested (so that queuing delays are negligible), the processing delay at each router and at the source host is d_{proc}, the transmission rate out of each router and out of the source host is R bits/sec, and the propagation on each link is d_{prop}. The nodal delays accumulate and give an end-to-end delay,

$$d_{end\text{-}end} = N\,(d_{proc} + d_{trans} + d_{prop}) \tag{1.2}$$

where, once again, $d_{trans} = L/R$, where L is the packet size. Note that Equation 1.2 is a generalization of Equation 1.1, which did not take into account processing and propagation delays. We leave it to you to generalize Equation 1.2 to the case of heterogeneous delays at the nodes and to the presence of an average queuing delay at each node.

- More specifically, each router has a forwarding table that maps destination addresses (or portions of the destination addresses) to that router's outbound links.
- In circuit-switched networks, the resources needed along a path (buffers, link transmission rate) to provide for communication between the end systems are reserved for the duration of the communication session between the end systems. In packet-switched networks, these resources are not reserved; a session's messages use the resources on demand, and as a consequence, may have to wait (that is, queue) for access to a communication link.
- Packet switching is often considered more efficient than circuit switching for data transmission. In circuit switching, dedicated bandwidth is reserved for each user, limiting simultaneous connections. In contrast, packet switching dynamically shares the link, accommodating varying user activity levels. This results in higher overall capacity utilization and faster data transmission, especially in scenarios with intermittent user activity or when a user generates a burst of data. Packet switching offers comparable performance to circuit switching with significantly more users and is simpler and cost-effective to implement.
- In a 1 Mbps shared link, circuit switching reserves fixed bandwidth for users, limiting simultaneous connections. With packet switching, dynamic sharing allows more users, and low probability of high activity ensures performance similar to circuit switching. Additionally, in a burst scenario, packet switching efficiently utilizes the full link rate, transmitting data faster compared to circuit switching's fixed time slots.
- The most important of these delays are the **nodal processing delay**, **queuing delay**, **transmission delay**, and **propagation delay**; together, these delays accumulate to give a **total nodal delay**.
- The time required to examine the packet's header and determine where to direct the packet is part of the **processing delay**. At the queue, the packet experiences a **queuing delay** as it waits to be transmitted onto the link. The **transmission delay** is L/R. This is the amount of time required to push (that is, transmit) all of the packet's bits into the link. Once a bit is pushed into the link, it needs to propagate to router B. The time required to propagate from the beginning of the link to router B is the **propagation delay**. The bit propagates at the propagation speed of the link. The propagation speed depends on the physical medium of the link (that is, fiber optics, twisted-pair copper wire, and so on)

Let's now consider the general case of sending one packet from source to destination over a path consisting of N links each of rate R (thus, there are N-1 routers between source and destination). Applying the same logic as above, we see that the end-to-end delay is:

$$d_{end\text{-}to\text{-}end} = N\frac{L}{R} \tag{1.1}$$

packets that the router is about to send into that link. The output buffers play a key role in packet switching. If an arriving packet needs to be transmitted onto a link but finds the link busy with the transmission of another packet, the arriving packet must wait in the output buffer. Thus, in addition to the store-and-forward delays, packets suffer output buffer **queuing delays**. These delays are variable and depend on the level of congestion in the network. Since the amount of buffer space is finite, an arriving packet may find that the buffer is completely full with other packets waiting for transmission. In this case, **packet loss** will occur—either the arriving packet or one of the already-queued packets will be dropped.

Why Modem?

• Signal in frequency domain

– Fourier Analysis: any periodic function can be constructed as the sum of a number of sines and consines

– Baseband DC (digital) signal has infinite harmonics

• Channel bandwidth: defined as the frequency between the highest and the lowest frequency that the channel can reliably transfer (without strong attenuation)

– limited

Channel capacity

- Nyquist limit (*noiseless channel*)
  - If an aribitrary signal has been run through a low-pass filter of bandwidth H, the filtered signal can be completely constructed by making only 2H samples per second.
  - $2\,H \log_2 V$ bps
    - bandwidth (Hz), baud, symbol, bit-per-second

Shannon limit

- Shannon limit (*noisy channel*)
  - $H \log_2 (1+S/N)$ bps
    - telephone local loop: H=3000Hz; S/N=30dB
- Modulation (telephone local loop): amplitude/frequency/phase
  - more bits / symbol * 2.4K symbol / second