ECE 333: Introduction to Communication Networks
Fall 2001

Lecture 20: Switching and Multiplexing II

Static multiplexing and circuit switching

Public switched telephone network

SONET

Circuit switch designs

Last time we began looking at point-to-point networks, and identified two key issues:

1) ***Multiplexing*** - how links are shared between sessions

2) ***Switching*** - how traffic is switched from input to output ports at nodes.

Conceptually multiplexing is very similar to medium access control. The main difference is with medium access control, users accessing the channel are geographically separate, so coordination and contention arise. With multiplexing the channel sharing is done at one location, namely in a switch, so contention and coordination are not as difficult (but, as we will see, these issues still can arise.) As with medium access control, multiplexing techniques can be divided into static and dynamic approaches (usually called statistical multiplexing). With static approaches each user is allocated a sub-channel, i.e. a fixed portion of the channel, while with dynamic approaches the allocation varies according to demand.

Recall, in Lecture 1 we identified 2 types of switching - circuit switching and packet switching. Circuit switching is used with static multiplexing techniques, while packet switching is used primarily with statistical multiplexing. Today we will consider static multiplexing and circuit switching; one place where this is used is in the public switched telephone network, we will discuss this today as well. In the next lecture, statistical multiplexing and packet switching will be addressed.

Static Multiplexing Techniques

Static approaches to multiplexing include:

- Frequency division multiplexing (FDM)
- Wavelength division multiplexing (WDM)
- Time division multiplexing (TDM)

FDM: FDM is analogous to FDMA (see lecture 13). The frequency spectrum of a link is divided up into disjoint bands and a separate band is assigned to each session. FDM is used in the Cable-TV network. It is also used on legacy analog telephone connections

Analog telephone system - In the 1930's FDM began to be used in the U.S. telephone network. In the telephone network, each analog voice channel is limited to ~3kHz by filters in the phone system. Historically, long-distance carrier systems multiplexed many analog channels using FDM over high capacity links such as microwave or coax. Each voice channel is assigned to a 4kHz band to provide adequate guard bands. In the U.S., AT&T designated a ***hierarchy*** of FDM schemes, as shown in the table on the next page. A similar, but not identical system, was standardized internationally by the ITU-T.

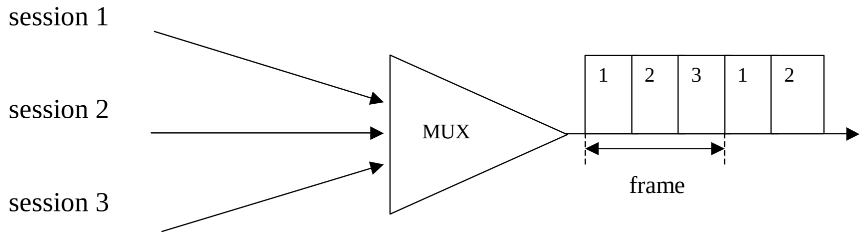
Legacy FDM telephone hierarchy:

Number of voice channels	Spectrum	AT&T	ITU-T
12	60-108kHz	Group	Group
60	312-552 kHz	Supergroup	Supergroup
300	812-2044 kHz		Mastergroup
600	564-3084 kHz	Mastergroup	

For example, 12 voice channels can be multiplexed into ***group***; 5 groups can be multiplexed into a ***Supergroup***, etc. Various larger combinations of ***Mastergroups*** have also been defined.

WDM: When FDM is used in optical communication systems it is called WDM. In WDM systems, different sessions modulate light sources (LED's or lasers) at different wavelengths (frequencies) over the same fiber. These signals can then be separated using optical components such as diffraction gratings. Using WDM current systems can multiplex 40-128 channels onto single optical fiber at rates of 2.5 - 10 Gbps each. Low rate optical streams can be multiplexed together "all-optically" without any electrical conversion. This may also be done using passive optical components. Each WDM wavelength can be treated independently and used for analog or digital data. In many cases, WDM is combined with TDM.

TDM: TDM is analogous to TDMA. Time is divided up into slots and each session is assigned a given set of slots in a cyclical order. Each slot could correspond to the time to transmit either a single bit or byte from a user or a fixed size packet. The sequence of time-slots, 1 per user is called a ***frame***.



TDM is used with digital data. TDM was introduced into the public telephone network in the 1960's and has largely replaced FDM systems in the U.S.

Note with TDM (or FDM) slots are assigned to users when a call is set-up, thus no header information is required in each slot to identify a session. However one does need to ensure that the receiver is synchronized to the start of the frame. TDM systems usually provide some method for the receiver to gain frame synchronization if it is lost.

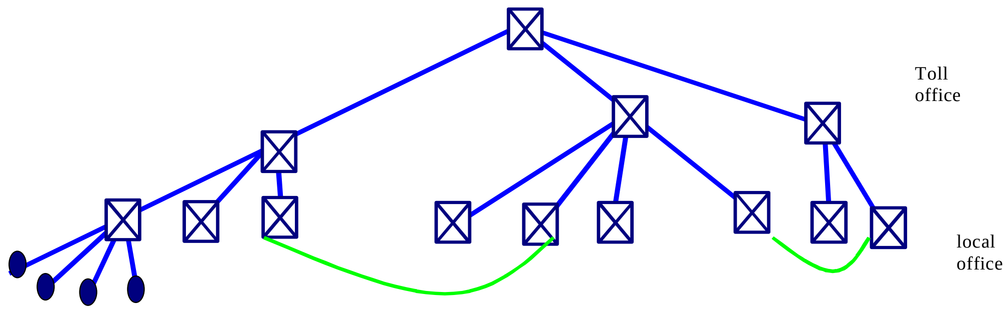
One method used for synchronization in TDM is known as ***added bit framing*** - an extra control bit is added to each TDM frame. On this "control channel" a predetermined sequence is transmitted, e.g. 10101010...The receiver searches for this pattern to synchronize, and monitors this bit to detect framing errors.

TDM is used to combine signals from constant rate incoming lines onto higher rate trunks. The synchronization of the low-rate lines is important to the operation of TDM multiplexers. For example suppose N low-rate lines are to be multiplexed at the bit level. Each line is supposed to have a rate of R bps and is to be combined onto a NR bps trunk. Thus every $1/R$ seconds the multiplexer expects to have a bit to send from a given line. If one line is not synchronized and generates bits at a rate greater than R bps, then eventually an arriving bit for this line will need to be dropped. Alternatively, if one line generates bits at a rate slower than R bps, then eventually there will not be an input bit for this line in a given frame. One solution to this is called ***bit stuffed synchronization*** - with this technique the multiplexer has slightly higher outgoing rate than NR and will periodically either add or remove extra bits to adjust the rate. The locations of these bit adjustments are signaled to the receiver so they can be accounted for.

We noted that the public switched telephone network was one example of where TDM is widely used. Besides telephone service, lines are often leased from the telephone network for connecting together nodes in WAN's. We take a closer look at the telephone network next

Public switched telephone network

The public switched telephone network basically has a hierarchical tree topology, with extra trunks added to improve efficiency (these are shown in green).



Most (residential) customers connect to a ***local central office*** through the ***local loop***. The local loop is generally copper wire and uses analog signaling. The first 3 digits of a telephone number (after the area code) identify the central office. Local central offices are connected to nearby ***toll offices*** by ***trunks***, which can be coaxial cable, microwave links, or optical fibers. The hierarchy extends up with primary secondary and regional offices, also connected via trunks. Trunks use digital signaling and TDM (in some places a combination of WDM and TDM is used.) A call between two users served by the same local office will be

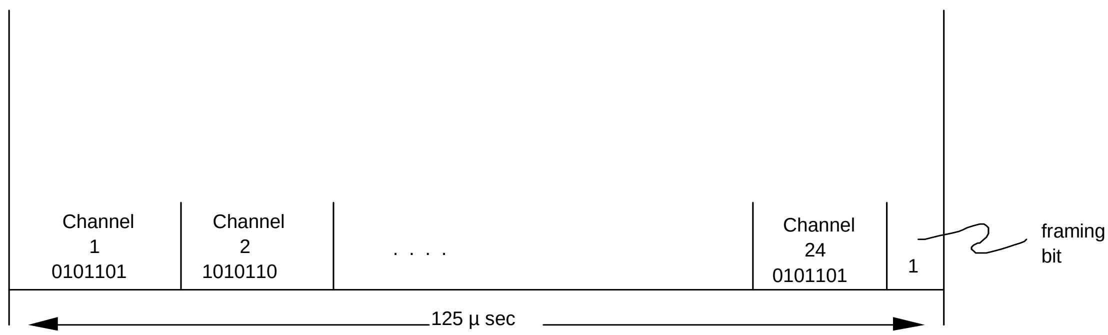
switched in the local office; other calls are switched at a higher level in the hierarchy.

In U.S. there are now several parallel hierarchies. Lower branches (e.g. end offices and local loops) are divided into regional areas called ***Local Access and Transport Areas (LATA's)***; LATA's are operated by ***Local Exchange Carriers (LEC's)*** such as Ameritech, Bell Atlantic, etc. Traffic between LATA's is handled by ***Inter-eXchange Carriers (IXC's)***, such as AT&T, MCI Woldcom, etc. Each IXC has ***point of presence*** in the toll offices of a LATA that connects to the IXC's own upper branches. This is organization is changing over time due to changes such as deregulation.

At the local office, an incoming analog signal from a user is converted into a digital signal by a ***Codec*** (coder-decoder). The codec samples the analog signal at 8000 samples per seconds and quantizes each sample using 8 bits. This results in a 64 kbps stream. At the central office on the receiveing end of a call, a Codec turns the 64 kbps stream back into analog signal. This sampling of the signal used in the telephone network is called: ***PCM - Pulse Code Modulation.***

TDM hierarchy in the telephone network

Digital signal are combined onto trunks using TDM. As with FDM, a TDM hierarchy has been standardized in the U.S. This hierarchy is based on the 64kbps signal generated by PCM. At the lowest level, 24 voice channels are multiplexed together at the byte level (note this correspond to one sample of a analog voice signal. As shown below, one additional bit is added to each frame for added bit framing, as discussed above. This results in a 1.544 Mbps signal; a trunk that carries this signal is called a "T1- line." Additional levels in this hierarchy are shown in the table on the next page.



North American TDM heirarchy

# of voice circuits	Bit rate	Name
24	1.544Mbps	T1
96	6.312Mbps	T2
672	44.736Mbps	T3

A T2 and T3 frame contain extra bits used for bit-stuffed synchronization. To be precise a there are actually two set of standards the "T"standards which specifies the electrical specifications and the "DS" standards which specify the multiplexing; thus some will refer to a T1 line as a DS1 line. Most use these terms interchangeably. Outside North America and Japan a similar system is used and designated with an "E", e.g. an "E1" runs at 2.048 Mbps and contains 32 voice channels.

Organizations can lease a T1 line from the phone company for digital data service. This provides 23 channels worth of data plus a special synch byte. Each channel allows 7 bits per frame for data and 1 bit for control.

SONET/SDH

In the late 1980's, a new digital TDM hierarchy called ***SONET (Synchronous Optical NETwork)*** was proposed by Bellcore and eventually standardized by ANSI. A related standard called ***SDH (Synchronous digital hierarchy)*** was adopted by ITU-T. SONET was designed to extend the TDM hierarchy to higher rates that could be supported over optical fibers. SONET/SDH also provided an approach that unified previously incompatible hierarchies in U.S., Europe and Japan. Today, almost all long distance telephone traffic in U.S. runs over SONET links.

SONET provides a much more efficient approach to multiplexing than most previous systems and allows access to low speed channels without having to demultiplex an entire stream. This is accomplished in part by providing a master clock to all network elements and working hard to keep them synchronized. Bits are transmitted according to this master clock. SONET also provides control channels for communication between Network Elements. A major use of these channels is to provide information for ***protection switching***- the switching of circuits to alternate paths when a failure occurs.

SONET Rates

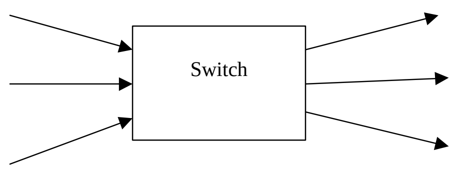
SONET specifies a TDM hierarchy starting with a basic rate of 51.84 Mbit/s, this is called an STS-1 or OC-1 signal (STS refers to the electrical standard, OC refers to the optical standard, both are often used interchangeably). In SDH the signals are called STM-n (synchronous transport mode) where n = 1,2, … Some other rates in the hierarchy are given below. The difference between the payload rate and the data rate is the bits required for control information.

ANSI SONET name		ITU SDH name	DATA Rate (Mbit/s)	Payload Rate (Mbit/s)
logical	optical			
STS-1	OC-1	--	51.84	50.112
STS-3	OC-3	STM-1	155.52	150.336
STS-12	OC-12	STM-4	622.08	601.344
STS-48	OC-48	STM-16	2488.32	2405.376
STS-192	OC-192	STM-64	9953.28	9621.504

A single fiber may use WDM to carry several wavelengths and then use SONET to time-division multiplex several lower rate streams onto each wavelength.

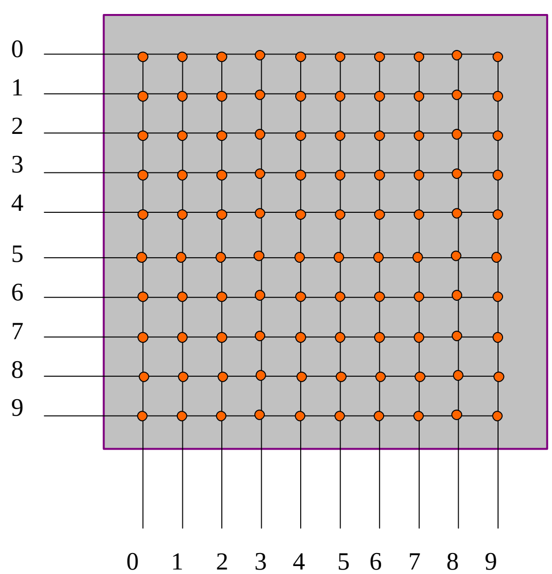
Switching

A switch is a multi-input/multi-output device. Each input and output is called a ***port***. The switch transfers signals from one input port to an appropriate output. A basic problem is then how to transfer traffic to the correct output port. In a circuit switch, the connection of input ports to output ports is determined at call set-up. This can be thought of as closing a circuit between the input and the corresponding output. In the early telephone network, operators closed circuits manually. In modern circuit switches this is done electronically in digital switches. If no circuit is available when a call is made, it will be blocked (rejected). When a call is finished a connection teardown is required to make the circuit available for another user. Switches must have some intelligence in order to perform the call setup/teardown. Note because a call set-up is required, circuit-switching networks are connection-oriented.

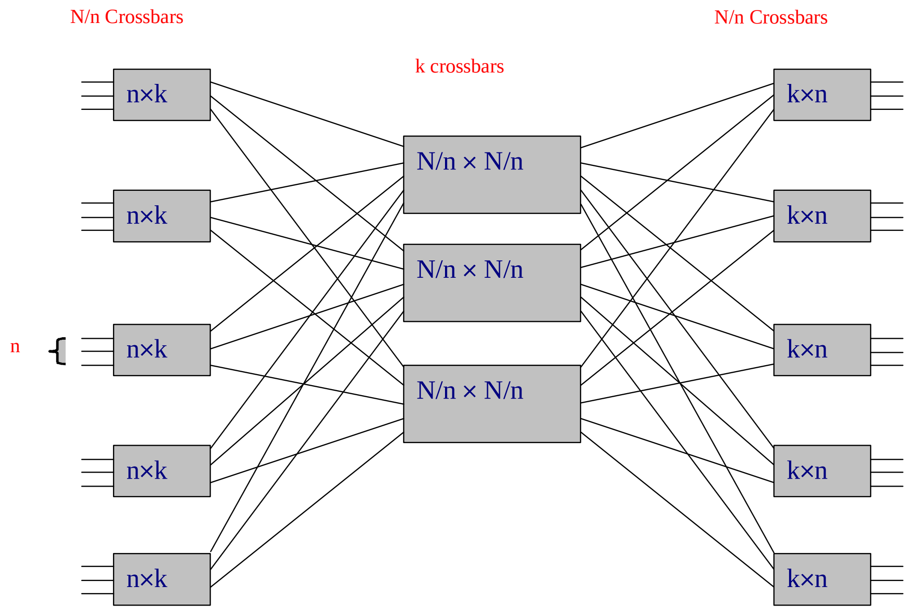


In the following we consider the design and operation of circuit switches in more detail.

Crossbar Switch - A basic example of a circuit switch is a ***crossbar switch*** as shown below. A crossbar switch with N input lines and N output lines contains an N x N array of cross points that connect each input line to one output line. In modern switches, each cross point is a semiconductor gate.



Space Division switches Crossbars are somewhat inefficient and require large number of cross-points. (The number of crosspoints is a measure of the complexity of a crossbar switch.) A more efficient design is to use several smaller crossbar switches and interconnect them. The resulting switch is called a **(multi-stage) space division switch.** An example is shown below.



Suppose there are N incoming and outgoing ports. These are divided into N/n groups of n ports. Each group of n incoming ports is attached to a nxk first stage crossbar switch. The k output ports of each of these switches are connected to one of k second stage N/n x N/n switches. Finally there is a last stage of N/n kxn switches. A incoming call is then switched through all 3 stages. In this type of switch, the number of crosspoints is

$$2kN + k(N/n)^2$$

Notice there are $(kN)/n$ paths leaving the center stage. Thus if $(kN)/n<N$, a new connection may be blocked even if the final output line for that connection is available. In this case the switch is said to be ***blocking***. Increasing k will decrease the blocking probability, but result in a more expensive switch. It can be shown that if $k \geq 2n-1$ then the switch will be ***non-blocking***, i.e. if an input and output line are both available, a connection can be made.

When the lines coming into a circuit switch use TDM, then it is desirable to be able to data on each input switch any input sub-channel to any output sub-channel. One approach would be to reconfigure the switch during every TDM slot. However this would allow a call on a given input slot to be switched only to the corresponding slot in the outgoing frame. Also this would require close synchronization on all the input lines.

Another approach is to simply demultiplex the input lines and feed each sub-channel into a port of the switch and then re-multiplex the output lines.

A final possibility is to use a ***time-slot interchanger,*** to re-order the slots, before the switch. This works by reading all the slots in a frame into memory and then writing them out in another order. This requires memory access times to be fast enough to keep up with the line rate.

Multistage switches that utilize both time-slot interchangers and small crossbar switches can be made - these are called ***time-space-time*** switches.