Agenda

- **Voice**
  - An overview of the voice network.
  - Decibels
  - Voice equipment.
  - Placing a call.
  - 2-wire and 4-wire.
  - Impairments.

Agenda (continued)

- **Data**

Introduction

- This presentation is divided into two parts: Voice and Data.
- Originally, the network was essentially all voice and when data usage began to grow, the first efforts to communicate data were directed at using the existing circuit switched network.
- Eventually, the network became majority data, to the point that voice is now carried on the data network.
- I'll start with how voice is carried in the traditional circuit switched network, and then show the transition to a data centric network. I will not cover how the old analog telephone network operated because that's now obsolete.

Short History of the US Network

- The precursor company to AT&T, American Bell Telephone Company, was formed in 1875 by Alexander Graham Bell, Gardiner Hubbard and Thomas Sanders.
- In 1882, they bought controlling interest in Western Electric, which became the manufacturing arm of AT&T.
- Gradually, this company bought up most of the independent telephone companies and, in 1885, changed its name to American Telephone and Telegraph Company.
- AT&T avoided being treated as a monopoly by promising "universal service".

Short History of the US Network

- The first digital line (a T1 line) was installed in 1962.
- The Carterfone decision in 1968 allowed equipment not manufactured by Western Electric to be connected to the system.
- Internet invented in 1969, although TCP was not implemented until 1974, and IP V4 was not until 1981.
- MCI (Microwave Communications, Inc) became the first Interexchange Carrier (IXC) in 1972, although it didn't offer switched services until 1975, and the market didn't fully open up unit 1984. Sprint was another IXC.

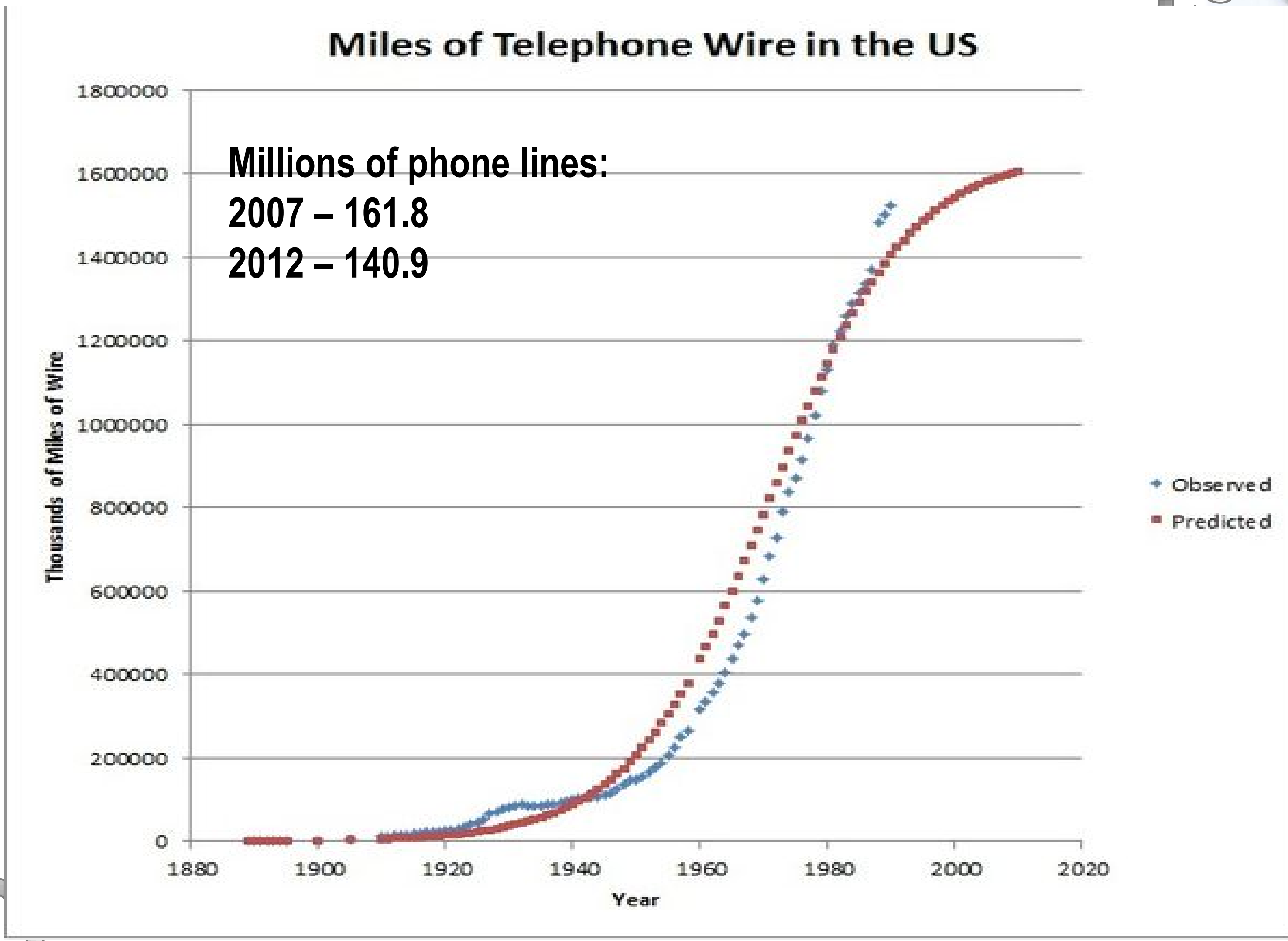
Short History of the US Network

- First cellular phone call in 1973.
- First digital switch, the 4ESS, deployed in 1976.
- First fiber optic link installed in 1977.
- Introduction of the IBM PC in 1981. Led to significant dial up modem traffic on the voice network.
- Breakup of the Bell System in 1984 (more on this later).
- SONET/SDH standardized in 1986.
- World Wide Web invented in 1989.
- Explosive growth in Cellular phone subscribers starting about 1995.

Short History of the US Network

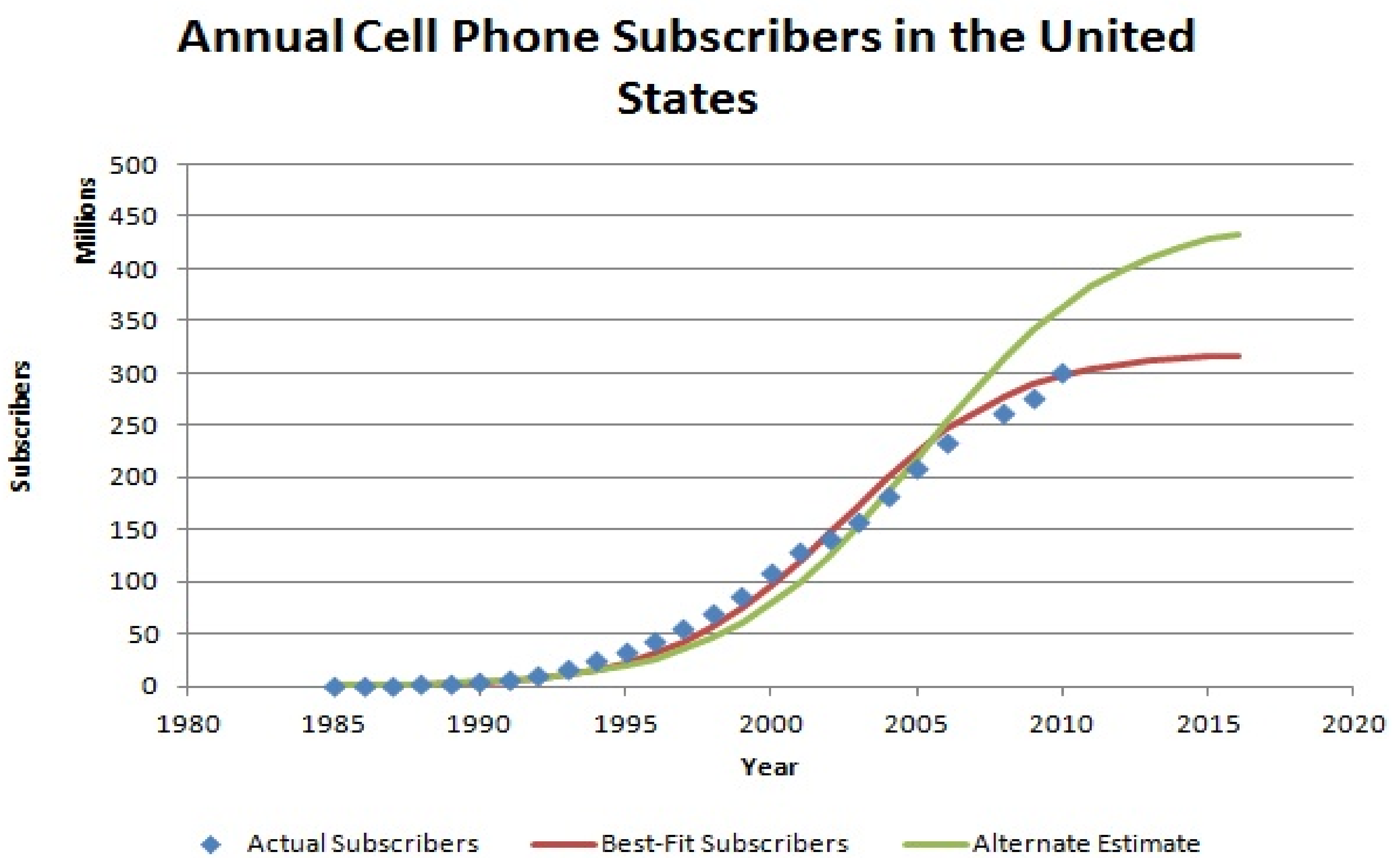
- Telecommunications Act of 1996. Allowed RBOCs to enter long distance.
- Explosive growth of the Internet starting about 1996.
- Apple iPhone introduced in 2007.

Miles of Telephone Wire in US

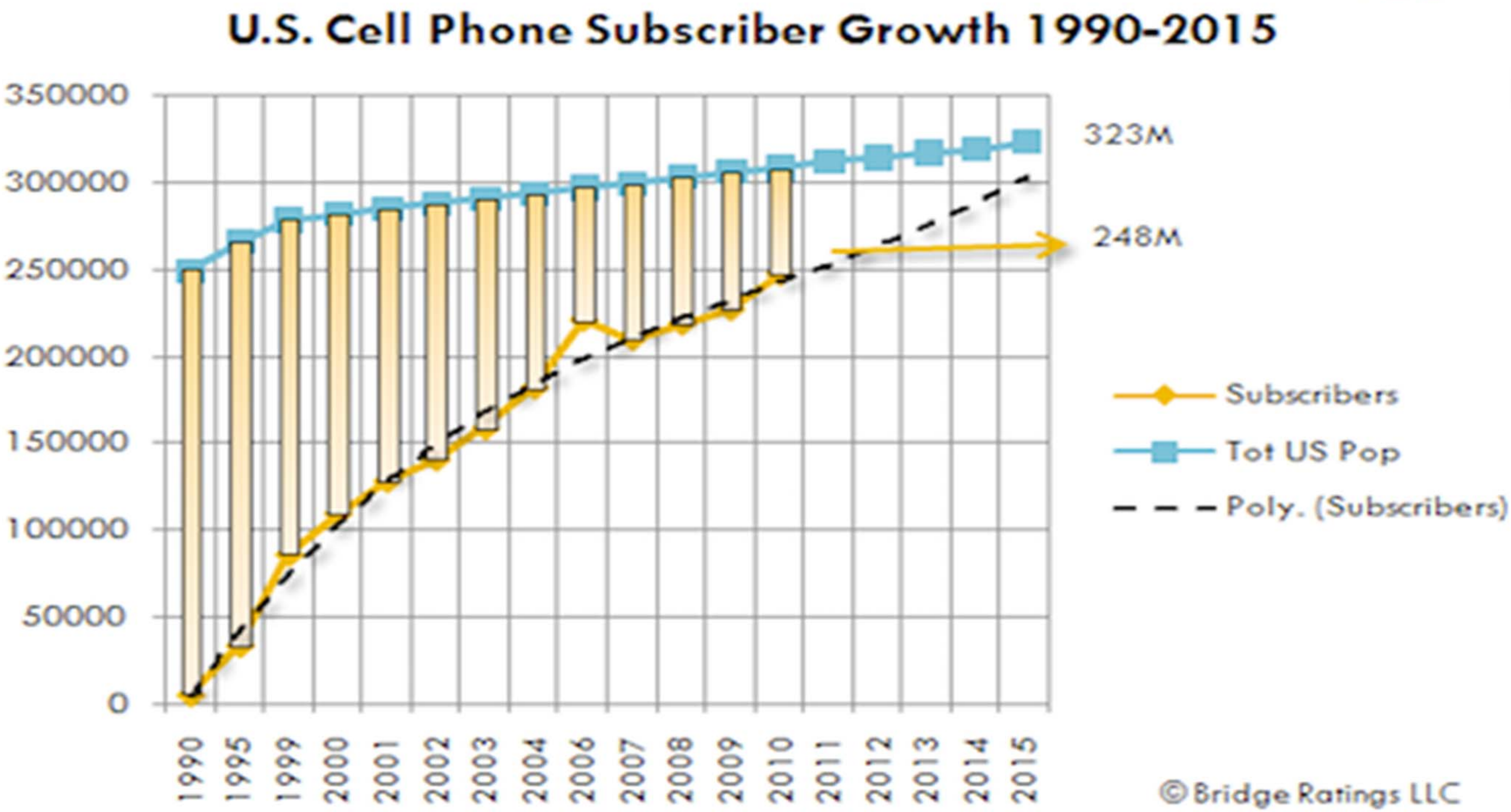


Millions of phone lines:
2007 – 161.8
2012 – 140.9

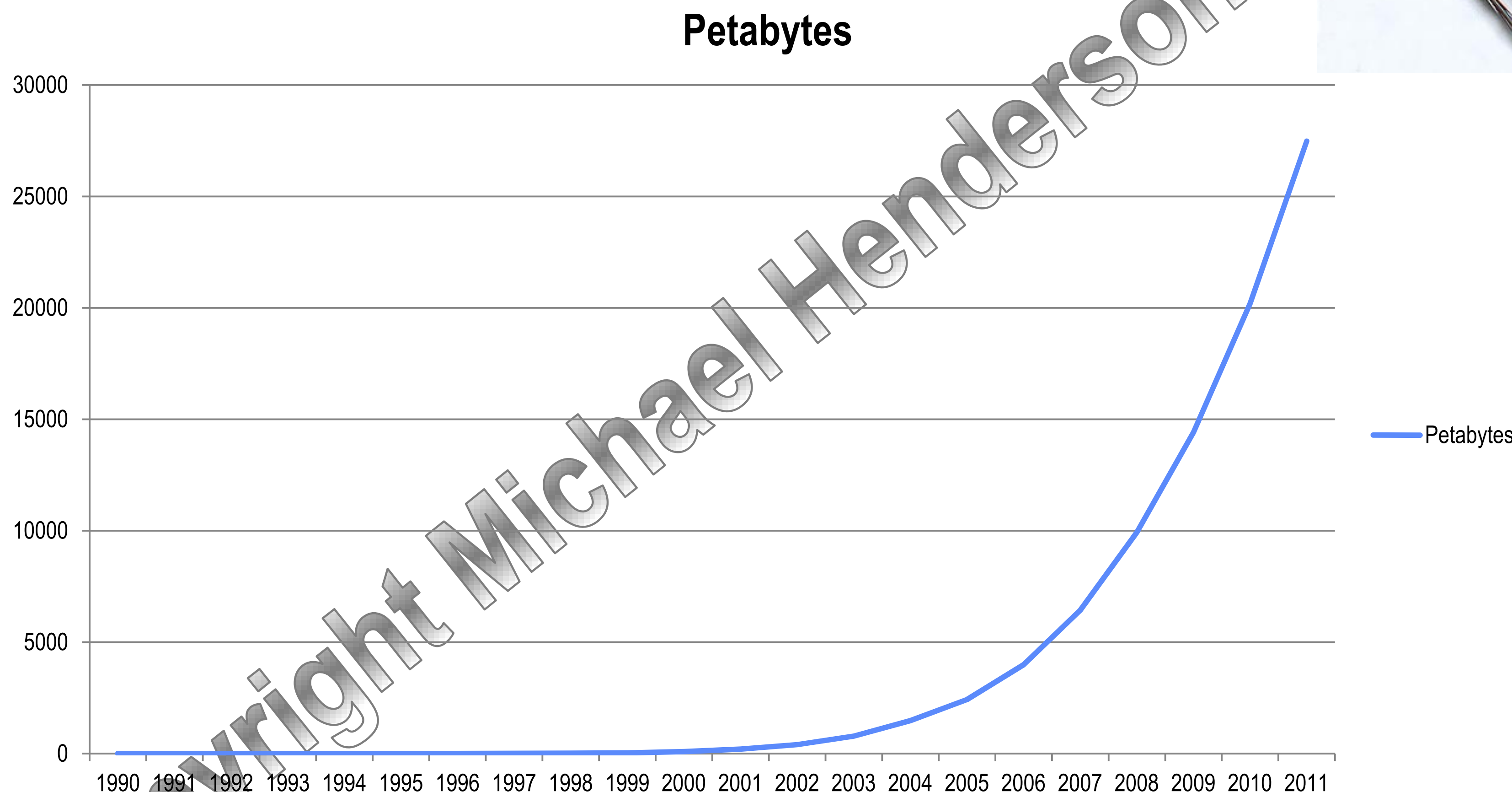
Cell Phone Subscribers in the US



Cell Phone Subscribers and US Population



Internet Traffic



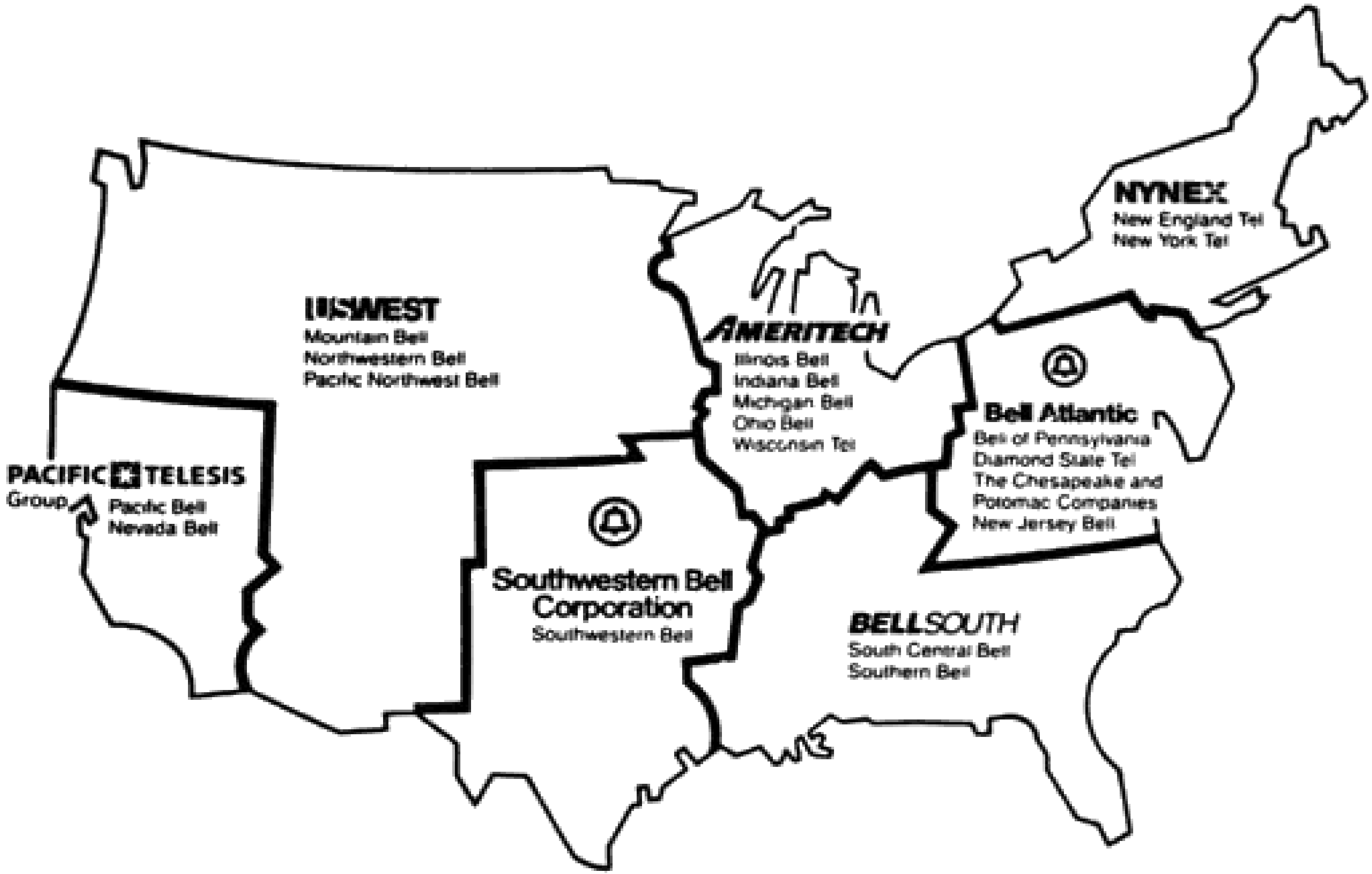
Overview of Telecommunications Industry

- **Major players by category**
  - Traditional telephone companies
  - Cable companies
  - Wireless companies
- **Major players, in order by size, 2012. Fortune 500 ranking indicated.**
  - ATT (11)
  - Verizon (15)
  - Comcast (49)
  - Sprint Nextel (90)
  - Direct TV (105)
  - Time Warner Cable (142)

Breakup of the Bell System, 1984

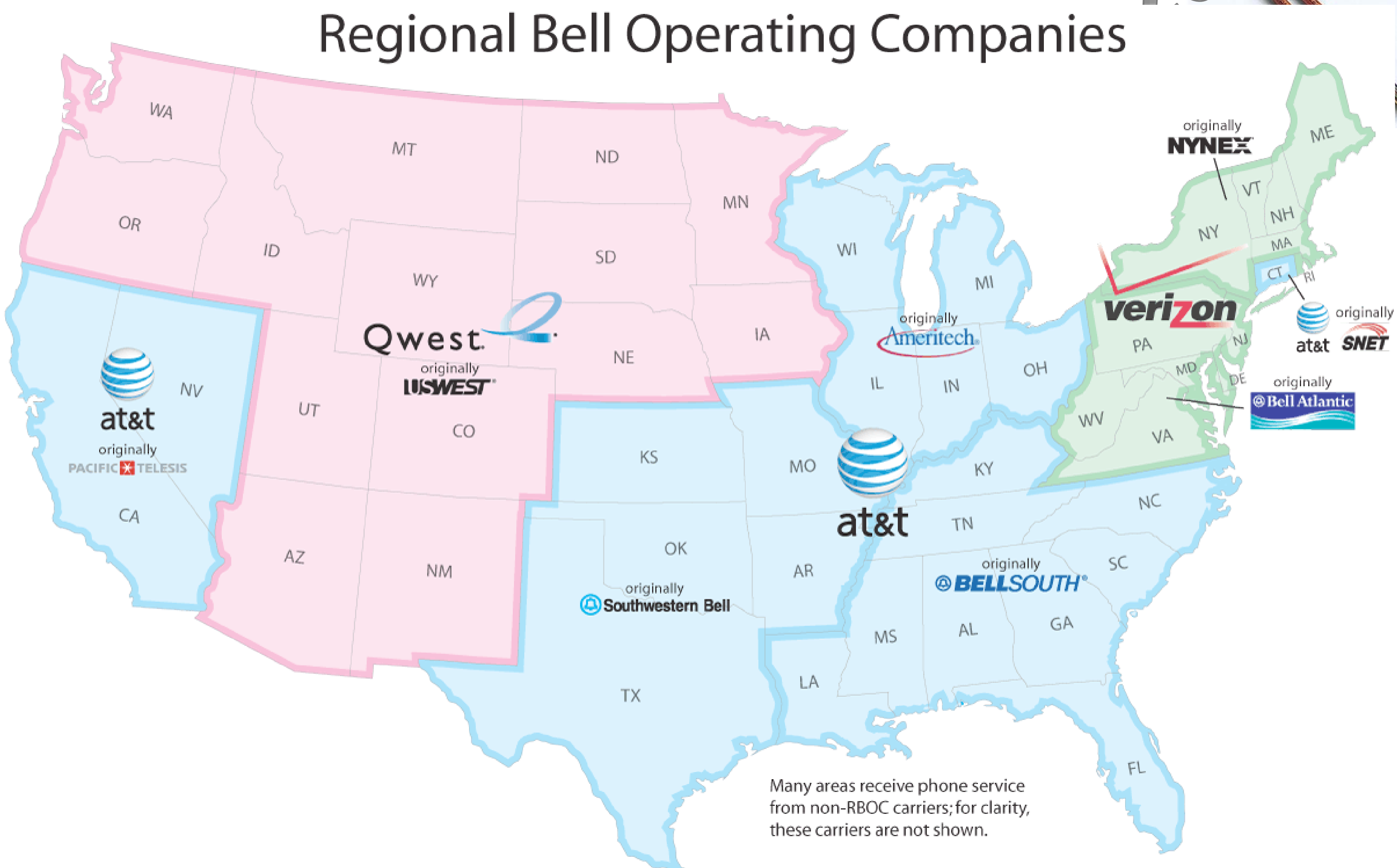
- **The Justice Department brought an antitrust suit against AT&T in 1974.**
- **Terms of the breakup were agreed to in 1982 and the breakup occurred in 1984.**
- **The breakup created seven Regional Bell Operating Companies (RBOCs)**
  - Ameritech – Now part of ATT
  - Bell Atlantic – Now part of Verizon
  - Bell South – Now part of ATT
  - NYNEX – Now part of Verizon
  - Pacific Telesis – Now part of ATT
  - Southwestern Bell – Bought AT&T, changed SBC name to ATT
  - US West – Now part of Centurylink

RBOCs 1984


US WEST
Mountain Bell
Northwestern Bell
Pacific Northwest Bell
PACIFIC TELESIS
Group
Pacific Bell
Nevada Bell
AMERITECH
Illinois Bell
Indiana Bell
Michigan Bell
Ohio Bell
Wisconsin Tel
NYNEX
New England Tel
New York Tel
Bell Atlantic
Bell of Pennsylvania
Diamond State Tel
The Chesapeake and
Potomac Companies
New Jersey Bell
Southwestern Bell
Corporation
Southwestern Bell
BELLSOUTH
South Central Bell
Southern Bell

RBOCs Today


2014


Regional Bell Operating Companies
WA
OR
ID
MT
WY
ND
SD
NE
MN
IA
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CO
NV
CA
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OK
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AR
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WI
MI
IL
IN
OH
KY
TN
MS
AL
GA
SC
NC
FL
WV
VA
PA
NY
VT
NH
ME
MA
CT
RI
NJ
DE
MD
Qwest
originally
USWEST
at&t
originally
PACIFIC TELESIS
originally
Southwestern Bell
originally
Ameritech
at&t
originally
BELLSOUTH
originally
NYNEX
verizon
originally
at&t SNET
originally
Bell Atlantic
Many areas receive phone service from non-RBOC carriers; for clarity, these carriers are not shown.

Basic Communications Concepts

- **Telecommunications is defined as "communications of information at a distance".**
- **For our purposes, "Communications of information in electrical form carried by wire, fiber, telephone, telegraph or broadcast".**
- **Information may be communicated over a single transmission medium (a link), or over a network, which consists of multiple links and intermediate nodes.**

Basic Communications Concepts

- **For purposes of this presentation, we'll define communications as the transfer of information in analog and/or binary form between two points.**

Basic Communications Concepts

- **There are three basic types of communications:**
  - Simplex: One way communications, such as a radio station.
  - Half Duplex: Data is sent in both directions, but only in one direction at a time. A CB radio is an example, where the speakers use a protocol to indicate that the other person is to speak. "Hello Alice, over". Hello Bob, how are your today? Over". (This also introduces the standard names in encryption and communications, Alice and Bob, A & B).
  - Full Duplex: Data is sent in both directions simultaneously. A standard telephone is full duplex.

Basic Communications Concepts

- **Asynchronous Transmission – Signals are only placed on the line when data is to be sent. Otherwise, the line is in an idle state.**
  - The start of transmission is signaled (from an idle line) by sending a "start" bit, a logic 0.
  - The data is then sent, followed by a stop bit, a logic 1. The primary purpose of the stop bit is to return the line to the idle state.
  - Provides character framing.
  - The receive clock is free running so the amount of data between the start bit and the stop bit is limited, usually a byte.
  - Allows use of a simple (cheap) receiver. Was used when equipment cost was very high.
  - Inefficient and usually slow.

Basic Communications Concepts

- **Synchronous transmission – There is continuous signaling on the line, even if no data is being transmitted.**
  - Allows the receiver to synchronize its receive clock with the transmit clock.
  - Requires frequent transitions on the line for the receive clock to maintain synchronization. We'll discuss some techniques to maintain frequent transitions when we talk about line codes. There are other methods, such as scrambling or encoding techniques, such as 8B/10B or 64B/66B as used in 10Gbit Ethernet.
  - Higher level protocols that use a synchronous line are required to send idle characters when no data is being sent. For example, HDLC sends the framing character 0x7E (0111 1110) repeatedly when there's no data to be transmitted.

Some Basic Telephone Terms

- **A Local Exchange Carrier (LEC) is the telephone company providing service to the subscriber.**
- **An Interexchange Carrier (IXC) is the company who transports the call between LECs.**
- **A Point of Presence (POP) is the place where the LEC and the IXC interconnect.**
- **An Incumbent Local Exchange Carrier (ILEC) is one of the baby Bells – a phone company in existance at the time of the breakup of AT&T.**

Some Basic Telephone Terms

- **A Competitive Local Exchange Carrier (CLEC) is a company who provides telephone service but is not an ILEC. An example is a cable system provider.**
- **Local Access and Transport Area (LATA) is usually defined as the serving area of an ILEC.**
- **The physical facility where the LEC houses the switch is called a "Central Office" (CO). Many COs are quite large because they were built to house mechanical telephone switches. The electronic switches are significantly smaller.**

Using decibels

- As used in communications, the decibel (dB) is used as a relative measure between two signal levels to indicate power gain or loss. It is a power ratio expressed as a log value.
- The dB is defined as $dB = 10\ log_{10}\frac{P_2}{P_1}$ (P_1 is input, P_2 is output)
- To convert from dB to power, $P = 10^{\frac{dB}{10}}$
- If $P_2 = 2*P_1$, how many dB is that?
- $dB = 10\ log_{10}2$
- dB = 10* 0.30102999
- dB = 3.010299
- So +3 dB indicates a doubling of power. -3dB indicates a halving of the power.

Decibels of Voltage and Current

- Power can be expressed as I^2R or as $\frac{E^2}{R}$
- <u>Current</u>: dB = 10 $log_{10}\frac{I2^2R2}{I1^2R1}$ = 20 $log_{10}\frac{I2}{I1}$+ 10 $log_{10}\frac{R2}{R1}$
- Since R1 = R2, dB = 20 $log_{10}\frac{I2}{I1}$

- <u>Voltage</u>: dB = 10 log $\frac{\frac{E2^2}{R2}}{\frac{E1^2}{R1}}$ = 10 log $\frac{E2^2}{R2}$ - 10 log $\frac{E1^2}{R1}$ =

  10 log $E2^2$- 10 log R2 – (10 log $E1^2 - 10\ log\ R1) =$

  10 log $\frac{E2^2}{E1^2}$ + 10 log $\frac{R1}{R2}$ Since R1=R2, and log 1 = 0

- dB = 20 log $\frac{E2}{E1}$

dBm and dBW

- There are two reference decibels, decibel referenced to a milliwatt, dBm, and decibel referenced to a watt, dBW.
- The equations are similar, but the units are different.
- $dBm = 10\ log_{10}\frac{P2}{1mW}$ and,
- $dBW = 10\ log_{10}\frac{P2}{1W}$
- To convert from dBm to power, $P_{mW} = 10^{\frac{dBm}{10}}$
- To convert from dBW to power, $P_{W} = 10^{\frac{dBW}{10}}$

Some dB Relationships

- 1mW = 0 dBm
- 1W = 0 dBW
- 30 dBm = 1 W = 0 dBW Add 30 dB to dBW to get dBm
- -30 dBW = 1 mW = 0 dBm Subtract 30 dB from dBm to get dBW
- +3 dB = a doubling of the power – factor of 2
- -3 dB = a halving of the power – factor of 1/2
- +6 dB indicates a doubling of the power, and then a doubling again. So 4 times the power.
- 1 dB is a factor of about 1.25
- 2 dB is a factor of about 1.6

Converting dB to power in your head

- Given a dB value other than a known value (see previous slide), choose the nearest power of 10. So if given 44 dB gain, choose 50 dB.
- Then take the difference between the chosen dB and the actual dB. Here, it's -6dB.
- We know that 50 dB is 10^5 or 100,000, and -6 dB is a factor of 4. So our output power $P_2 = \frac{100{,}000 P_1}{4}$ or $25{,}000 * P_1$

Now, try these

- 30 dBm =
- 26 dB =
- 45 dBm = ? dBW
- 1.6W = ?dBW = ? dBm

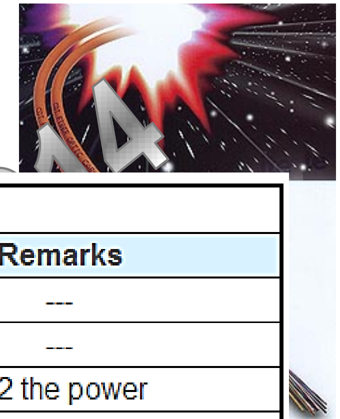
Now, try these

- 30 dBm = 1,000 mW (or 1W)
- 26 dB = $10^2 * 4 = 400$
- 45 dBm = (45-30) dBW = 15 dBW
- 1.6W = 2 dBW = 32 dBm

Rule of 3, Rule of 10

- Here's a good table showing how a 3dB change is related to the change in power.
- And also how a 10dB change is related to a change in power.

Rule of 3		Rule of 10	
Watts	dBm	Watts	dBm
0.015625	12	0.000001	-30
0.03125	15	0.00001	-20
0.0625	18	0.0001	-10
0.125	21	0.001	0
0.25	24	0.01	10
0.5	27	0.1	20
1	30	1	30
2	33	10	40
4	36	100	50
8	39	1000	60
16	42	10000	70
32	45	100000	80
64	48	1000000	90
128	51	10000000	100



dB Loss Table

Table 1 - Decibel to Power Conversion			
dB	**Power Out as a % of Power In**	**% of Power Lost**	**Remarks**
1	79%	21%	---
2	63%	37%	---
3	50%	50%	1/2 the power
4	40%	60%	---
5	32%	68%	---
6	25%	75%	1/4 the power
7	20%	80%	1/5 the power
8	16%	84%	1/6 the power
9	12%	88%	1/8 the power
10	10%	90%	1/10 the power
11	8%	92%	1/12 the power
12	6.3%	93.7%	1/16 the power
13	5%	95%	1/20 the power
14	4%	96%	1/25 the power
15	3.2%	96.8%	1/30 the power
16	2.5%	97.5%	1/40 the power
17	2%	98%	1/50 the power
18	1.6%	98.4%	1/60 the power
19	1.3%	98.7%	1/80 the power
20	1%	99%	1/100 the power
25	0.3%	99.7%	1/300 the power
30	0.1%	99.9%	1/1000 the power
40	0.01%	99.99%	1/10,000 the power
50	0.001%	99.999%	1/100,000 the power

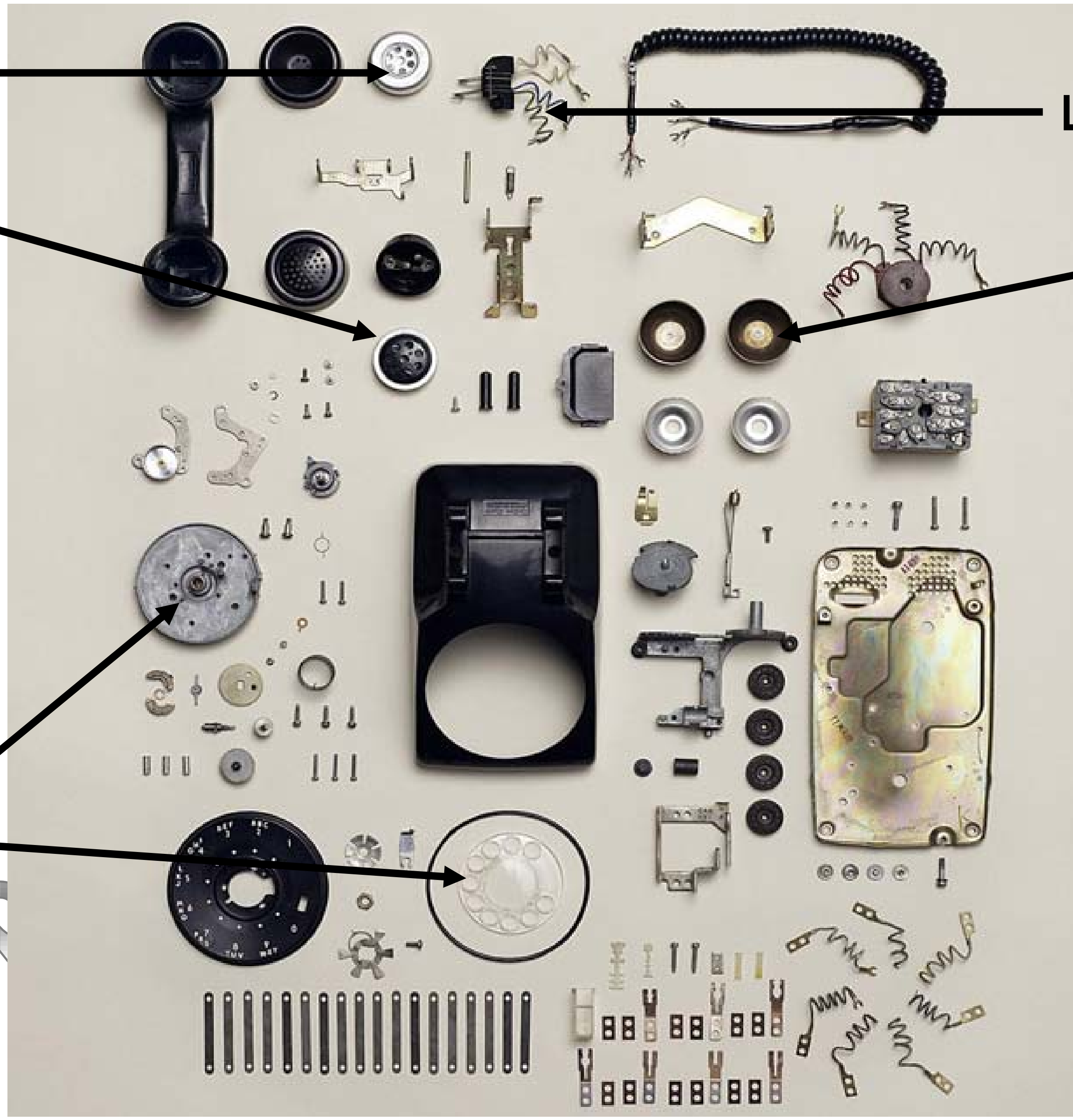
Voice Telephony

- **The telegraph was the first electrical telecommunications system.**
- **Voice was second but we'll start our discussion with it.**
- **The human ear can hear a wide range of frequencies, but speech can be understood in a narrow range. The telephone system carries voice in the range of 300 Hz to 3400 Hz.**

The Telephone

- **The telephone instrument consists of the following components:**
  - Microphone
  - Speaker (earphone)
  - Dialing apparatus (rotary dial in the past, touch pad today)
  - Alerting device (bell)
  - Switch to connect it to the line.

The Telephone Instrument



The Telephone Handset

- The microphone was originally made from carbon granules that varied in resistance with pressure.
- The loop current flowed through the microphone and varied based on the resistance of the carbon.
- The earpiece consisted of a diaphragm controlled by a magnetic coil. Changes in the flow of current through the loop caused the diaphragm to flex in time with the current changes.
- The alerting device was usually two bells, with a clapper between them. The electromagnet ringer is connected to the line through a capacitor and is disconnected when the handset is lifted.

The Subscriber Loop

- The telephone is connected to the network through a subscriber loop – a twisted pair of wires, usually 24 or 26 gauge.
- Question: Why is the wire twisted?
- Question: Why is it called a “loop”?
- The local loop connects to the telephone instrument and to the switch in the central office.
- The telephone company puts a voltage on the loop but no current flows until the subscriber takes the phone “off hook” which completes the circuit.
- The flow of current signals the switch that the subscriber wants to make a phone call.

Call Signaling on the Local Loop

- **For an outgoing call:**
  - Subscriber lifts instrument which connects the local loop.
  - Current flows through the loop, which the switch detects.
  - The switch puts a line card on the line, which places dial tone on the line.
  - The subscriber dials the called number, which the switch receives.
  - The switch places the call, and gives feedback tones: ringing, busy, etc.
  - When the called person answers, the circuit is completed.
  - When call is completed and one party "hangs up", loop current stops, which signals the switch that the call is complete.
  - Circuit is then taken down.

Call Signaling on the Local Loop

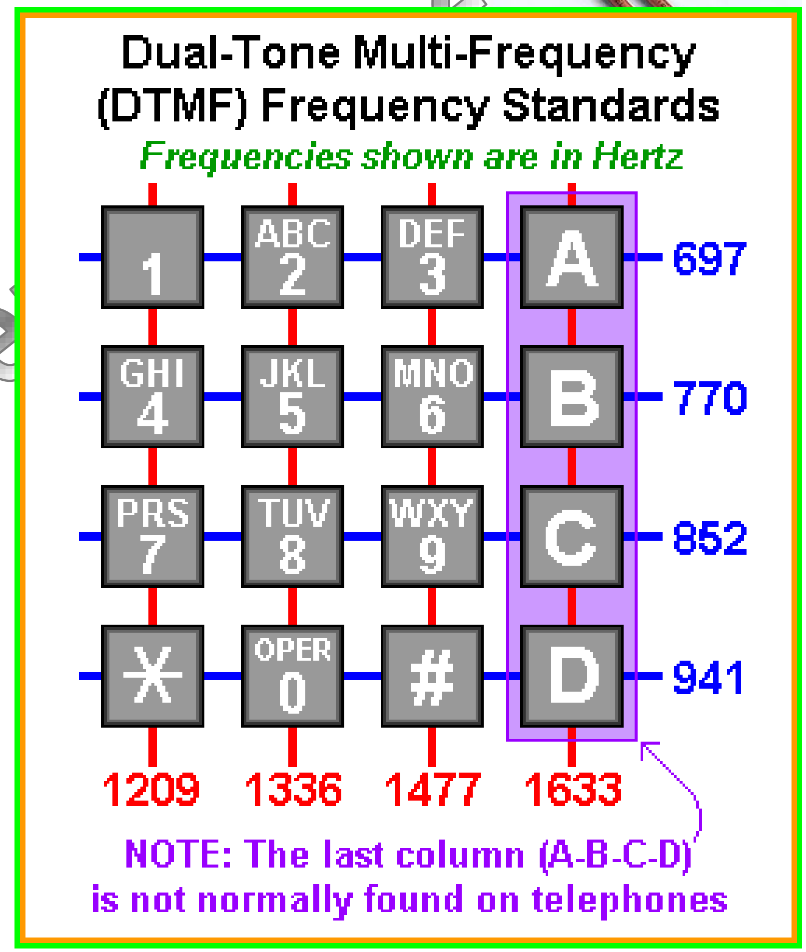
- **For an incoming call:**
  - The switch receives the called number and cross references the number to the specific local loop.
  - Ringing voltage is applied to the line. Ringing voltage is about 100 volts, 20 Hz. In the US, it is applied for about 2 seconds, followed by about 4 seconds of silence.
  - Ringing cadence is modified for different parties on a “party line”.
  - Note: Caller ID is sent between the first and second ring, via Bell 202 FSK modem technology.
  - When the subscriber lifts the receiver, loop current flows and the switch knows the called party is ready to receive the call. The circuit is completed.
  - If no answer after some number of rings, the switch may forward the call to voice mail.

Dialing

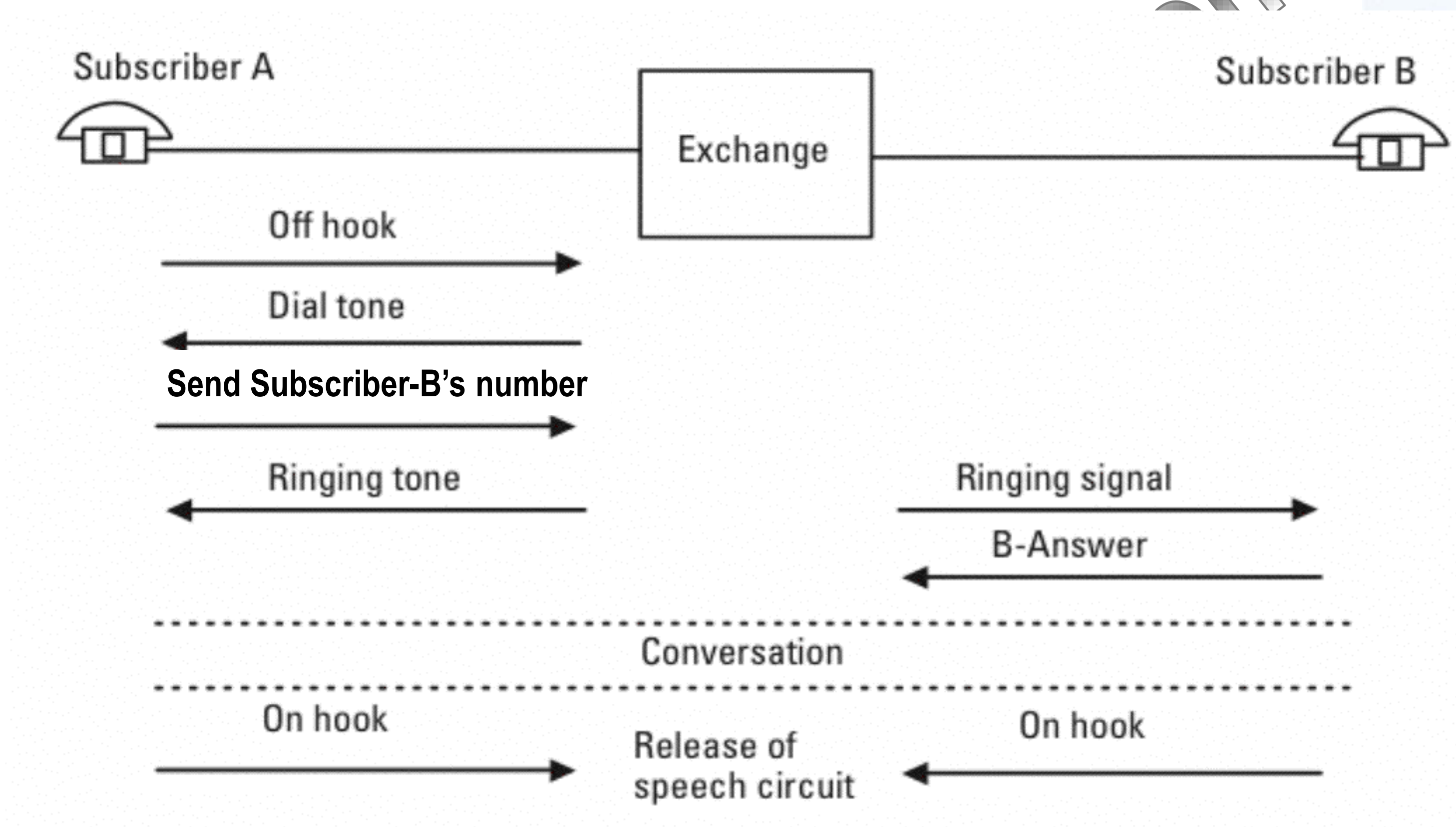
- Rotary dialing caused the line to be disconnected (broken), which was detected as interruptions in the current flow by the switch.
- A 1 digit broke the line one time, a 2, two times, etc.
- Pulses were sent 10 times per second.
- Required a “rest” time between digits to signal breaks in digits sent.
- A person could dial numbers from a “locked” phone (locked dial) by “tapping the hook”.

DTMF Dialing

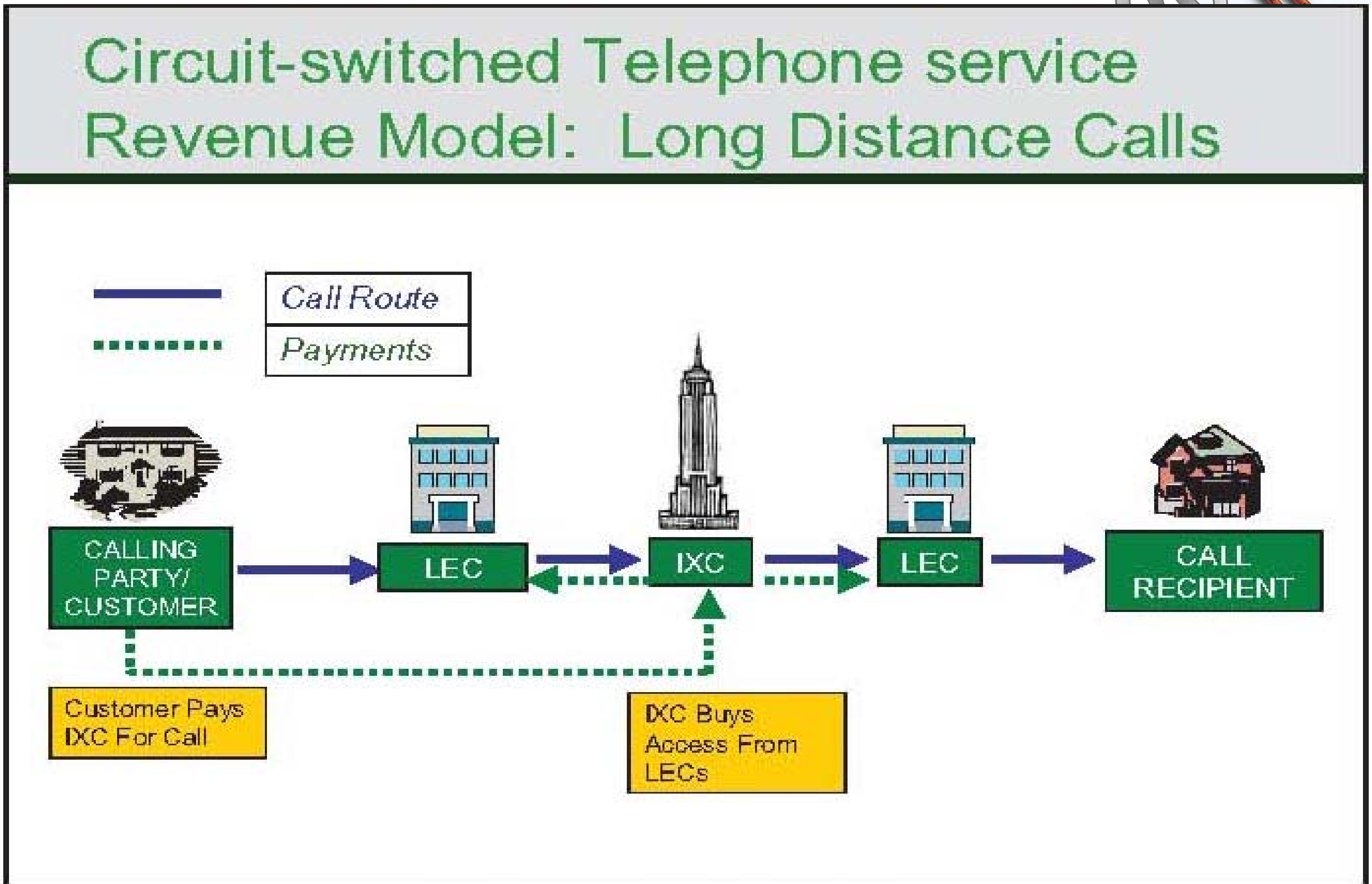
- When a button is pressed, two tones are sent to the switch.
- The last column is not normally used. The US Armed Forces used that column on their (now defunct) Autovon network.



Call Sequence



A Long Distance Call

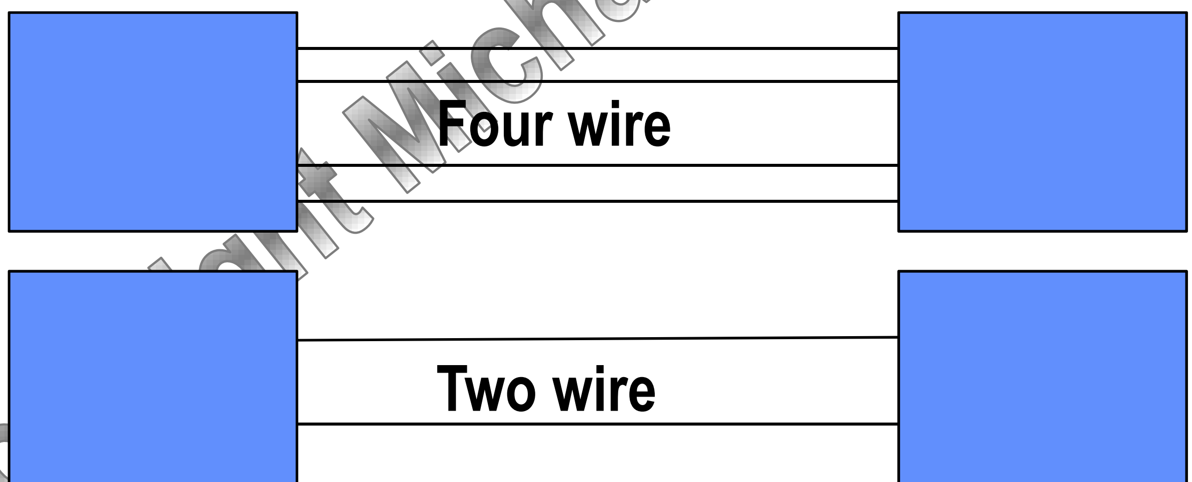


Signaling System 7

- Signaling System 7 (SS7) is a signaling protocol used to set up and tear down calls.
- Defined by the ITU and used by telephone companies worldwide.
- It is communicated in a channel separate from the telephone call.
- Called “Common Channel Signaling” (CCS).
- Older techniques sent the signaling information in the voice channel and was called “Channel Associated Signaling” (CAS). The ITU definition was Signaling System 5 (SS5).

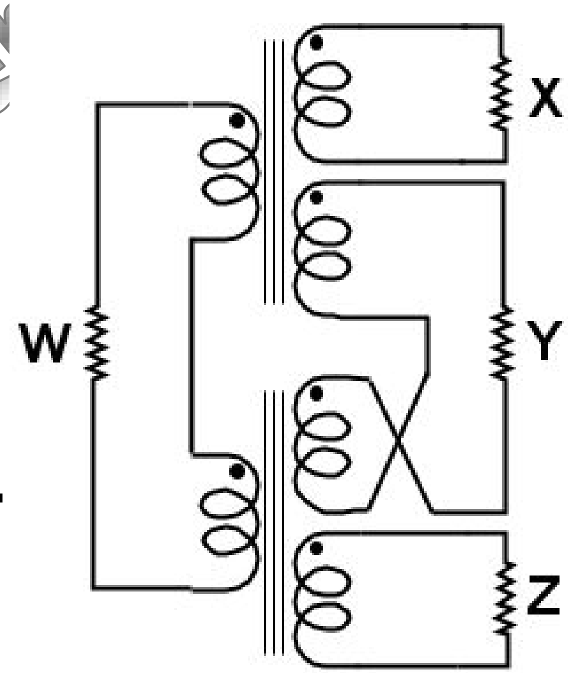
The Local Loop

- **Any bi-directional communication channel requires a transmit path and a receive path.**
- **Often this is a four wire path, with two wires sending data one way and the other two sending data the other way.**
- **But the local loop is only two wires and information is sent both ways. How is that accomplished?**



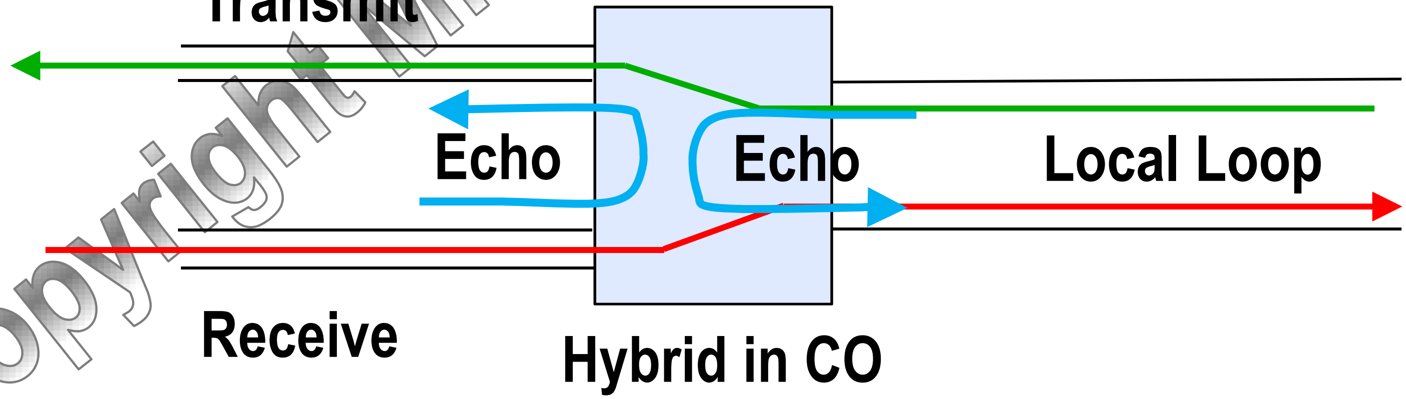
The 2 Wire – 4 Wire Hybrid

- The hybrid has the characteristic that a signal applied at X will appear at W and Y but not at Z.
- A signal applied at W will appear at Y and Z but not at X.
- So a hybrid introduces at least a 3 dB loss.
- But allows use of 2 wires instead of 4 wires.
- Requires impedance matching between the line and the hybrid so usually some signal “leaks” to the other port.



The 2 Wire – 4 Wire Hybrid

- The hybrid converts between the four wire system in the telephone and the two wires of the local loop.
- A hybrid is used at the switch in the central office to convert to four wire. The call is carried on a four wire system.
- If the hybrid does not provide a good impedance match it may cause echo – signal reflections off the hybrid.



Transmission of Information

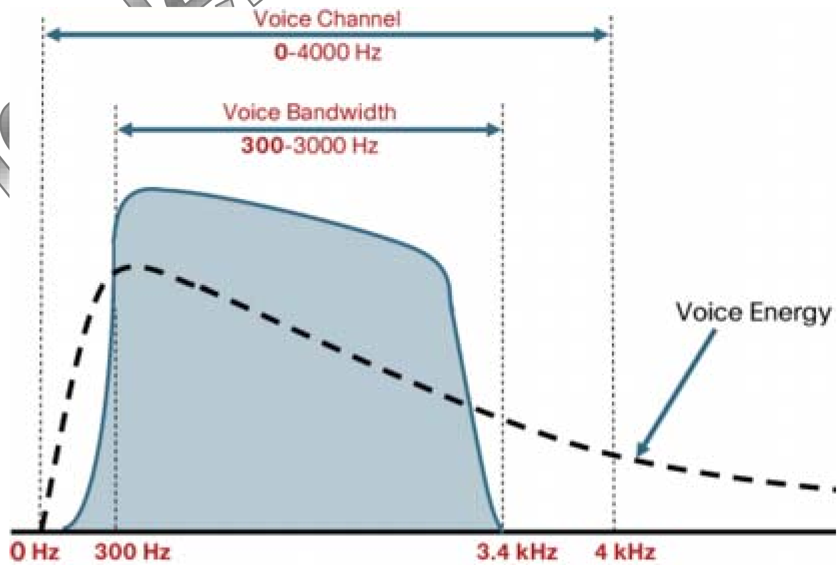
- **The most common impairments on a transmission system are:**
  - Attenuation
  - Interference
    - Coupling between pair in the wire bundle.
    - Near end cross talk (NEXT) – From Tx to Rx at the point of observation
    - Far end cross talk (FEXT) – From Tx to Rx at a distant location.
  - Noise – Thermal noise. Noise is measured in dBrn, (dB above reference noise, which is 1 picowatt).

Impairments (continued)

- **Echo**
  - If the echo occurs on the hybrid on your end, it is near end echo.
  - If the echo occurs on the hybrid at the central office (or beyond), it is far end echo. Far end echo (with high delay in the echo) is the most noticeable.
- **If the echo is severe, feedback will occur and "singing" will result. The same sound that you hear when you put a microphone too close to a speaker.**
- **Echo is dealt with by inserting loss in the transmission line or by using echo cancellers.**

Noise on a Telephone Call

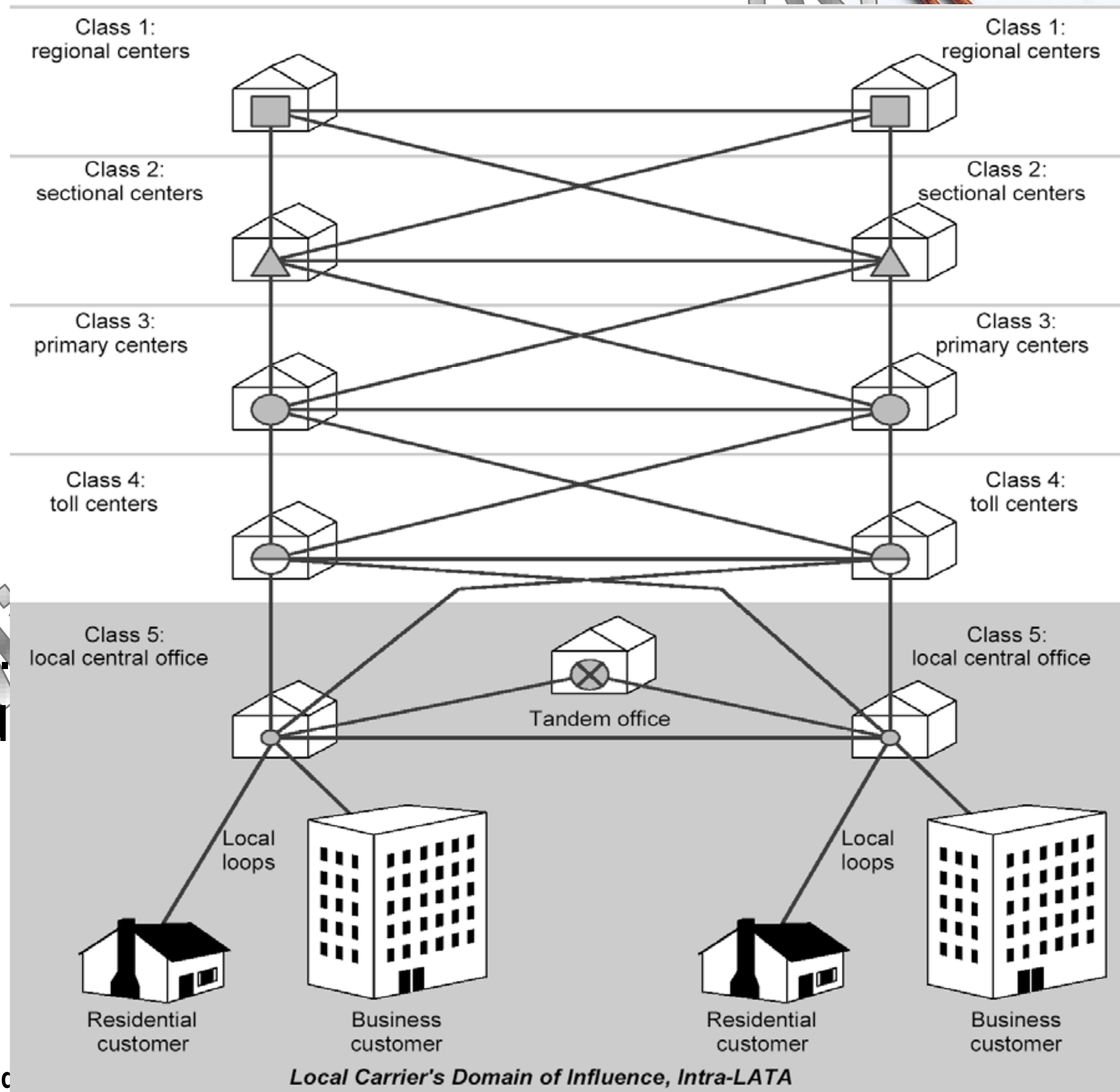
- The telephone channel is not perfect but usually has a response curve, meaning that the signal has amplitude distortion.
- The selective attenuation of certain frequencies affects the perceived quality of the call.
- The lowest sound that a person can hear is about -90 dBm. If noise rises above that, the caller will notice.



- Since the network went digital, noise is generally not a problem.

Hierarchy of Telephone Switching

- Class 1 was for international calls.
- Class 2 was for calls betwee regions (LECs).
- Class 3 handled major metropolitan centers.
- Class 4 interconnected class 5 switches in a town.
- Class 5 was a serving switch.
- Pretty much only Class 4 and Class 5 switches today.



Telephone Numbering

- At the highest level is the country code. The US and Canada is 1.
- Next is the area code. Area codes can cover large areas or parts of a metropolitan area – it depends on how many phones are in the area. When area codes were first assigned, rotary phones were in use so “short dialing” codes were given to major metropolitan areas, such as 212 to NYC, and 213 to LA.
- Next is the three digit exchange code.
- Finally, the four digit phone identifier.

Routing Based on the Telephone Number

- For international calls, the caller first dials the international prefix, which is 011 in the US.
- Then dial the country code. For France the country code is 33. For England, the country code is 44.
- If calling within the United States, you'd dial 1 to indicate long distance", next dial the area code, then the exchange, then the phone identifier.
- The switches route the calls based on the numbers. For a call within the US, the area code opens a channel to a switch in that area code area. When the exchange number is dialed, the circuit is extended to the Class 5 switch serving that exchange. Then the phone identifier is used to select the proper loop to ring the phone.

Phone Number Summary

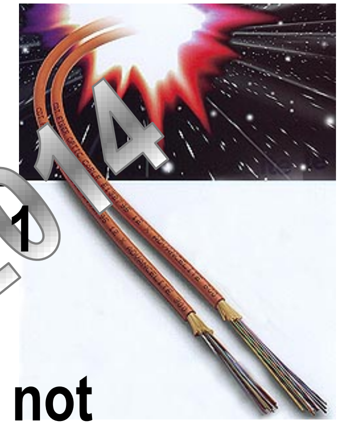
- **For US dialing:**
  - 1 for long distance
  - Three digit area code, such as 212
  - Three digit exchange code, such as 634
  - Four digit phone identifier, such as 1234
- **For international calls**
  - 011 for international call
  - Country code, 1 to 4 digits
  - City code
  - Phone identifier

How Many Phones can be Supported?

- **There are 800 possible area code numbers but some are reserved for special uses. Let's assume 700 possible area codes. (Can't start with 0 or 1, so 8*10*10) (800, 888, 900 not real numbers)**
- **The exchange number can not have 0 or 1 as the first digit, nor x11, nor 555 (Hollywood #)*, which leaves 791 exchange #s. So the maximum number of phone numbers in an area code is 7,910,000. But some numbers are used for special purposes so the actual number is less. Let's assume about 7,500,000. (Actually, 555 is beginning to be used for some real #s.)**
- **Using these assumptions, the number of phone numbers possible in the US and Canada is around 5,250,000,000, or about five and quarter trillion phone numbers.**

***Actually, only the numbers from 555-0100 to 555-0199 are now fictitious.**

Area Codes



- x11 is not used because if can cause misdialing if the 1 prefix is not used (for example, 911, 411, etc.)
- Numbers with the second and third digit the same are not used. So 233 would not be used.
- Numbers with 9 in the center, such as 392, are not used because they are used for service codes or reserved Same for 37x, and 96x - not used because they are used for service codes or reserved.
- The toll free numbers are 800, 822, 833, 844, 855, 866, 877, 880, 881, 882, 883, 884, 885, 886, 887, 888, 889. Not all of these are used today. Note that many of these violate the requirements specified above - specifically that the second and third digits not be the same.
- 890 to 899 are reserved so far.

Some Dialing Codes

- 211 - Local community information or social services.
- 311 - City government or non-emergency police matters.
- 411 - Local telephone directory service.
- 511 - Traffic, road, and tourist information.
- 611 - Telephone line repair service or mobile telephone company customer service.
- 711 - Relay service for customers with hearing or speech disabilities.
- 811 - "Dig safe" pipe/cable location in the United States.
- 911 – Emergency dispatch for fire department, ambulance, police.

The Telephone Network

- Designed to carry two way voice – digital voice these days.
- Full duplex circuit switched, low delay circuits.
- Switching via computer controlled electronic switches.
- The backbone is fully connected with redundant links.
- The telephone number indicates the routing, except for 800/888/700/900 type calls.

Components of a Network Call

- End systems, such as a telephone, modem, or fax
- Signaling
- Switching
- Transmission

End Systems

- Telephone already described, modems will be addressed later.
- Uses Pulse or Tone dialing.
- Today, the dialed numbers are collected at the central office before the call is set up. In the old days, the circuit was set up in stages, as the subscriber dialed the number.

Signaling

- The network uses Signaling System 7 today, which uses a separate channel to communicate with the other switches required to set up the call.
- Called Common Channel Signaling, as opposed to Channel Associated Signaling which used the call circuit to communicate the call set up information.
- Circuits are temporary, set up for the call and torn down when the call is completed.

Signaling Controller

- **A switch requires both a switch matrix and a switch controller.**
- **The switch controller directs the switch and does not participate in the actual transfer of the data.**
- **Functions**
  - Collects the dialed number and routes the call.
  - Provides common control signals - Dial tone, ringing, busy tone
  - Provides ringing voltage to signal incoming call to subscriber.
  - Billing for calls.
  - Lookup and routing of 800/888/700/900 type calls.

Switch Controller

- **Switch controllers are special purpose computers, linked by their own internal communications network.**
  – Called Common Channel Signaling (CCS).
- **Earlier switching control was through in band signaling.**
  – Could be “hacked” to place free calls.
  – Inflexible.

Switching

- **Any subscriber can call any other subscriber.**
- **Switches establish temporary circuits through trunks between switches.**
- **Switches have two components:**
  - Switch controller
  - Switch matrix
- **A switch transfers data from an input to an output.**
  - Modern switches can handle many simultaneous calls, perhaps 200,000.
  - Usually uses a combination of space and time-division switching.

Transmission

- **Trunks between offices carry the telephone calls.**
- **Many circuits share the same trunk through multiplexing.**
- **Originally, frequency multiplexing was done.**
  - Each call of 4 KHz was multiplied by another frequency which shifted the channel upward in frequency.
  - Multiple calls were stacked in frequency into a “group”.
  - Then groups were stacked into “Supergroups”
  - Supergroups could then be stacked into larger aggregations called Master- groups or Jumbo-groups.
- **Managing the frequency multiplexing was difficult and led to significant crosstalk between channels, including the ability to hear another party’s conversation.**
- **Obsolete in today’s network (thank goodness!)**

Digital Multiplexing

- **Must convert voice to digital (will be described later). Eight bits per sample, 8,000 samples per second, giving 64 Kbps per voice channel.**
- **Channels are digitally multiplexed, either byte or bit interleaved, and carried over high speed digital circuits. Example: T1 circuits operating at 1.544Mbps.**
- **Requires additional bits for framing. T1 framing will be described later.**

Digital Hierarchy in the US

- Called the Plesiochronous Digital Hierarchy (PDH) because the clocks are close in frequency but not exact.
- Four levels of signals, each carrying more channels.
- We'll examine each of these in more detail later.

Signal Name	Number of voice channels	Data rate
DS0	1	64Kbps
DS1	24	1.544 Mbps
DS2	96	6.312 Mbps
DS3	672	44.736 Mbps

Transmission Mediums

- Twisted pair
- Coax cable
- Terrestrial microwave
- Satellite microwave
- Optical fiber – extremely high capacity

Optical Fiber

- **A strand of very pure glass.**
  - Has two components, a core and cladding.
  - The core has a higher refractive index than the cladding.
  - Guides light by total internal reflection.
- **Extremely high capacity – can support multiple channels (lambdas) each running at perhaps 10 Gbps.**
- **Low attenuation, and amplification in the optical domain.**
- **Extremely low BER (bit error rate).**
- **Immune to electromagnetic interference.**
- **Very difficult to tap.**

Satellites

- **Geosynchronous satellites.**
  – Long distance transmission at very high data rates.
  – Orbit about 36,000 Km.
  – High delay – about 125ms one way, 250ms up and back. Does not work well for voice. Certain data protocols do not work well with that much delay.
  – Excellent for broadcast TV – delay not a problem and very wide coverage.
- **Low earth orbit (LEO) satellites**
  – Requires multiple satellites – e.g., Iridium
  – Satellites must be tracked to communicate with them
  – Handoff a complicating factor.

Cellular

- Cell systems use many different channels (frequency slots).
- Frequencies are reused across the cell serving area, but adjacent cells do not use the same frequency slots.
- Frequency slots are allocated when a call is placed or received.
- As users move within the serving area, they are handed off to another cell and assigned a different frequency slot.

Telephone Network Challenges

- **World is moving from voice to data. Much more data transmitted today than voice.**
- **Characteristics of data is significantly different from voice.**
  - Bursty
  - Requires a high BER.
  - Exponential growth in bandwidth needs.
- **Future network will be integrated voice and data (VOIP).**
  - Challenge is manage the transition.
  - Must continue to support legacy systems, e.g., T1 links.

Questions?

Handling Speech

- **Speech is an analog function, made up of multiple frequencies of sound.**
- **There are three important aspects of an analog signal:**
  - Frequency
  - Amplitude
  - Phase

Handling Speech

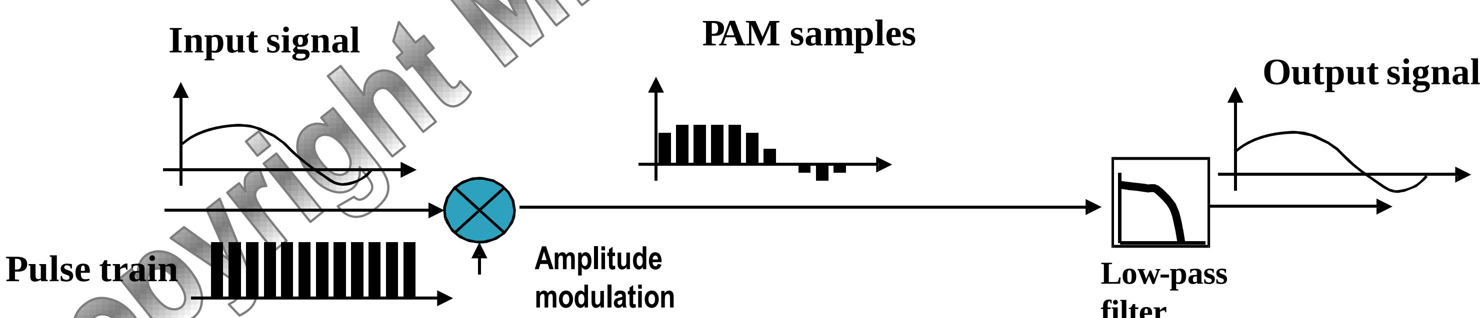
- The fundamental frequency for men is between 85Hz and 180Hz, and between 165Hz and 255Hz for women. However, speech contains many harmonics so even if the fundamental frequency is missing (as it is in the telephone system) enough harmonics are there to give the impression of the fundamental.
- While speech contains frequencies to 10KHz (and perhaps beyond) we bandpass filter speech between 300Hz and 3400Hz for the telephone system and maintain good intelligibility.

Speech Digitization

- **Why digitize speech for the telephone network? The network started as analog – why change?**
  - Provides MUCH better voice quality.
    - Noise does not accumulate as the signal travels through the system.
    - Eliminates cross talk between voice calls.
  - Easier to handle digital signals with today's electronics.
  - The system can handle many more calls in digital form than in analog form.
  - Easier to switch.
  - The speech can be easily processed, including coding it in lower bit rates and encrypting it.

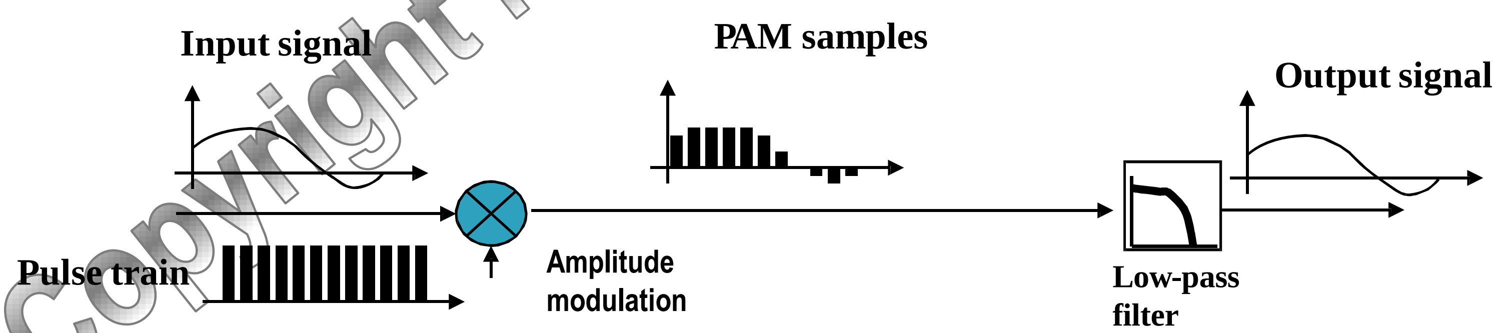
Converting Speech to Digital

- Speech is converted to digital for the telephone network using a technique known as Pulse Code Modulation (PCM).
- Did you ever wonder why it's called "Modulation"?
- Let's examine how the digitization is done.
- First, let's produce a Pulse Amplitude Modulation signal from an input speech signal.



Converting Speech to Digital

- You can see from the diagram that we're modulating a pulse train (8,000 pulses per second for the telephone system) with the speech.
- This produces a pulse train with amplitudes that reflect the incoming signal.
- If we low pass filter the pulse train, we will recover the original signal.
- But, there are some limitations…

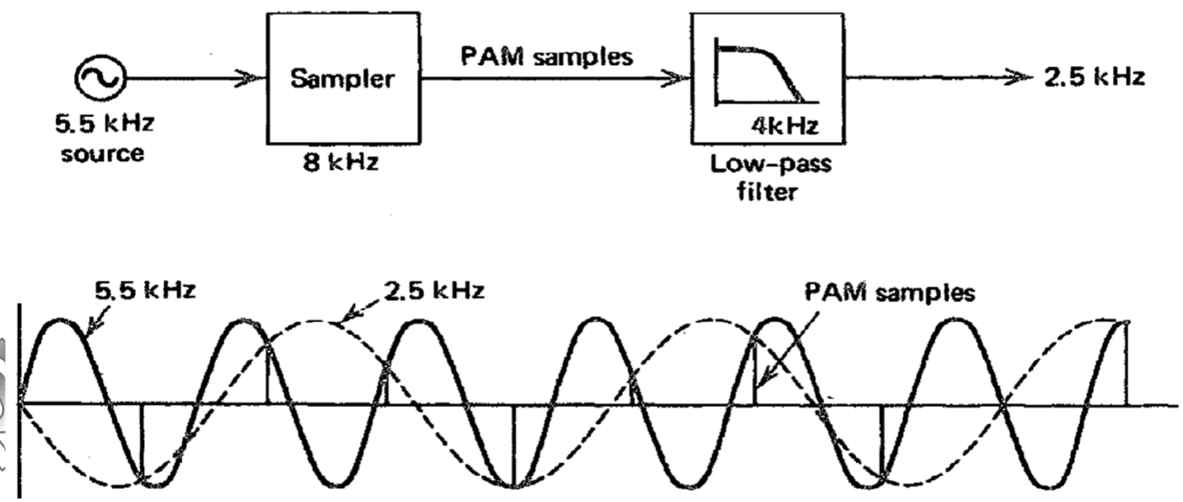


Nyquist-ShannonSampling Theorem

- Harry Nyquist (1889-1976) worked for Bell Labs and in 1924 (long before we could digitize signals) showed that the maximum rate that telegraph signals could be sent was twice the bandwidth of the channel.
- Claude Shannon (1916-2001) also worked for Bell Labs and laid the mathematical foundation for information theory. Using Nyquist's earlier work, Shannon showed that the minimum sampling rate for a signal was twice the bandwidth of the signal. Or, $f_s > 2 * BW$
- This is generally called the "Nyquist Criterion" today.
- So by sampling speech 8,000 times per second, we can transmit a signal with a maximum bandwidth of 4KHz.

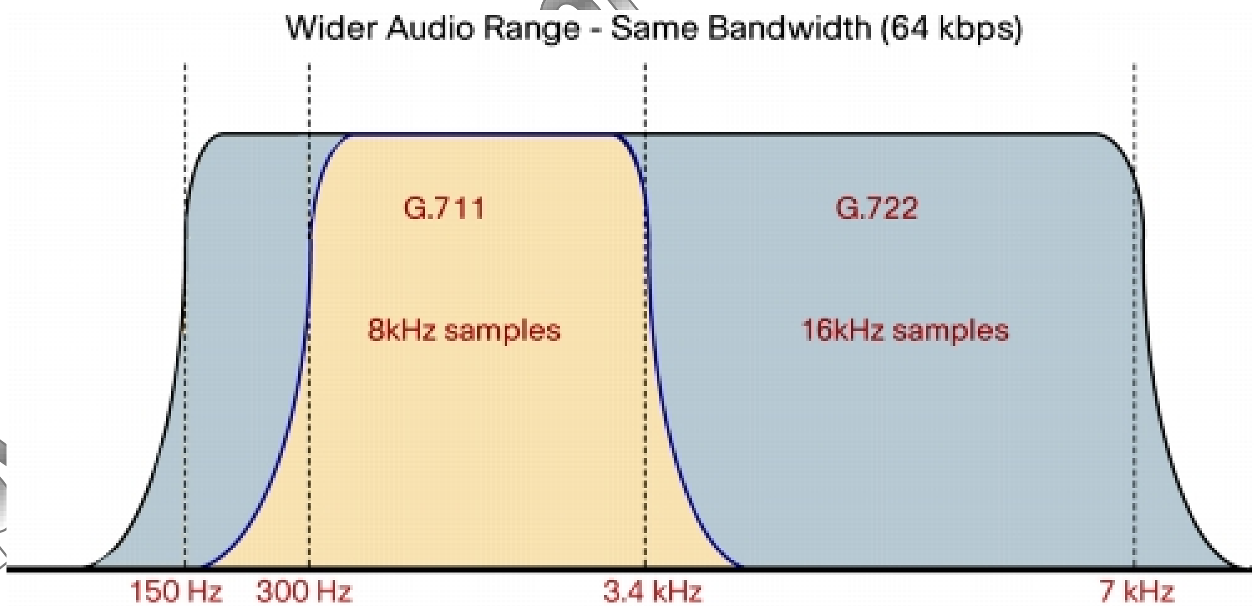
Aliasing

- If a signal is not sampled to the Nyquist Criterion, aliasing will occur.
- The higher frequency signal will be reconstructed as a lower frequency signal.
- Here, a 5.5KHz signal is sampled at 8K samples per second.
- When recovered, the signal appears as a 2.5KHz signal.



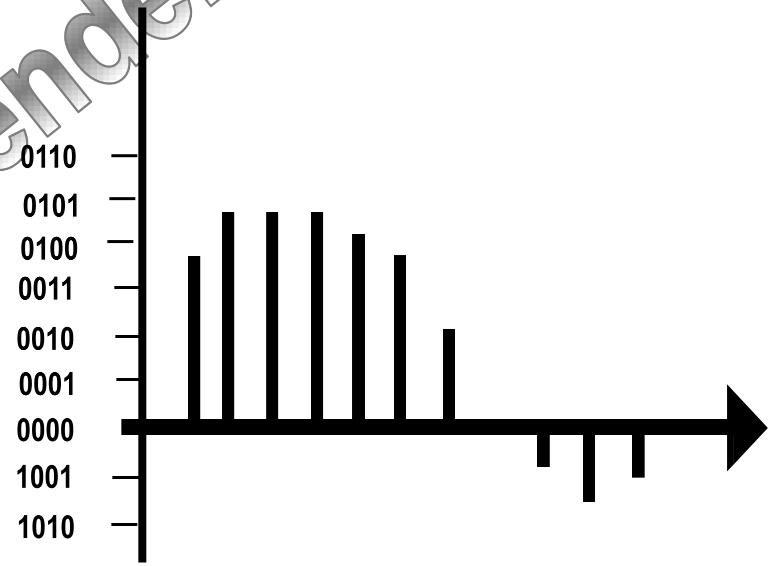
Aliasing

- This means that the input signal must be bandpass filtered before it is sampled – to remove any frequencies that are not in the band of interest.
- For speech, the filter is down 3 dB at 3400Hz and is required to have at least 14 dB of attenuation at 4 KHz.



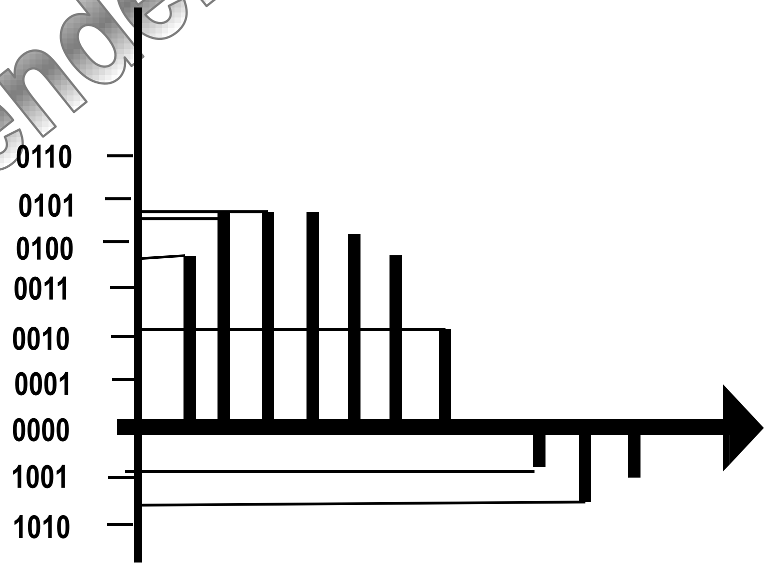
PCM

- **Looking back to the PAM example, let's assume we have the modulated pulse train and we divide the Y axis into some number of equal divisions and assign binary numbers to each point on the Y axis.**



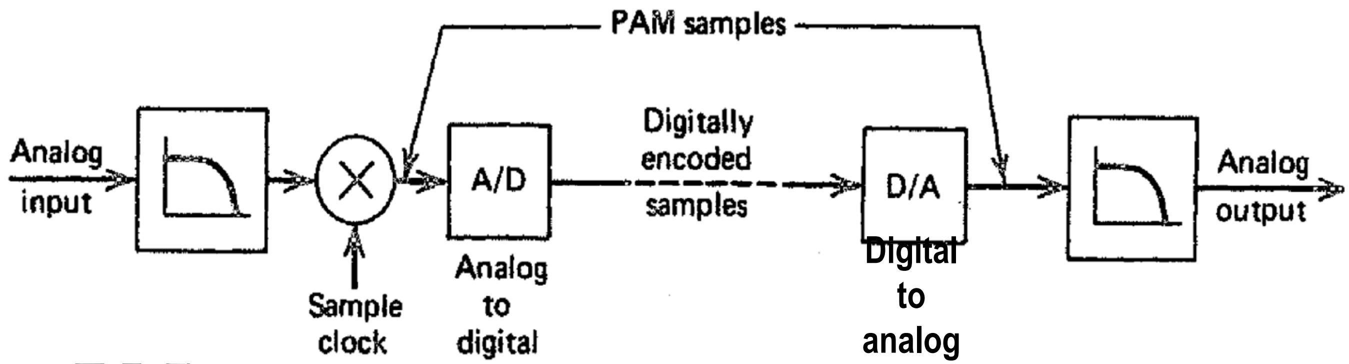
PCM

- Then, we can take each pulse and compare its amplitude to the Y axis marks, and assign to each pulse the number that is closest to the amplitude.
- This is Pulse Code Modulation. We took PAM and represented the amplitude of each pulse by a binary number.



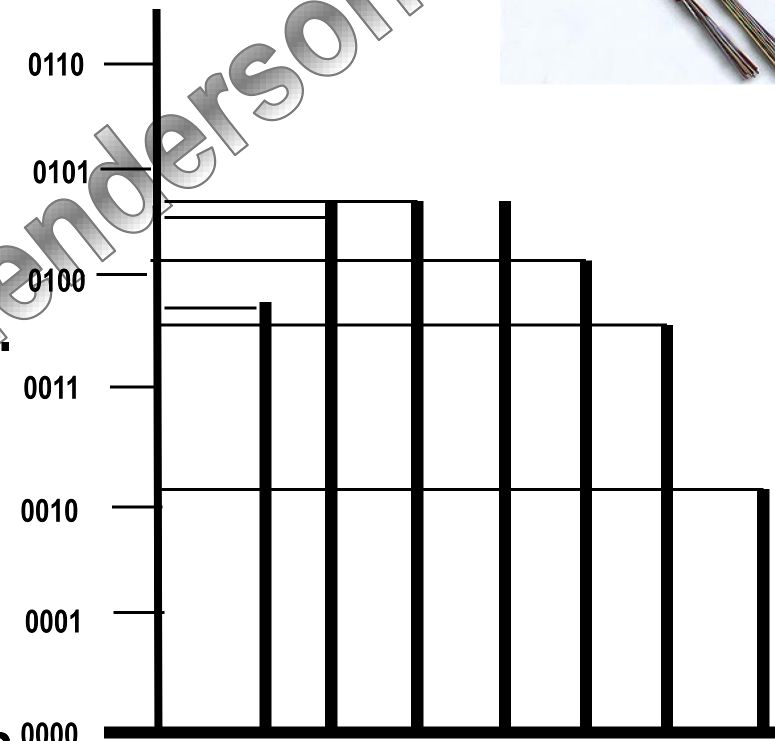
PCM

- **We can convert a PAM speech encoder to a PCM speech encoder by including a codec at both ends. These are shown as an A/D on the transmit end and as a D/A on the receive end.**



Quantization Noise (or error)

- Note that few of the amplitudes line up exactly with the points on the Y axis.
- The quantizer will assign the binary code that is closest to the amplitude.
- The difference between the actual amplitude and the code assigned is called "quantization error" or "quantization noise".
- It's called noise because the effect is exactly the same as if noise affected the pulse in the analog domain.



PCM

- **In the previous diagram, I showed the quantization points as being linear on the Y axis.**
- **In PCM they are not linear – they are closer together close to the X axis and wider apart in the higher levels of amplitude. Why do that?**
- **One reason is to partially equalize the Signal to Quantization Noise (SQR).**
  - Quantization noise can be half an interval.
  - For small signals, this leads to a high SQR. It's much lower for high amplitude signals.
  - With unequal intervals, the SQR can be made approximately equal over the amplitude range.

PCM

- **There are two “standards” for digitizing voice in the telephone network**
  - µ-law encoding which is used in North America.
  - A-law encoding which is used in most of the rest of the world
  - Encoding is defined in the ITU Recommendation G.711
- **When a call is made between a µ-law area and an A-law area, A-law is used for the call.**

μ-law PCM

- **Each voice sample is 8 bits. For the μ-law quantization, the bits are allocated as follows: (A-law is similar)**
- **The first bit indicates the polarity: 1 for positive and 0 for negative.**
- **The next 3 bits indicate the segment number, 8 segments above the x-axis and 8 segments below. The two segments just above and below the x-axis are co-linear so there are 15 total segments**
- **The last four bits indicate the quantized level within a segment.**
- **Since most sounds are of smaller amplitude, most of the samples would have segment numbers with lots of zeroes. To get good ones-density, the codewords are inverted before transmission (A-law inverts every other bit).**

VOIP

- Speech is usually encoded with a “speech coder” when it will be communicated via VOIP.
- A speech coder can generally encode speech at a much lower bit rate than 64Kbps.
- All “vocoders” introduce delay because they have to buffer a certain amount of voice information before processing it.
- Waveform encoders do not introduce as much delay.
- Some examples of speech coders are G.726 (32Kbps waveform encoder), G.728 (16Kbps vocoder), and G.729 (8Kbps vocoder)

Transporting Speech in the Network

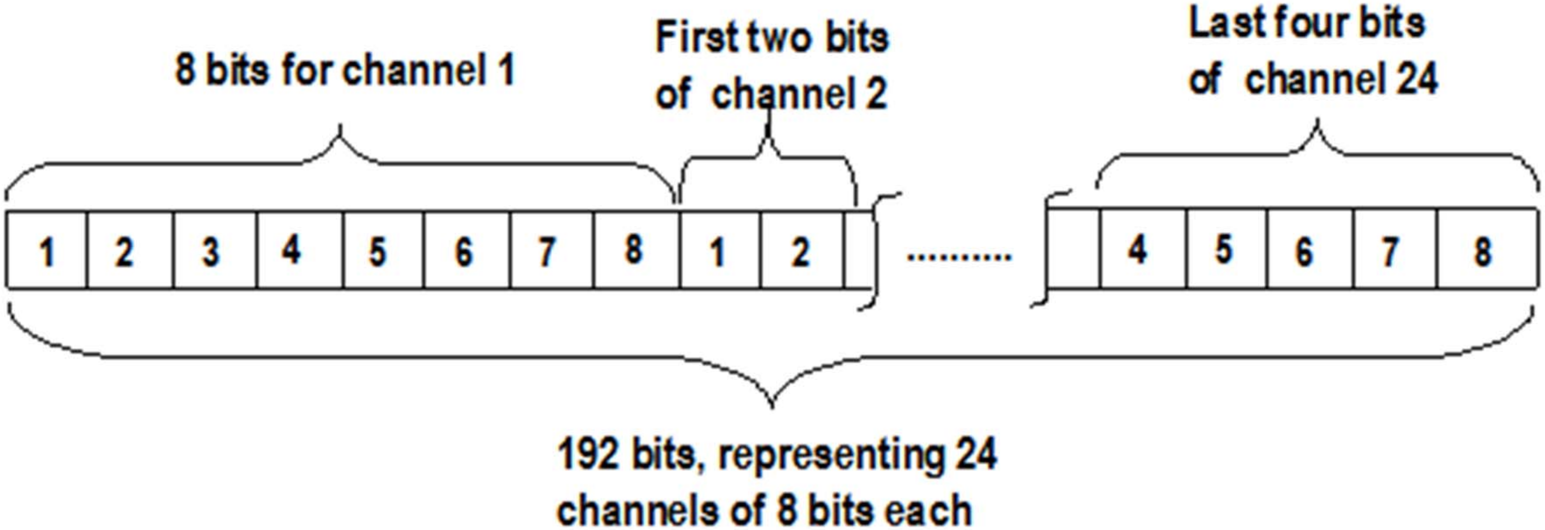
- In the US network, speech samples are transmitted in a DS1 signal.
- A DS1 carries 24 voice channels and the voice samples are byte mulitplexed.

DS1

- The DS1/T1 technology was developed at Bell Labs and introduced in 1962 to carry voice channels.
- The 1.544Mbps signal can be referenced as either a DS1 or a T1. What's the difference?
- Normally, when working with the digital signal prior to putting it on the transmission line, it's referred to as Digital Signal 1 or DS1.
- To transmit a DS1, a line code must be used (Alternate Mark Inversion or AMI). Once the signal is in the AMI domain, it is referred to as T1.
- But in the US, a 1.544Mbps line is usually referred to as a T1 line.

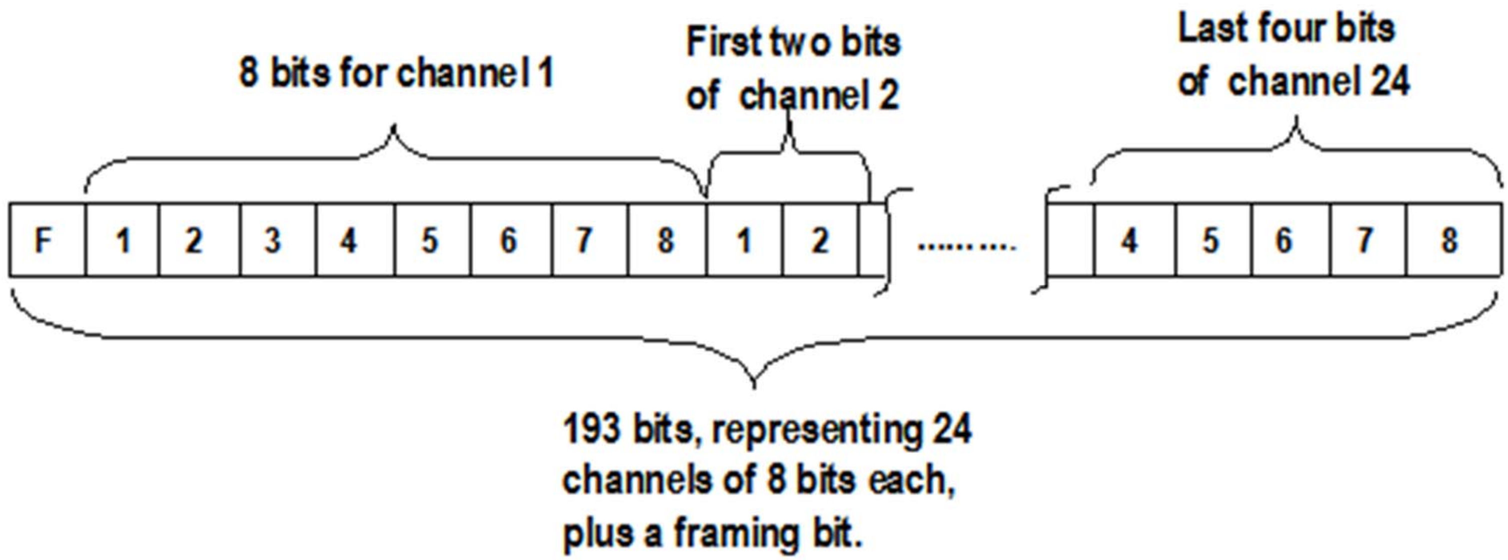
DS1

- **The basic structure of the DS1 signal is shown in the picture.**
- **8 bits from each of 24 voice channels are put into the bit stream. All 8 bits from each channel are together. The channels are octet multiplexed, not bit multiplexed.**



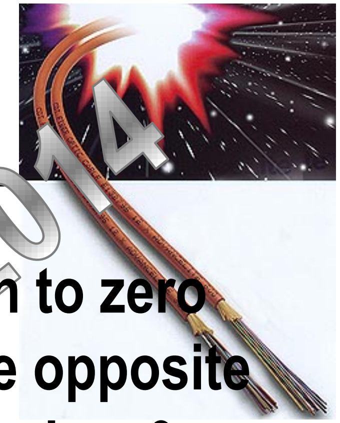
DS1

- **But we can't just send that stream of bits. We have to have some way to find the start of the block so that we can recover the octets of the voice channels.**
- **This is done by the addition of a Framing bit at the front of the frame. So a DS1 frame is 193 bits.**
- **We'll talk more about framing later.**



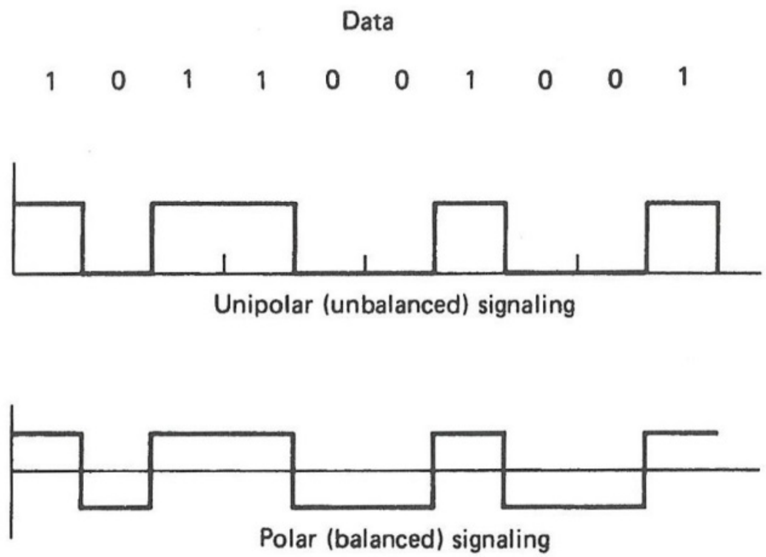
Line codes

- Now that we understand how voice is digitized and that it's put in a frame for transmission, let's discuss how it's transmitted.
- Information is communicated across metallic media through differential signaling. That is, a varying voltage is placed across the two conductors.
- One requirement is that any signal be DC balanced, that the positive and negative pulses are balanced.
- Let's look at some line codes.



NRZ (Non-Return to Zero)

- **Non-return to zero means that the signal does not return to zero after indicating an information bit. It only returns for the opposite information bit. Usually a voltage is a 1 and zero voltage is a 0.**
- **NRZ can be unipolar, which means it represents bits with a voltage and zero voltage.**
- **Polar means that it represents bits by opposite voltages**

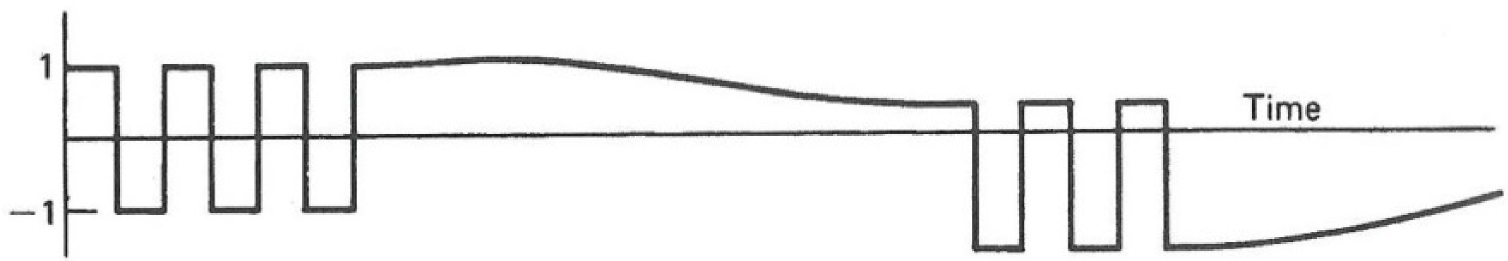


Comment on Polar Encoding

- **Polar encodes one bit as a "positive" voltage level and the other as a "negative" voltage level.**
- **But positive and negative are relative so it creates ambiguity for the receiver.**
- **It's possible for the receiver to receive all zeroes as ones, and all ones as zeroes.**
- **So polar is rarely used. One bit is generally sent as a change of state, while the other bit is sent as no change of state. Usually, a change of state is a 1 and no change of state is a 0.**

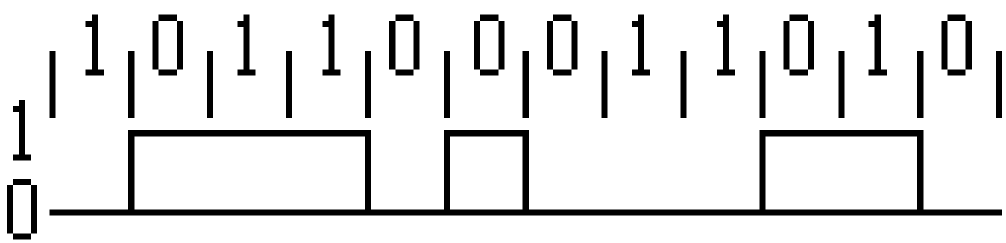
NRZ

- NRZ is not DC balanced and is subject to DC wander if long strings of 1s or 0s are sent.
- In the figure, a long string of 1s are being sent.
- This can also lead to receive clock problems.



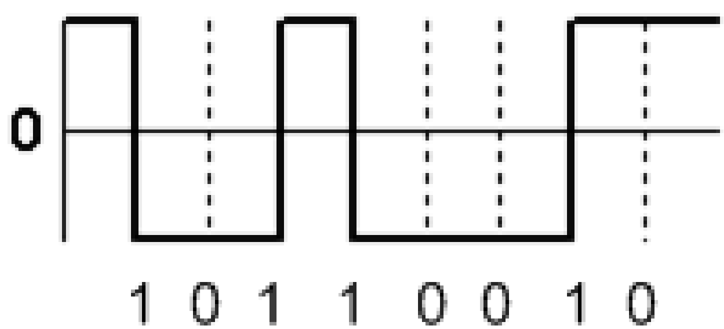
Non-Return to Zero Space

- The signal transitions whenever there's a 0.
- You can see in the diagram that a voltage change occurs whenever there's a 0 bit.
- Whenever there's a 1 bit, the voltage stays the same as it was for the previous bit.



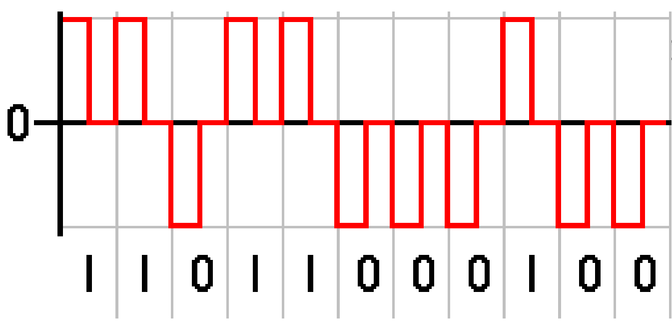
Non-Return to Zero Inverted

- Non-Return to Zero Inverted (NRZI) is the opposite of Non-Return to Zero Space.
- You can see in the diagram that a voltage change occurs whenever there's a 1 bit.
- Whenever there's a 0 bit, the voltage stays the same as it was for the previous bit.

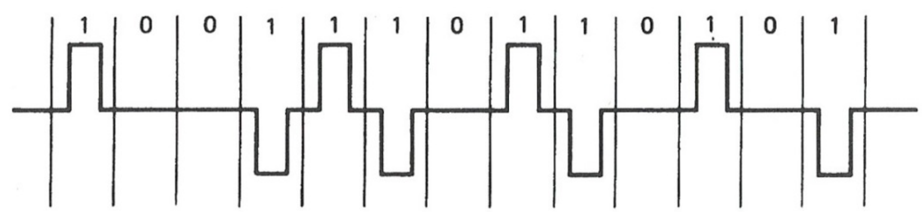


Return to Zero (RZ)

- Return to Zero may be polar, as shown in the diagram, with a pulse in one direction indicating one bit and a pulse in the other direction indicating the other bit.
- Or, it could be bipolar, with one bit indicated by a + or – voltage and the other bit indicated by a 0 voltage.
- Polar is not DC balanced. Bipolar is DC balanced but long strings of zeroes could cause loss of timing.



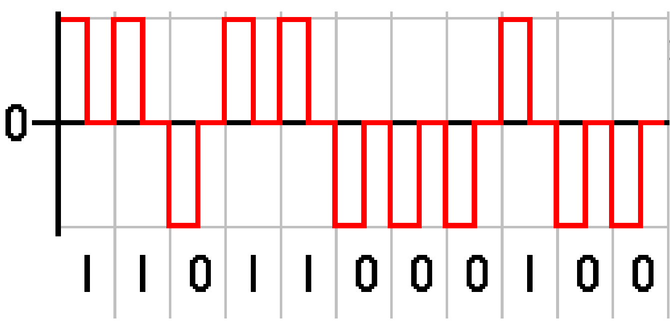
Polar RZ



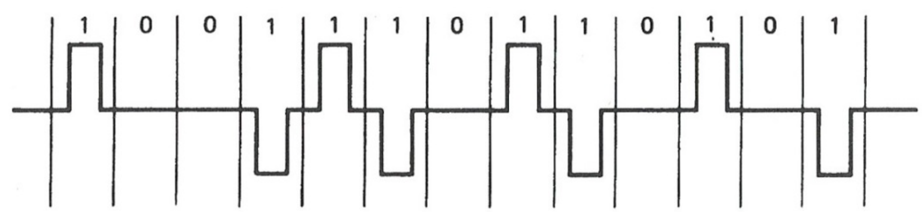
Bipolar RZ, also known as AMI

Return to Zero

- Note that each pulse must be less than the full bit time in order to give the voltage time to return to zero.



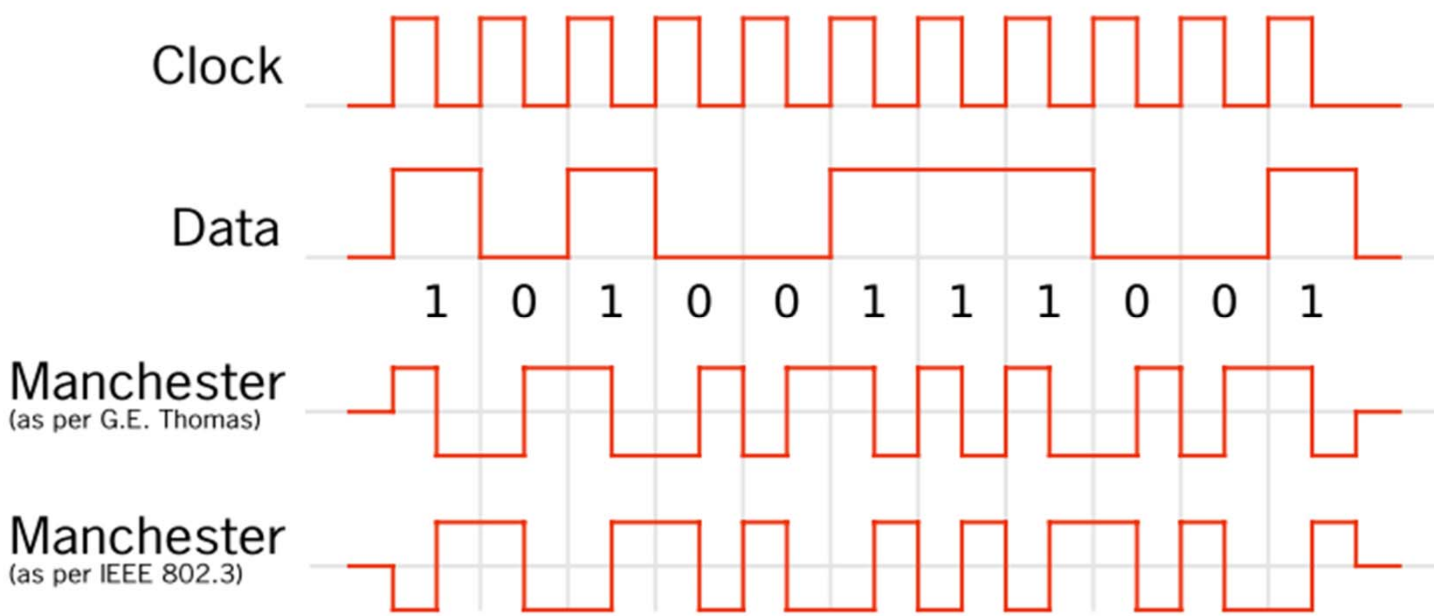
Polar RZ



Bipolar RZ, also known as AMI

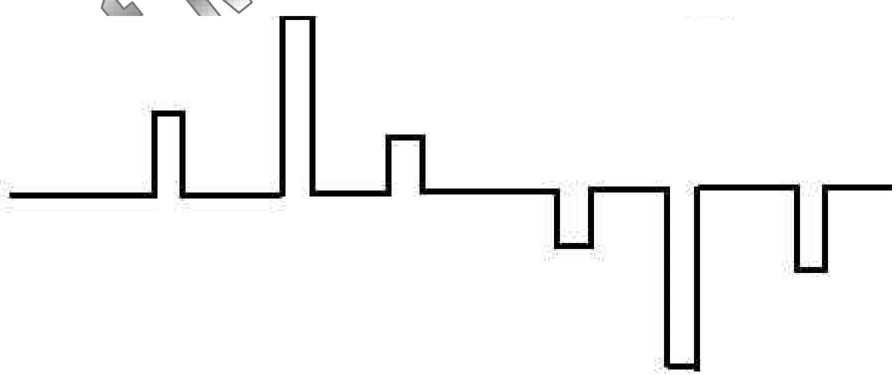
Manchester Coding

- **Manchester coding is a type of phase encoding.**
- **Each bit has at least one transition, and it is DC balanced.**
- **A Manchester code insures frequent line voltage transitions and thus provides good clocking.**
- **But it takes a lot of bandwidth.**



Pulse Amplitude Modulation (PAM)

- **Pulse Amplitude Modulation (PAM) is used for more than just encoding voice.**
- **It's used as a line code in a number of applications**
  - 56Kbps modems.
  - 1000BASE-T Ethernet (5 level PAM – 2 bits per level)
  - 10GBASE-T Ethernet (16 level PAM – 4 bits per level)
- **Bandwidth is equal to half the symbol rate – it takes a minimum of two pulses to equal 1 Hz.**



Pulse Transmission

- By "pulse transmission" I mean all the previously described line codes.
- Note that by Nyquist, we cannot transmit pulses faster than twice the bandwidth of the transmission medium.
- But we can't just keep increasing the bandwidth.
- Some channels, such as a microwave channel, may be limited by license.
- Other channels may be limited by the characteristics of the channel, such as twisted pair.

What we want in a line code

- As small a bandwidth as possible.
- DC balance.
- Power efficient.
- Error detection and correction.
- Adequate timing content so we can recover and maintain a receive clock.
- Prevent long strings of 1s or 0s. This can be accomplished at a higher level so it's low priority for the line code.

Bipolar RZ

- **Bipolar has much to recommend it:**
  - It has no DC component.
  - It has reasonable bandwidth.
  - It has a form of error detection. If two consecutive pulses have the same polarity, there was an error on the line. However, this function was used for another purpose in AMI.
- **But some problems, also**
  - It requires more power (3 dB) than unipolar.
  - It requires that there not be a long sequence of 0s or timing can be lost. A long string of 1s provides good timing content since there will be a transition each bit.

Solution: Binary N Zero Substitution

- Binary N Zero Substitution (BNZS) will prevent long strings of 0s or 1s. (N is a number, such as 3).
- B3ZS is used on T3 lines and I'll describe that first.
- Each string of three 0s is coded according to the table below, depending on the number of "normal" alternations prior to the three 0s. Note that the substitution introduces a violation.
- Remember that the count is from a violation so this substitution takes into account the last violation.

Polarity of preceding pulse	Number of bipolar pulses since the last violation	
	Odd	Even
Minus	00– **00V**	+0+ **N0V**
Plus	00+ **00V**	–0– **N0V**

N = Normal alternation
V = Violation

More on B3ZS

- Let’s look a bit closer at the substitution table.
- The “odd” or “even” is from the last violation. So the last violation left a disparity, either + or -.
- Assume three pulses before the 000 and the last one was a minus. If an odd number, and the last one was minus, the last violation left a + disparity.
- The three pulses since the violation cancelled out the disparity (+V, –, +, – = 0 disparity). After sending 00– we will be left with a negative disparity and the next pulse will be +.
- The – in 00– will be a violation.

Polarity of preceding pulse	Number of bipolar pulses since the last violation	
	Odd	Even
Minus	00– **00V**	+0+ **N0V**
Plus	00+ **00V**	–0– **N0V**

N = Normal alternation
V = Violation

More on B3ZS

- Now, let’s look at the situation were there were an even number of pulses since the last violation and the last one was a minus.
- Assume four pulses before the 000. If an even number, and the last one was minus, the last violation left a – disparity.
- The four pulses since the violation left a – disparity (-V, +, –, +, – = – disparity). After sending +0+ we will be left with a positive disparity. The second + will be a violation.

Polarity of preceding pulse	Number of bipolar pulses since the last violation	
	Odd	Even
Minus	00– **00V**	+0+ **N0V**
Plus	00+ **00V**	–0– **N0V**

N = Normal alternation
V = Violation

More on B3ZS

- **Working out the situation when the last pulse was a positive will be left to the student.**
- **Hint: It's the same as explained earlier, with the polarity reversed.**
- **Since positive and negative is ambiguous to the receiver, voltage assignments could go either way.**

Polarity of preceding pulse	**Number of bipolar pulses since the last violation**	
	Odd	**Even**
Minus	00– **00V**	+0+ **N0V**
Plus	00+ **00V**	–0– **N0V**

N = Normal alternation
V = Violation

Decoding B3ZS

- The receiver should never receive three sequential zeroes. If it does, a bit error has occurred.
- Otherwise, when a bipolar violation occurs, the receiver can use the table to determine that the bit string is a 000. Any one of those four strings will resolve to 000.

B8ZS

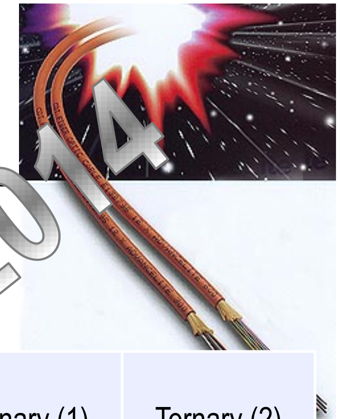
- B8ZS is used on North American T1 lines. It's actually simpler than B3ZS, and I'll describe it now.
- The substitution is done for a sequence of eight zeroes and is based on the preceding pulse as shown in the table below.
- Both substitutions have zero disparity so the only thing that makes a difference is that the first pulse after the 0 has to be a violation.
- Note that each substitution will have two violations, the one after the three 0s and one inside the string.

Polarity of preceding pulse	Substitution	N = Normal alternation V = Violation
Minus	000 – + 0 + –	000VN0VN
Plus	000 + – 0 – +	000VN0VN

4B3T Line Code (4 Binary, 3 Ternary)

- **Note that we have three voltage levels on the line, +, 0, and –.**
- **For a sequence of three signals, we have 27 combinations – 3*3*3 = 27.**
- **Four bits can have only 16 combinations – 2*2*2*2 = 16**
- **Therefore, we can map four bits into three voltage levels – 4 binary bits into 3 signals.**
- **This provides a more efficient transport of information than bipolar, for example, where one bit is represented by one voltage level.**
- **This also introduces the concept of "symbol" which we'll see more of when we study modems. A combination of three voltage levels is a symbol that represents 4 binary bits.**

4B3T Line Code



- Let’s begin by looking at the assignment of symbols to bits for European ISDN BRI (US uses 2B1Q).
- The specific symbol chosen for a particular set of four binary bits depends on the disparity.
- If the disparity is +, choose from the first column. If negative, choose from the second column. The maximum disparity is 3.
- The symbols from 1010 to 1111 are balanced.
- 26 of the possible 27 symbols are used. Which one is missing?

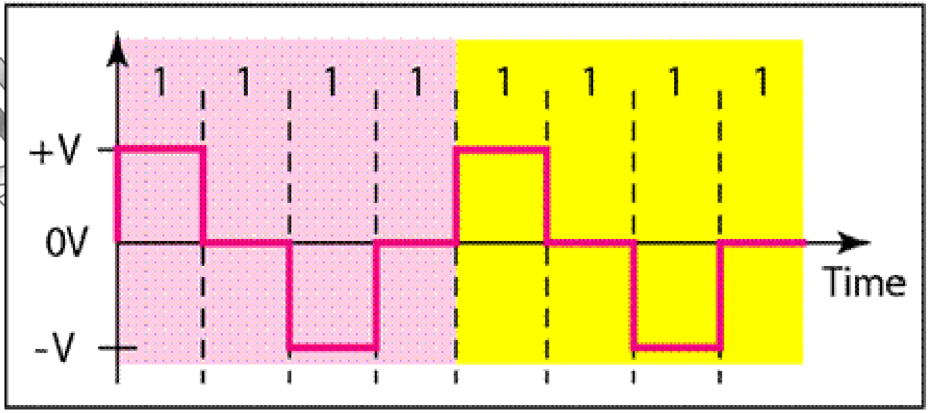
Binary	Ternary (1)	Ternary (2)
0000	- - -	+ + +
0001	- - 0	+ + 0
0010	0 - +	- 0 +
0011	0 - -	0 + +
0100	- - +	+ + -
0101	- + -	+ - +
0110	+ - -	- + +
0111	- 0 0	+ 0 0
1000	0 - 0	0 + 0
1001	0 0 -	0 0 +
1010	0 + -	
1011	0 - +	
1100	+ 0 -	
1101	- 0 +	
1110	+ - 0	
1111	- + 0	

4B3T Line Code

- **Why use 4B3T?**
- **Advantages**
  - Lower bandwidth.
  - Strong timing content (lots of transitions).
- **Disadvantages**
  - Loss of error detection.
  - Higher bandwidth than 2B1Q used in the US.

Multi-Level Threshold -3 (MLT-3)

- **Used in Ethernet 100BASE-TX.**
- **Three voltage levels, perhaps +1, 0, and -1.**
- **Changes state sequentially to indicate a 1 bit and stays in the same state to indicate a 0 bit.**
- **So a sequence of 0010111000 would be coded as +1, +1, 0, 0, -1, 0, +1, +1, +1, +1.**
- **Gives four bits per Hz, so bandwidth is one quarter of bit rate.**
- **100BASE-TX transmits at 125Mbps so bandwidth is max of 31.25MHz.**

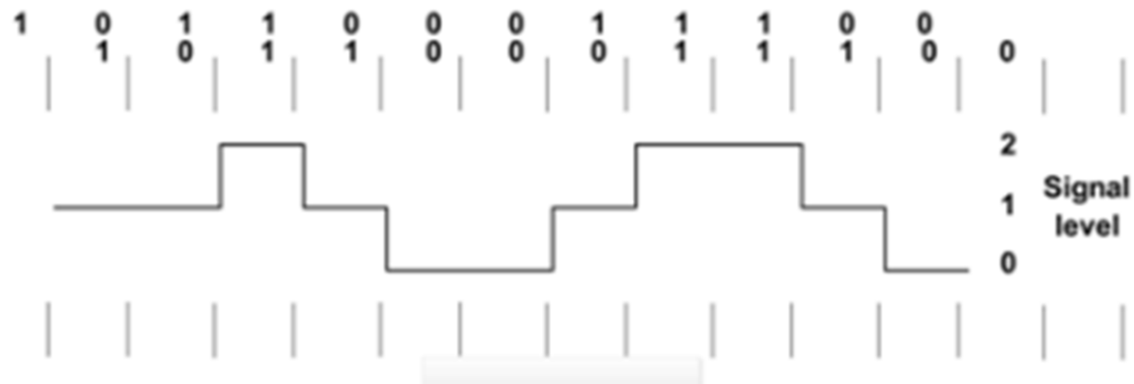


Duobinary

- An interesting line code, with perhaps more applicability to optical fiber than metallic cable.
- Requires three symbols, such as three voltage levels.
- Each level represents the bit being sent, as well as the previous bit sent.

Coding rules:

0+0=0
0+1=1
1+0=1
1+1=2



Duobinary

- **The 1 level voltage is ambiguous – it could be a 0 or a 1 and is evaluated based on the previous bit sent.**
  - If the previous bit sent was a 0, it represents a 1.
  - If the previous bit sent was a 1, it represents a 0
- **A 0 level always represents two 0s sent sequentially.**
- **A 2 level always represents two 1s sent sequentially.**

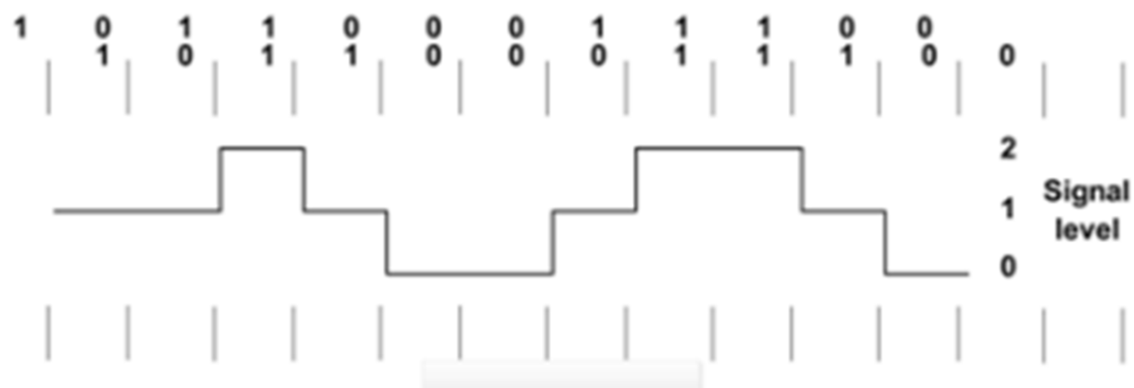
Coding rules:

0+0=0

0+1=1

1+0=1

1+1=2

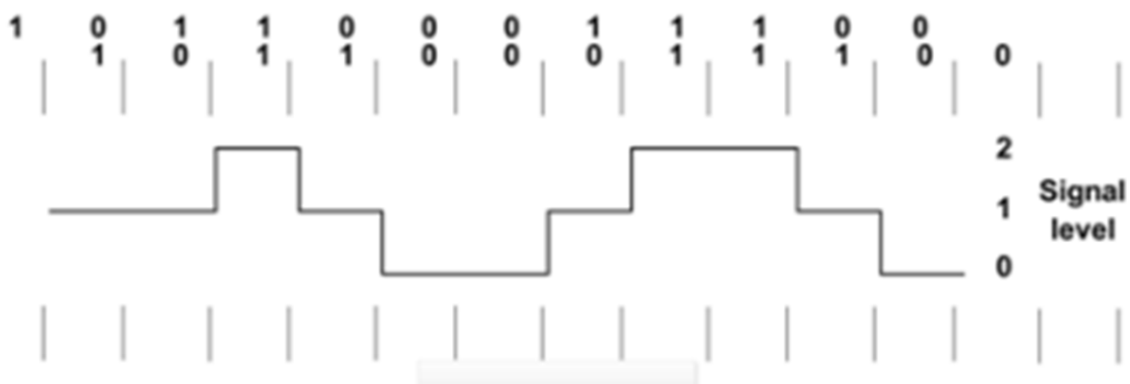


Duobinary

- **A bit error will cause multiple errors, but the string of errors will be limited in length.**
  - A 0 level says that the previous bit and this bit is 0.
  - A 2 level says that previous bit and this bit is 1.
  - So receiving a correct 0 or 2 level will reset the bit stream.
- **Problem: A bit string of 0101010101… will produce a single voltage with no transitions, as will long strings of 1s or 0s. Countered by pre-coding.**

Coding rules:

0+0=0
0+1=1
1+0=1
1+1=2



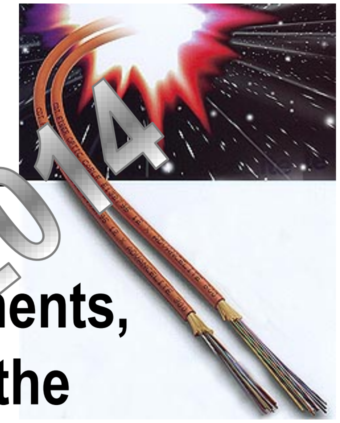
Transmission Media

- **Now that we've talked about line codes, let's examine some of the transmission media.**
- **Some of the transmission media used in the network are:**
  - Twisted pair
  - Coaxial cable – not really used a great deal any more.
  - Terrestrial microwave
  - Optical fiber

Twisted Pair

- **Twisted pair is used in the local loop, and for the most common versions of Ethernet.**
- **The local loop is considered Category 1 (Cat-1) cable, while Ethernet requires a better cable, perhaps Cat-5 or Cat-5e.**
- **Let's look at the characteristics of twisted pair.**
  - A lot of the characteristics described are only applicable to twisted pair used for Ethernet, but an understanding of them will help you understand any twisted pair application.
  - Also, Ethernet cable is limited to 100 meters, while the local loop can be kilometers in length.

Twisted Pair - Attenuation



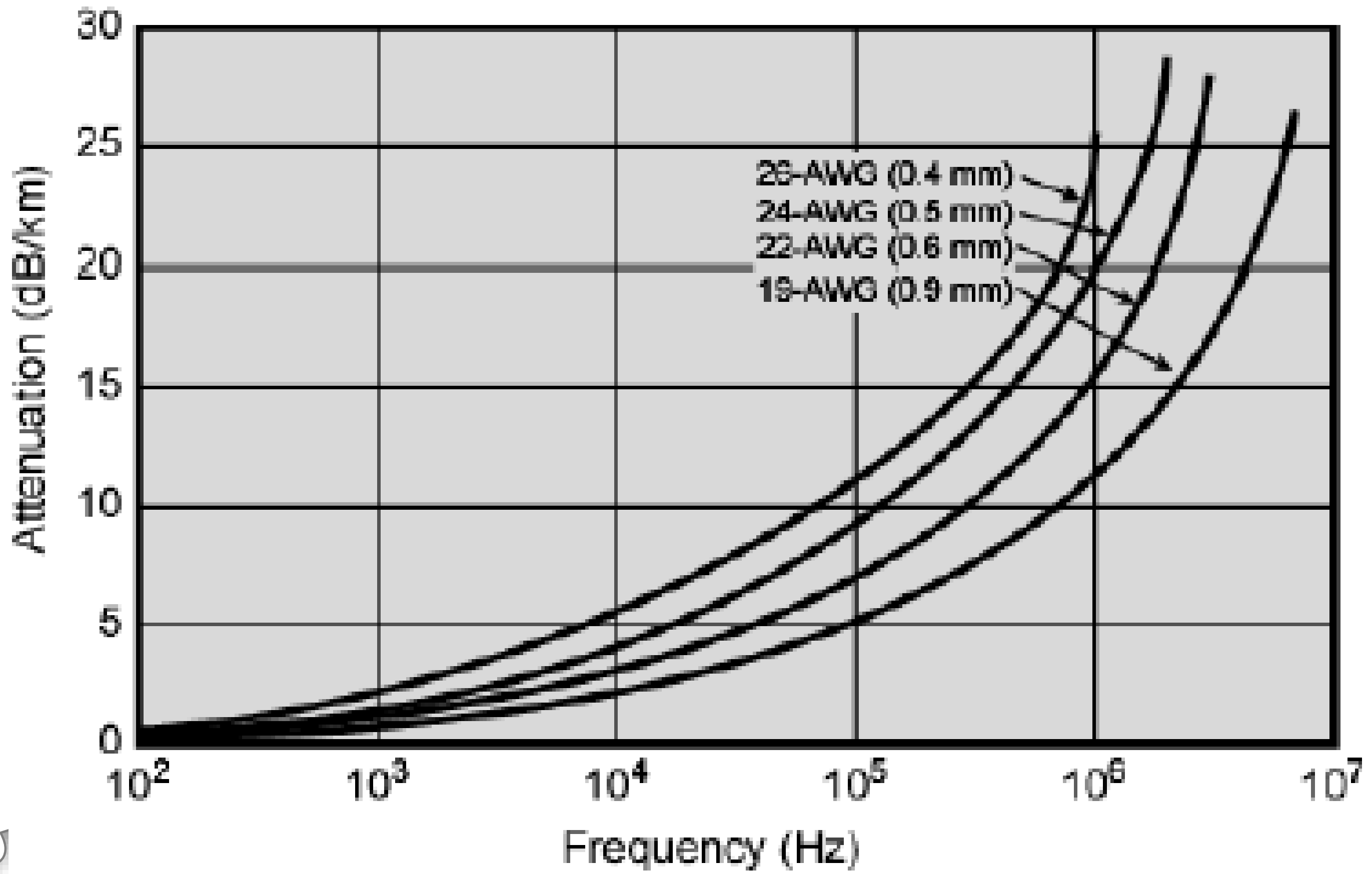
- Attenuation, called Insertion Loss in standards documents, is the loss of signal power as the signal travels down the wires.
- One component of attenuation is due to the resistance of the wire. But the wire can be made larger to reduce that component. And yet, we still have significant attenuation in twisted pair. There must be another component of attenuation.
- That component of attenuation is frequency dependent. That is, there's more attenuation at higher frequencies than at lower frequencies. Why is that?

Twisted Pair - Attenuation

- When a signal travels down a wire, it creates a magnetic field that varies at the frequency in question.
- Remember how a transformer works – The fluctuating magnetic field from one winding cuts the copper conductors of the other winding and induces a voltage in that winding.
- The same thing happens in twisted pair. The magnetic field around each wire induces a voltage in any conductor that it is (mostly) parallel to. That takes energy from the signal. It attenuates the signal.

Attenuation

- **The figure below shows attenuation in Category-1 (subscriber loop) twisted pair.**



Twisted Pair - Crosstalk

- Crosstalk is coupling of signals from one set of twisted pair to another set of twisted pair, usually by capacitive or inductive coupling.
- There is two types of crosstalk, Near End Cross Talk (NEXT) and Far End Cross Talk (FEXT).
- NEXT is the most serious.

Near End Cross Talk (NEXT)

- NEXT is the signal coupled to the other pairs in the bundle, and measured at the same end as the transmitter.
- Most crosstalk is coupled close to the transmitter end because the transmit signal is strongest there.
- It continues all along the cable but is reduced away from the transmitter because the signal is reduced.

Far End Cross Talk (FEXT)

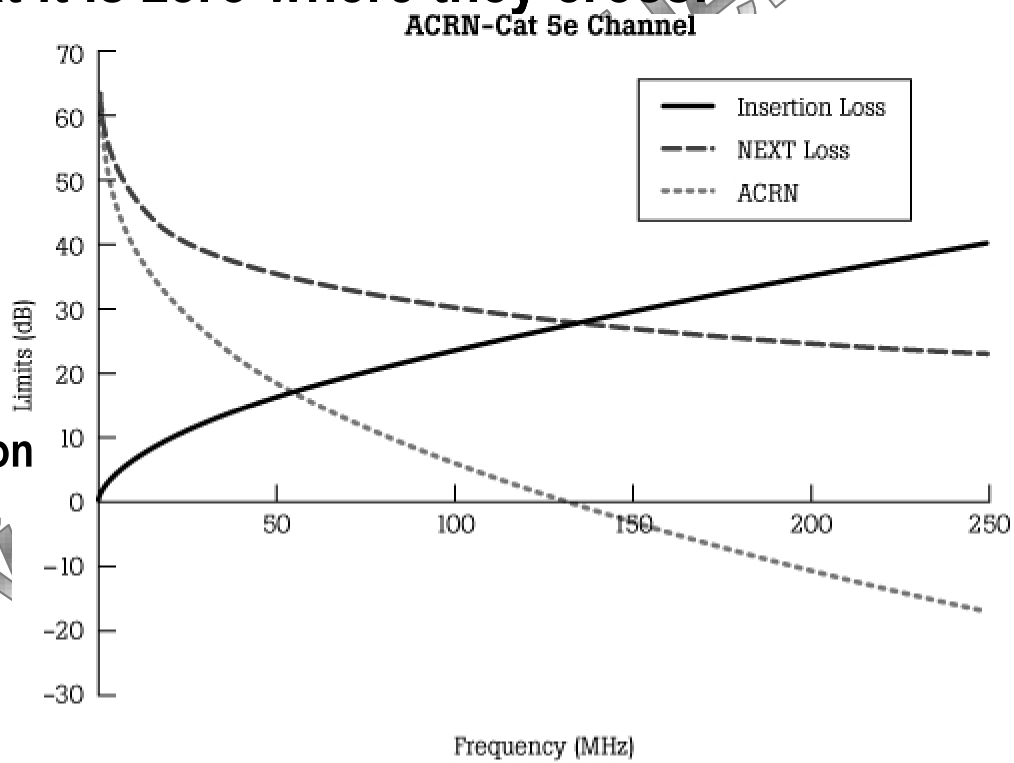
- **FEXT is the signal coupled to the other pairs in the bundle, and measured at the opposite end from the transmitter.**
- **FEXT is not as serious as NEXT because the FEXT is attenuated across the cable.**

Bandwidth of Twisted Pair

- In twisted pair, bandwidth is usually spoken of as the frequency where the signal induced by NEXT is as strong as the remaining signal transmitted from the other end of the cable, for the maximum length of the cable.
- The Attenuation to Cross Talk Ratio is the difference, in dB, between the attenuation and the crosstalk.
- See the figure on the next slide for Cat-5e which is specified as having 125MHz of bandwidth.

Bandwidth and ACR of Cat-5e

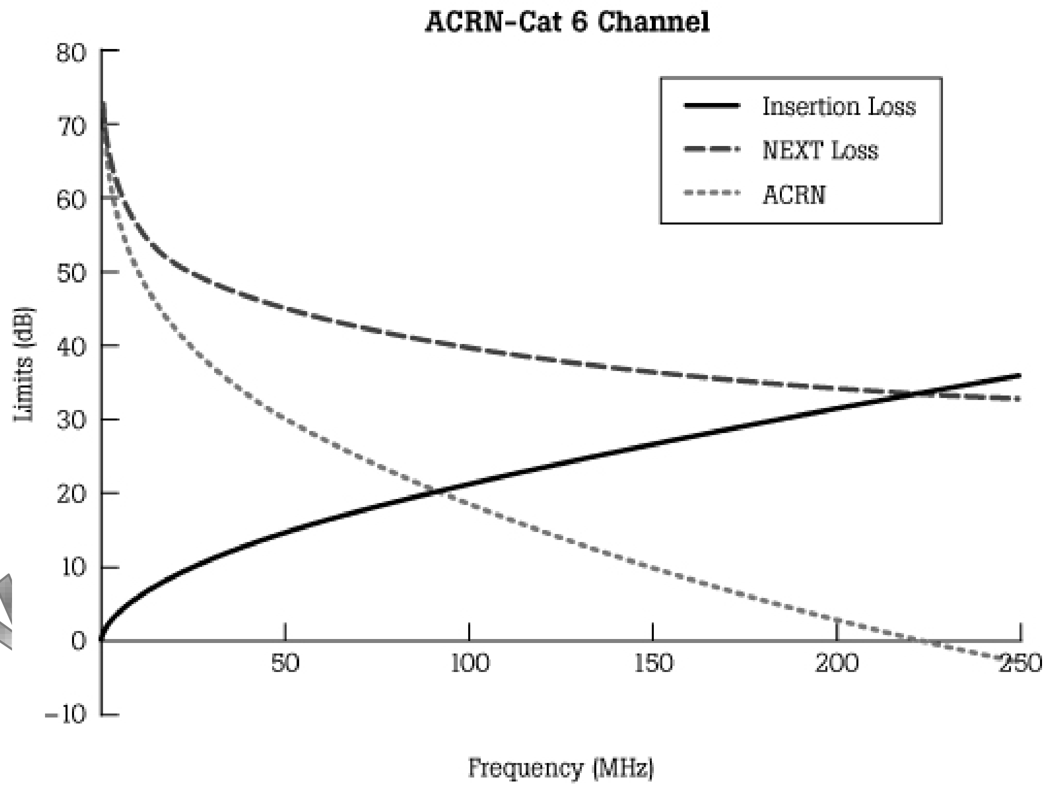
- **Note that the attenuation and NEXT cross close to 125MHz.**
- **There is more than NEXT that affects the bandwidth so the actual bandwidth is less than shown.**
- **ACRN is the difference, in dB, between Attenuation and NEXT. Note that it is zero where they cross.**



ACRN= attenuation to crosstalk ratio, near end

Category 6 cable

- **Category 6 cable gives a slight improvement in attenuation but a significant gain in NEXT. This provides a bandwidth of about 200MHz.**



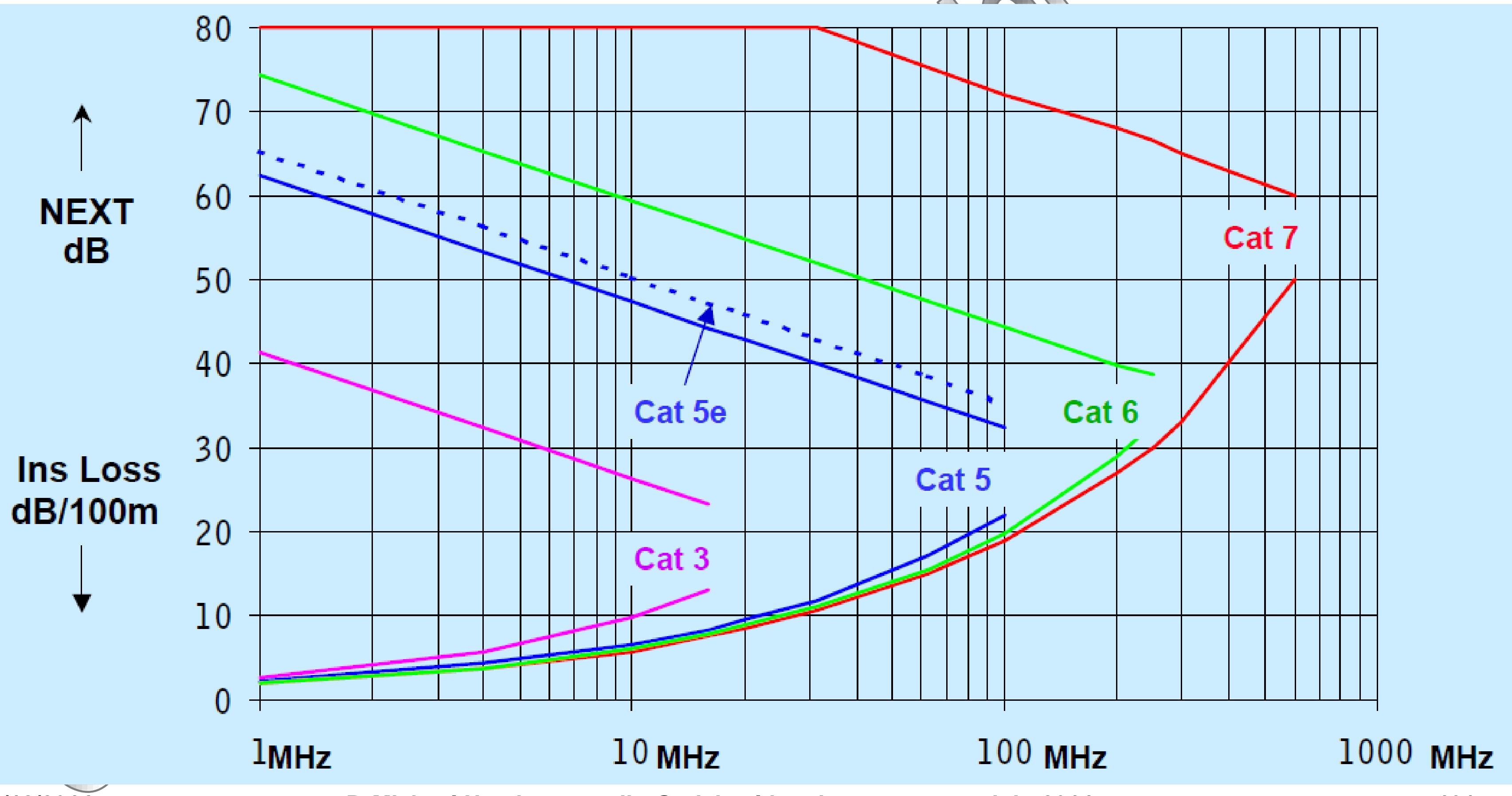
ACR

- **Here's another look at ACR for Cat-5 and Cat-6 cable, which includes all the disturbers on the cable.**
- **PS-ACR = PowerSum Attenuation to Cross Talk Ratio. PowerSum takes into account the crosstalk from all sources (all pairs in the bundle).**

Frequency (MHz)	Category 5e Channel PS-ACR (dB)	Category 6 Channel PS-ACR (dB)
1.0	54.8	59.9
4.0	46.0	56.5
8.0	39.3	49.9
10.0	36.9	47.7
16.0	31.5	42.6
20.0	28.8	40.0
25.0	25.9	37.2
31.25	22.8	34.3
62.5	12.0	24.1
100.0	3.1	15.8
200.0	—	0.4
250.0	—	-5.7

Twisted Pair Cable

- **Here's a chart that shows attenuation (insertion loss) and NEXT loss for Cat-3, 5, 5e, 6 and 7 twisted pair cable.**



Twists in Twisted Pair

- This brings up the question of how often is the wire twisted?
- A table for Cat-5 is shown below.
- Note that the number of twist are different between pair.
- The Cat-1 wire used in the local loop may have seven to nine twists per meter.

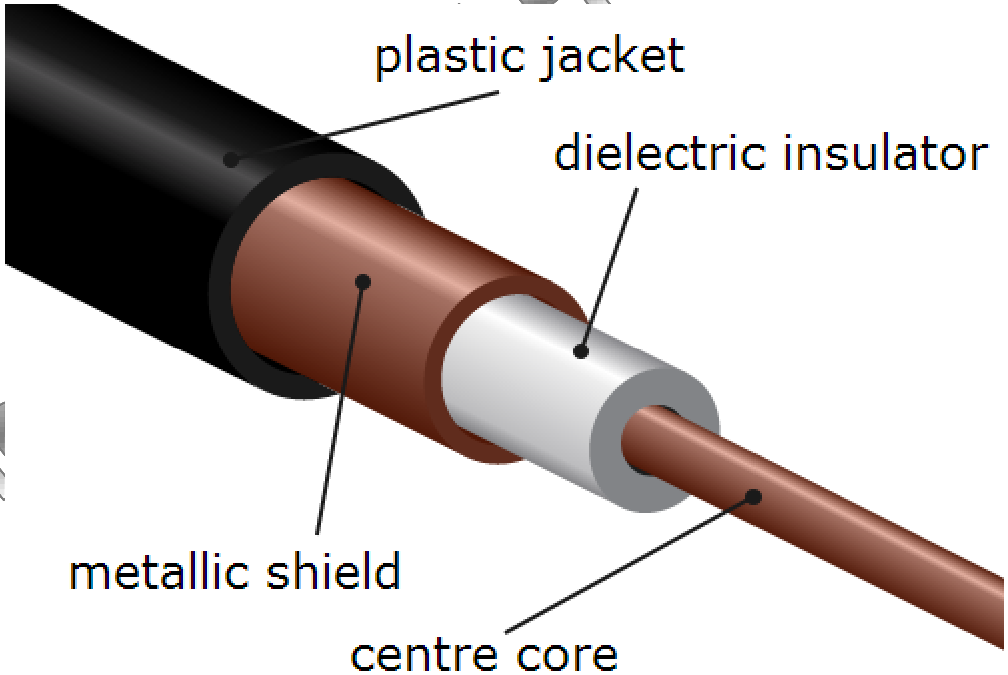
	Pair color	[cm] per turn	Turns per [m]
	Green	1.53	65.2
	Blue	1.54	64.8
	Orange	1.78	56.2
	Brown	1.94	51.7

Making Twisted Pair with Wider Bandwidth

- **Some of the things that are done to make cable with a wider bandwidth are:**
  - Bigger wire to reduce attenuation.
  - Thicker insulation to keep the wires further apart.
  - More twists per meter to reduce crosstalk.
  - Better control of impedance to reduce reflection.
  - (Sometimes) cross splines to keep wires apart and keep the cable round.
  - Better connectors.

Coaxial Cable

- Coaxial cable is a type of cable that contains a center conductor, surrounded by an insulator and a circular metallic shield.
- In a perfect coax cable, the electric field exists only in the space between the center conductor and the shield.



Coaxial Cable

- Coax is primarily used to carry RF signals.
- Coax is not used very much in the network today but was used at one time to carry many simultaneous telephone calls.
- It's still used in CATV systems, although mostly to connect to the residence and for distribution within the residence.
- We won't go any deeper into coax cable here.

Terrestrial Microwave

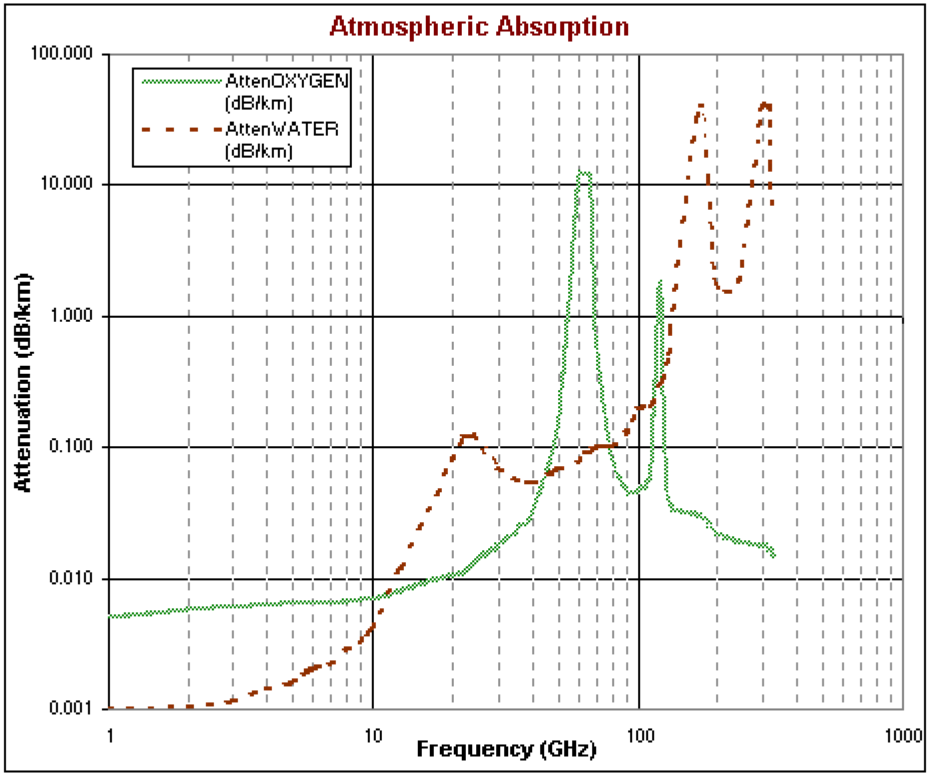
- There is disagreement as to what frequencies should be termed “microwaves”. Some sources quote 300MHz to 300GHz, while others quote 1 to 30GHz or 2 to 40GHz.
- The 300MHz to 300GHz gives wavelengths from one meter to one millimeter.
- The wide bandwidth assigned to each channel allows for a very high information capacity.
- Early in the telephone network, microwave was used to carry many simultaneous analog voice calls, similar to coax.
- It was (and still is) widely used because it could be deployed quicker and at less cost than laying coax (or fiber, today).
- Today it’s almost all digital microwave.

Terrestrial Microwave

- **Terrestrial microwave is "line of sight" meaning that the two stations have to be able to "see" each other. They cannot be below the horizon.**
  - Limits transmission distance
- **Microwave signals can also be impacted by weather conditions.**
  - "Ducting" due to atmospheric layers.
  - Rain fade. (also snow)
- **Compensated for by frequency diversity and/or space diversity.**

Terrestrial Microwave

- Frequencies are limited by absorption by water in the atmosphere and by oxygen.
- The first attenuation peak occurs at 22 GHz due to water, and the second at 63 GHz due to oxygen.



Optical Fiber

History of Optical Fiber

- In 1965 Charles K. Kao (Nobel prize, 2009) and George Hockham showed that attenuation in glass was primarily due to impurities in the material, opening the way to development of practical fiber.
- In 1970 Robert Maurer, Donald Keck, and Peter Schultz of Corning produced a fiber with 17 dB/km in the 633nm window, opening the way for practical fiber.
- By 1976, fiber loss had been reduced to less than 1 dB/km at 1310 nm, paving the way for the first commercial installation in Chicago in 1977 (operating at all of 45 Mbps).

Basic Principles

- Optical fiber is a dielectric waveguide, not a conductor.
- The fiber is made of silica (glass) doped to modify the refractive index.
- It consists of a center core, surrounded by what's called the cladding. The core has a slightly higher index of refraction than the cladding. This is necessary to produce the waveguide.
- Light is guided by "total internal reflection" at the boundary between the core and the cladding.
- The refractive index of glass used in fiber optic cable is between 1.45 and 1.48. The difference between the refractive index of the cladding and the core is small – about 0.4%.

Basic Principles

- The speed of light in a medium is related to the refractive index of the medium.
- In the glass used in fiber optic cable, light travels about 0.69 c
- This is close to the speed of electricity in copper.

$$v = \frac{c}{n}$$

Where:

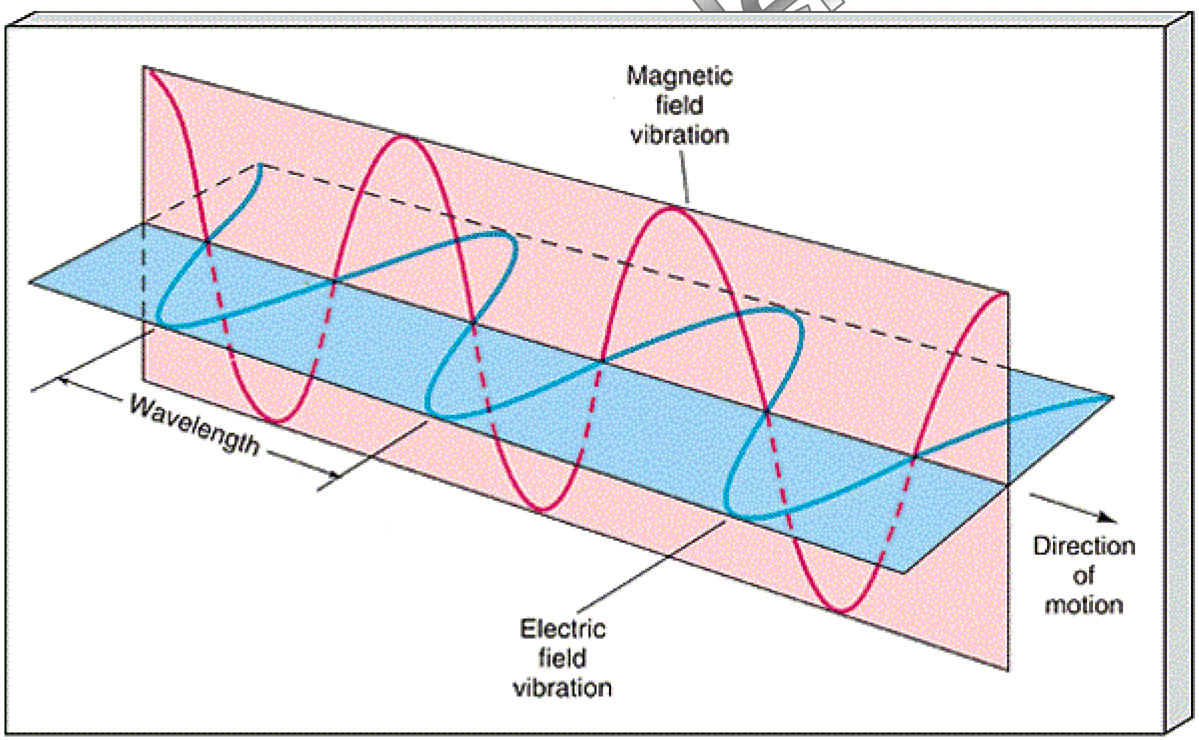
v = velocity of light in the medium
c = the speed of light in a vacuum
n = the refractive index of the medium

Light

- And now, a few words on the concept of “light” before we discuss fiber optic cable.
- Light can be viewed as waves, or as photons, whichever makes more sense to the situation.
- Additionally, light can be analyzed through “ray theory”, as traveling in a straight line and reflecting off certain surfaces, or as being bent as it enters a medium with a different refractive index.

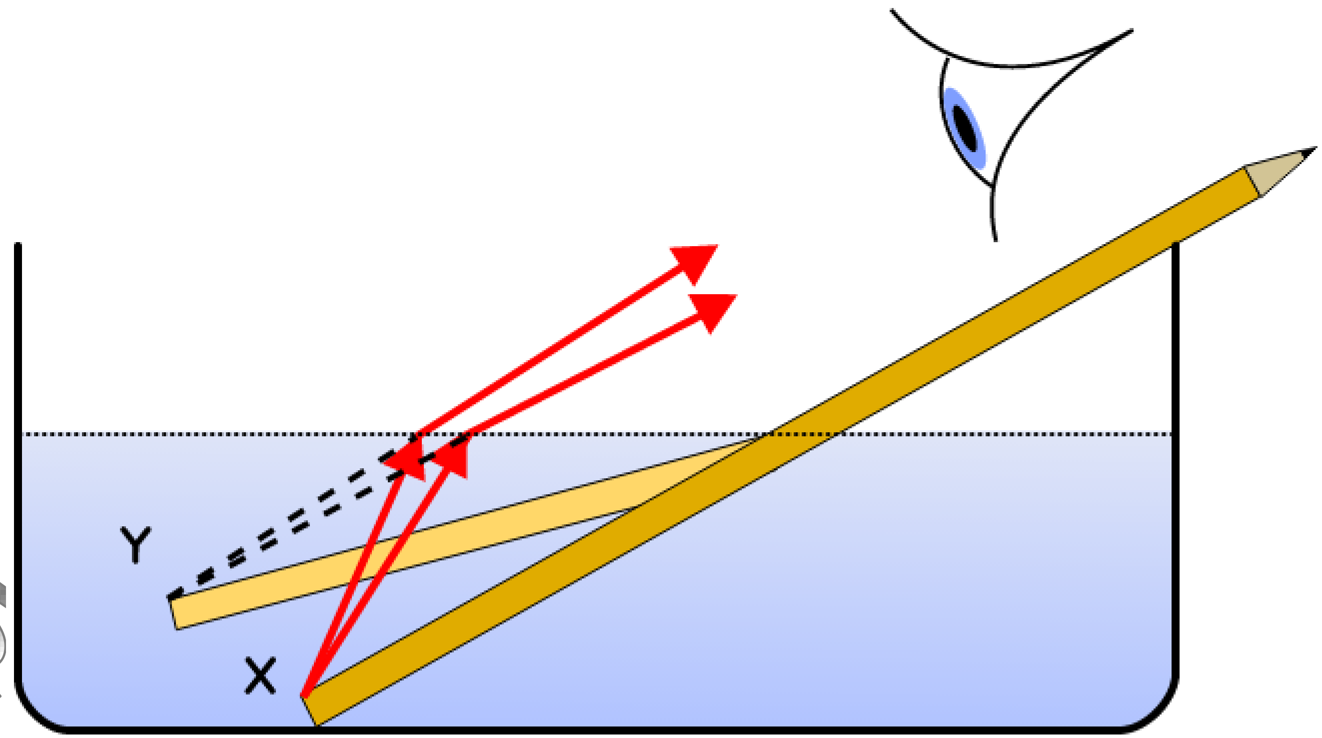
Light

- Light is an electromagnetic wave, exactly like electromagnetic waves at lower frequencies. All of the mathematical tools you used at radio frequencies can be used with light. Modulation of light is mathematically the same as modulation of RF signals.



Refraction and Reflection

- If you put a straw into a glass of water and look down into the water, it appears that the straw is bent. This is due to refraction at the boundary of the water and air.



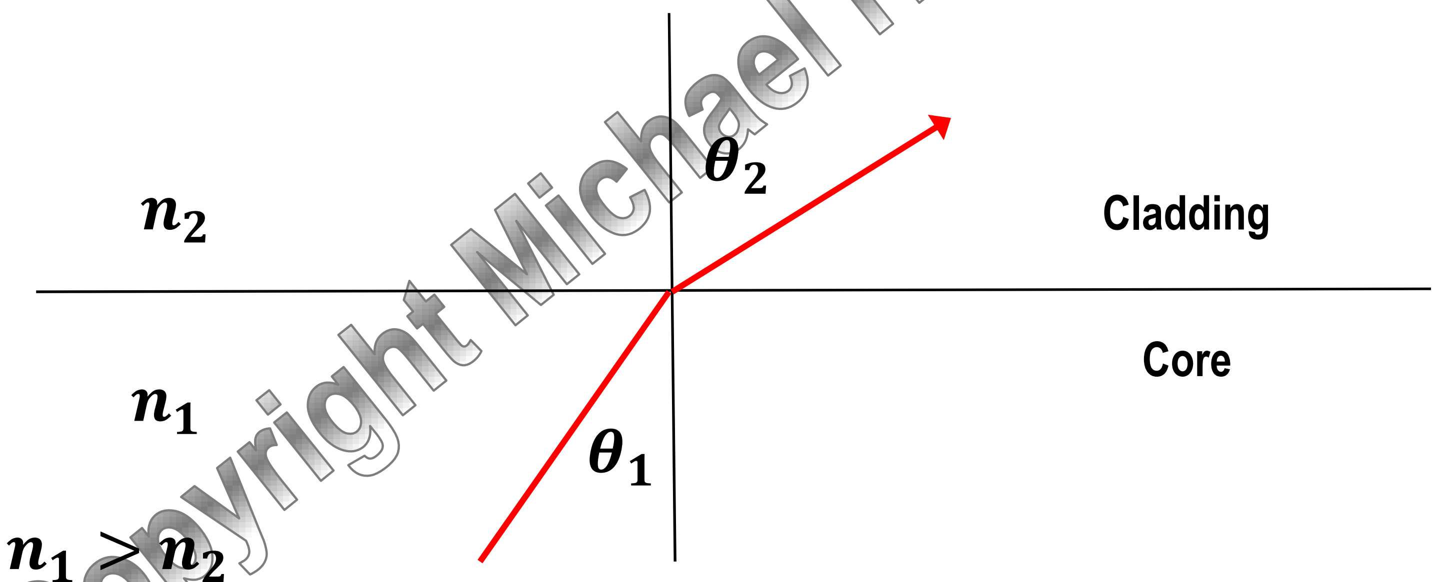
Refraction

- **Here's another example of refraction.**

Snell's Law

- Snell's Law tells us how much the light rays will be bent (refracted) as they move from a medium of one refractive index to a medium of another refractive index

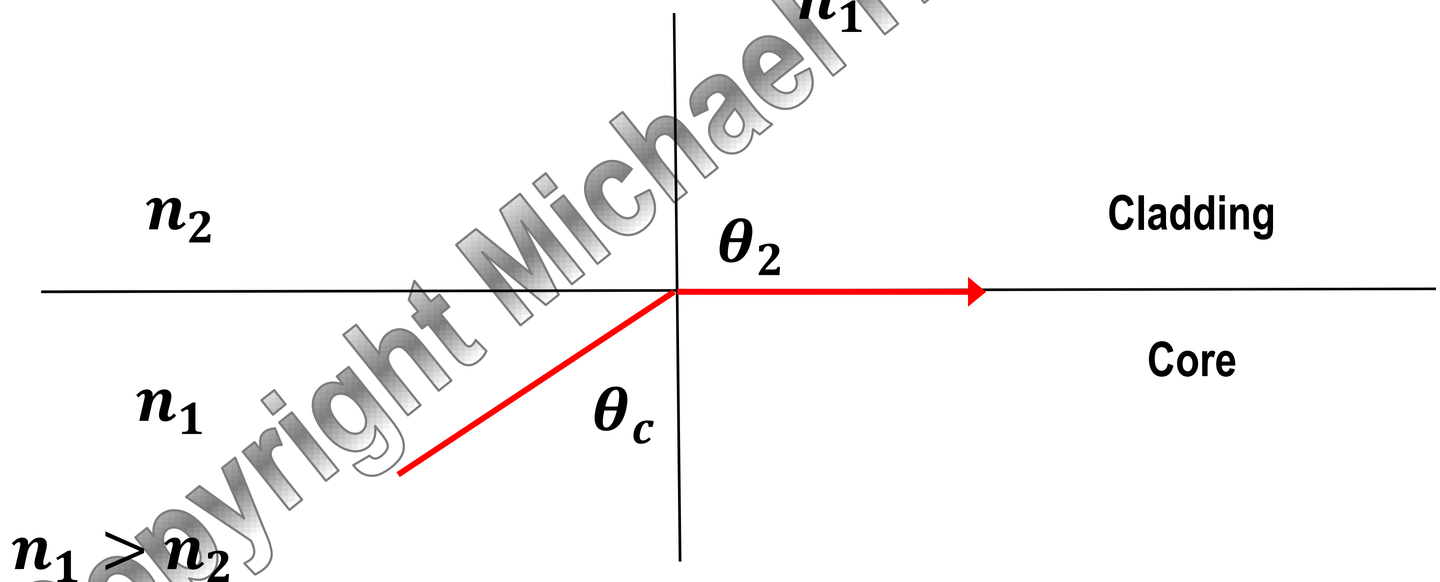
Snell's Law $\quad n_1 \sin\theta_1 = n_2 \sin\theta_2$



Snell's Law

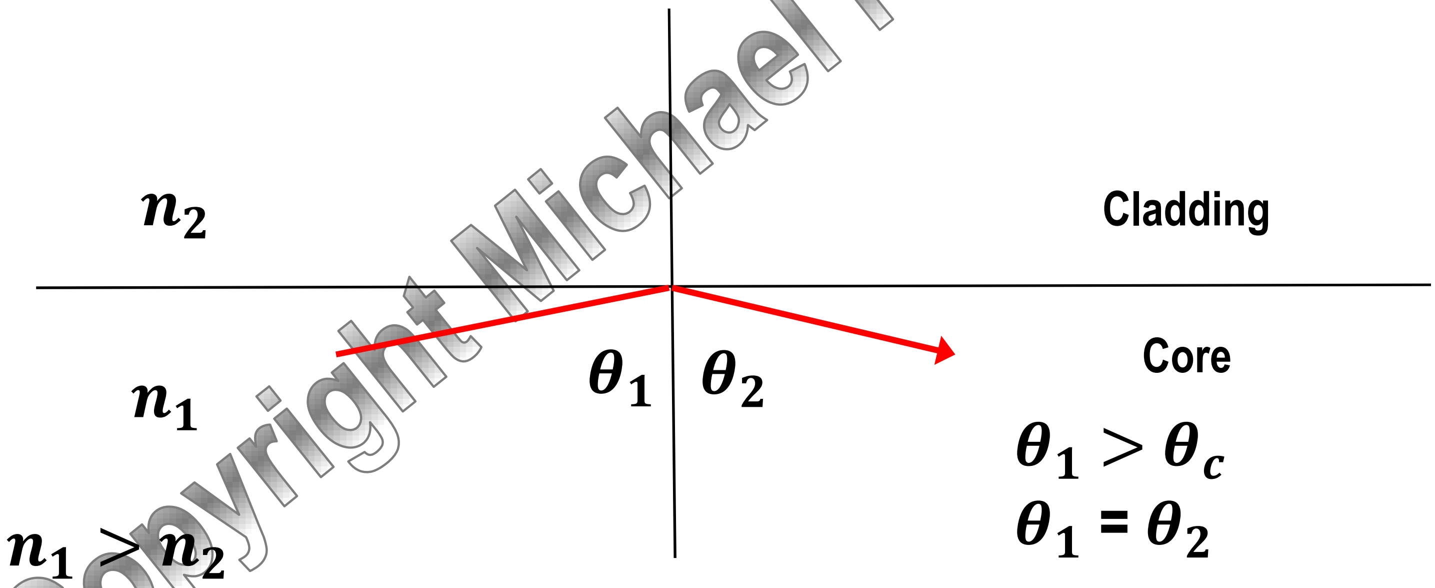
- Depending on the values of n_1 and n_2, there will be an angle of $\theta_1$where sin θ_2 = 1 (90 degrees). This is known as the critical angle, θ_c. Higher angles of θ_1 will result in total reflection.

Critical angle $\quad \sin\theta_c = \dfrac{n_2}{n_1}$

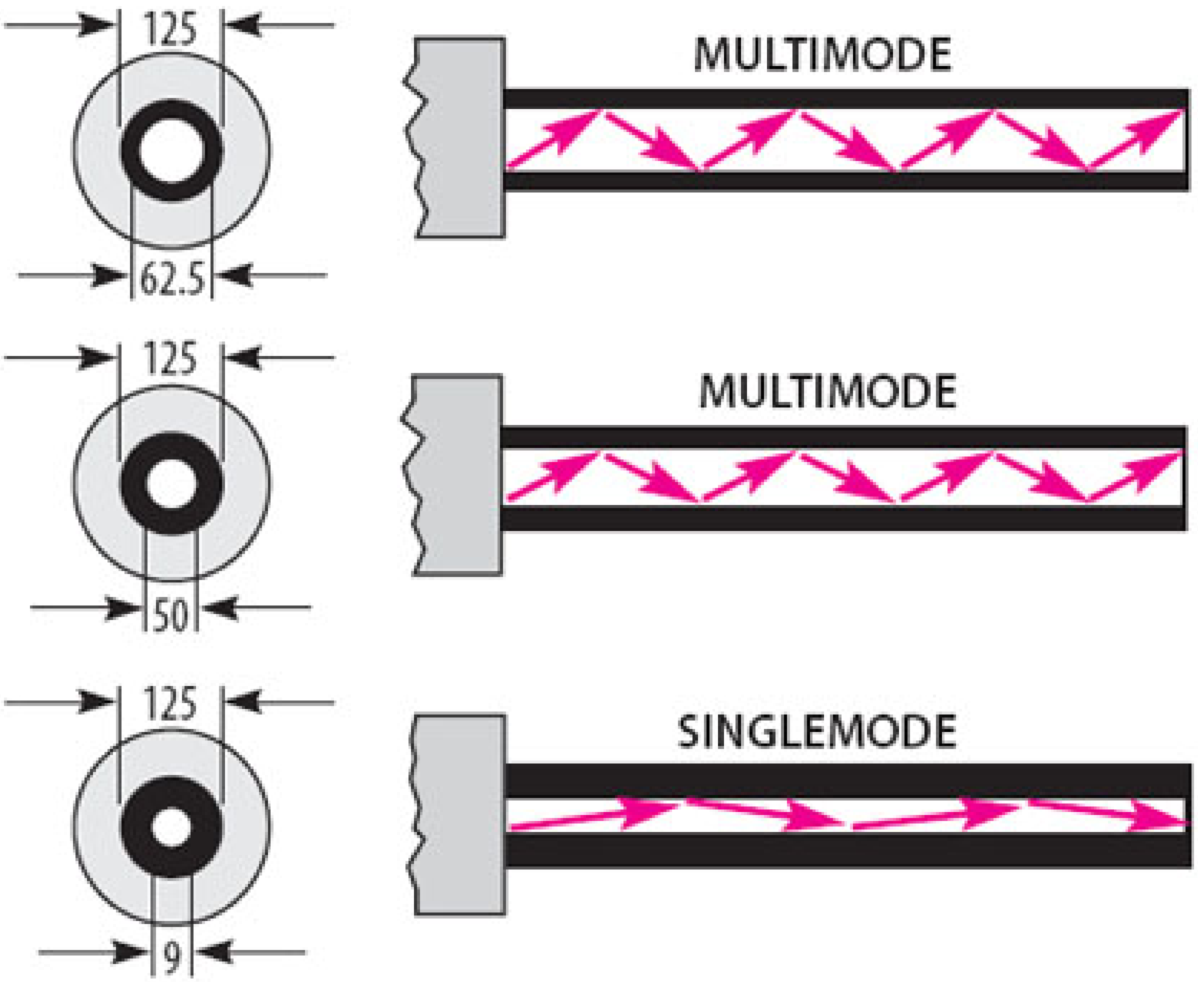


Total Internal Reflection

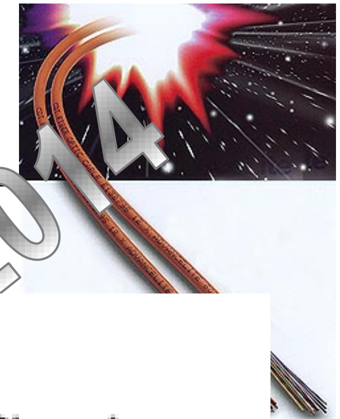
- Angles of θ_1 larger than the critical angle ($\theta_1 > \theta_c$) produce total reflection.
- This is how optical fiber guides the light signals down the fiber.



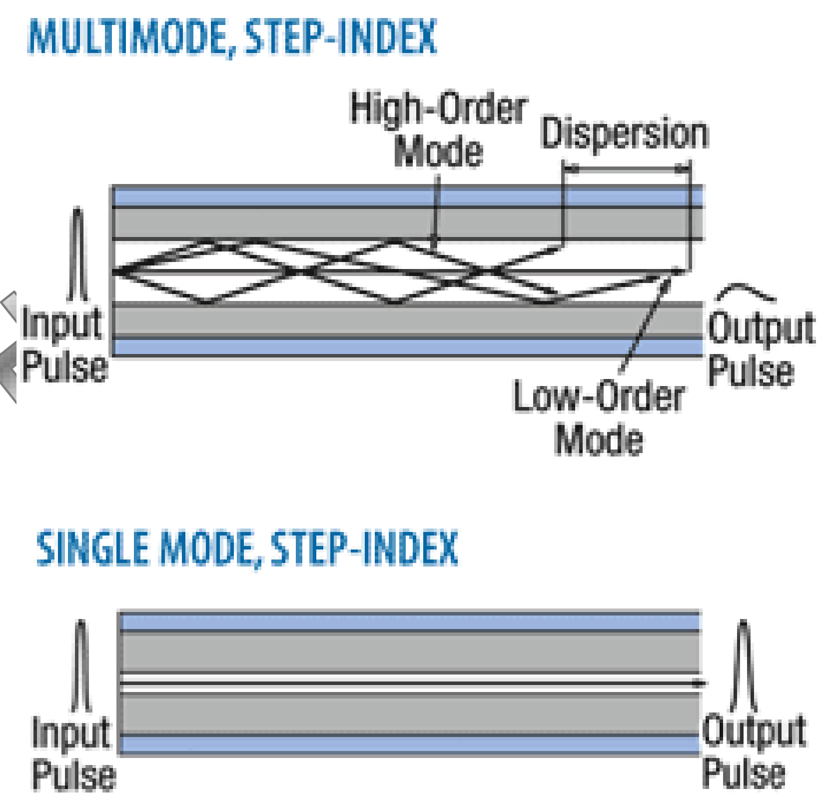
Multimode and Single Mode Fiber



Modes in Optical Fiber



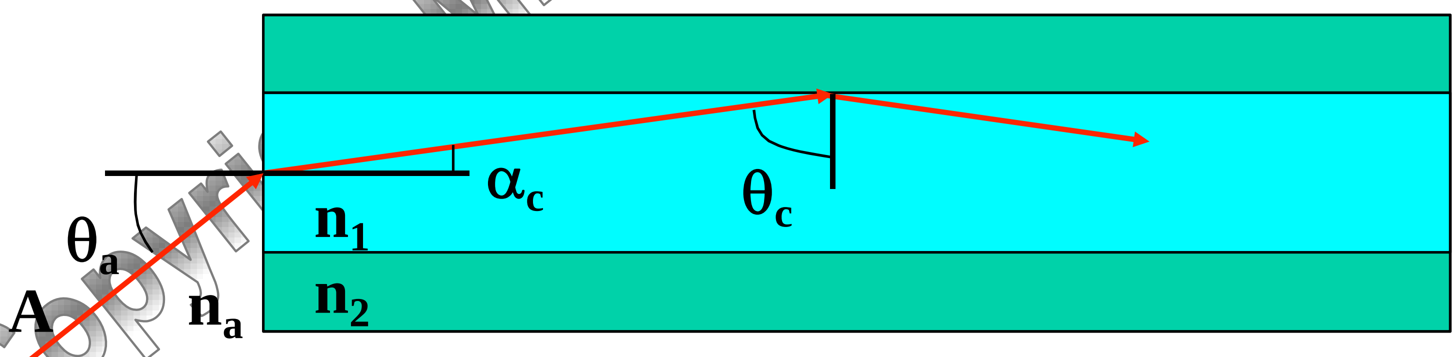
- The large core of the multimode fiber allows the light to follow multiple paths, of different lengths, causing distortion of the pulse. This limits multimode fiber to relatively short runs.
- Single mode fiber only allows the light to follow a single path through the core, reducing the spreading of the pulse.



Acceptance Angle

- The acceptance angle affects our ability to launch an optical signal into the fiber.
- There is an angle, θ_a, which, after being refracted as it enters the cable, resolves to the critical angle, θ_c, inside the fiber.
- We can analyze this situation using Snell's Law.

Snell's Law $\quad n_a \sin \theta_a = n_1 \sin \alpha_c$



Numerical Aperture

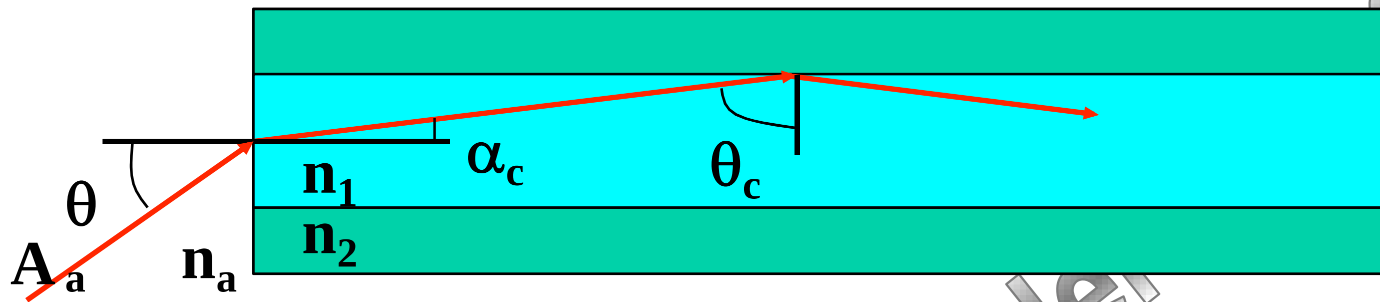
- **Through some algebraic manipulation we can show that**

$$n_a \sin \theta_a = \sqrt{n_1^2 - n_2^2}$$

- **And we define Numerical Aperture as**

$$NA = \sqrt{n_1^2 - n_2^2}$$

Numerical Aperture



- Remember, from Snell's Law, $\sin\theta_c = \frac{n_2}{n_1}$

$$n_a \sin\theta_a = n_1 \sin\alpha_c$$

$$n_a \sin\theta_a = n_1 \cos\theta_c$$

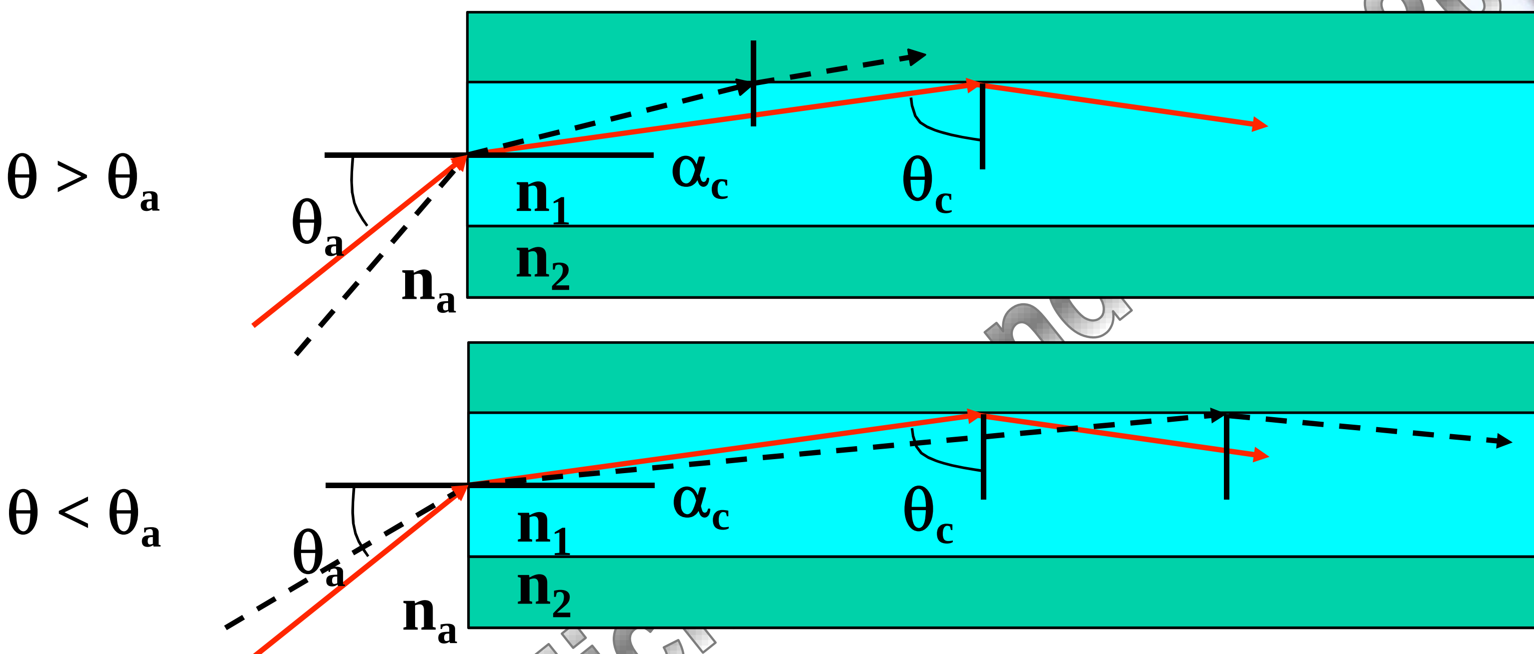
$$\frac{n_a}{n_1}\sin\theta_a = \cos\theta_c$$

$$\frac{n_a^2}{n_1^2}\sin^2\theta_a = \cos^2\theta_c = 1 - \sin^2\theta_c = 1 - \frac{n_2^2}{n_1^2}$$

$$n_a \sin\theta_a = \sqrt{n_1^2 - n_2^2}$$

$$\text{NA} = \sqrt{n_1^2 - n_2^2}$$

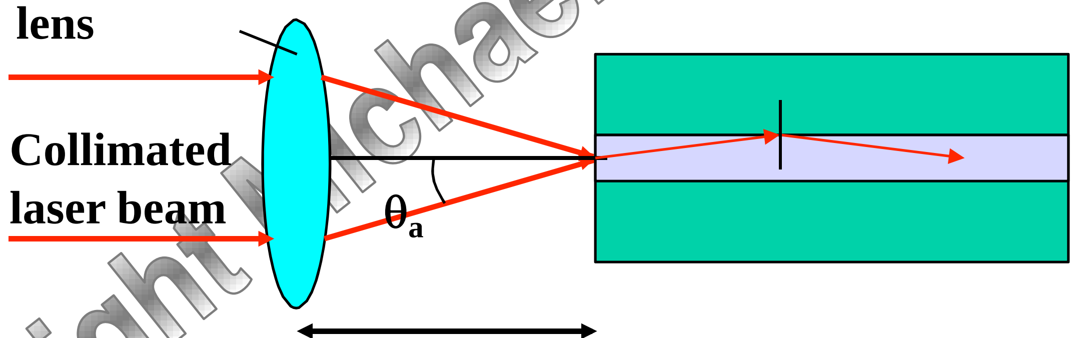
Numerical Aperture



- **Any angle greater than the acceptance angle will cause the light to refract into the cladding and be lost.**
- **Any angle smaller than the acceptance angle will be propagated down the fiber.**

Why do we care about NA?

- Because we have to focus the laser light into the core of the fiber.
- The light from the edges of the lens come into the core at an angle.



Linear Impairments in Fiber - Attenuation

- **As the signal propagates down the fiber it is reduced in power. This is known as “attenuation” or “insertion loss”.**
- **Attenuation in fiber has a number of causes**
  - Material absorption.
    - Intrinsic attenuation is caused by characteristics of the fiber, itself.
    - Extrinsic attenuation is caused by impurities in the fiber.
  - Scattering loss.
  - Bending loss.

Intrinsic Attenuation

- **There is an absorption curve called the "*UV absorption tail*", caused by electron transitions in the glass material.**
  - Remember how ordinary window glass absorbs UV light? This is the tail of that absorption (cutoff about 300nm).
- **There is a second absorption curve called the "*infrared and far-infrared absorption tail*" caused by molecular vibrations (very high attenuation beyond 2000nm).**
- **Together, these two curves, along with Rayleigh scattering, set the floor for attenuation in fiber.**
- **The best fiber today has about 0.2dB/km attenuation in the 1550nm band. That's about as good as it's going to get.**

Extrinsic Attenuation

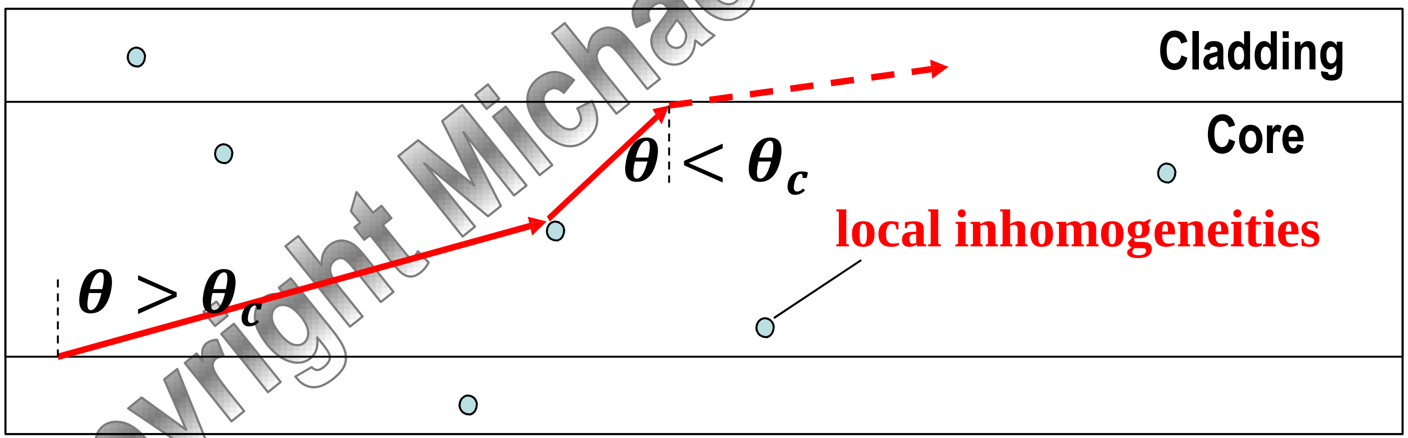
- The major impurity in optical fiber used to be water – the hydroxyl ion from the oxyhydrogen flame used to heat the ingot when pulling the fiber.
- These OH^- ions have an absorption peak at 1380nm.
- Since this peak is quite narrow, the bands at 1330nm and 1550nm are unaffected by this absorption.

Scattering Loss

- The major scattering loss in fiber is Rayleigh scattering.
- Glass is amorphous (not crystalline), and thus has areas of inhomogeneity which are much smaller than the wavelength of the signal.
- When light strikes one of these areas, it absorbs energy, vibrates, and then reradiates the light in a random direction.
- The intensity of Rayleigh scattering is inversely proportional to the fourth power of the wavelength (shorter wavelengths are affected more than longer wavelengths).
- Rayleigh scattering is one of the primary reasons the sky is blue.

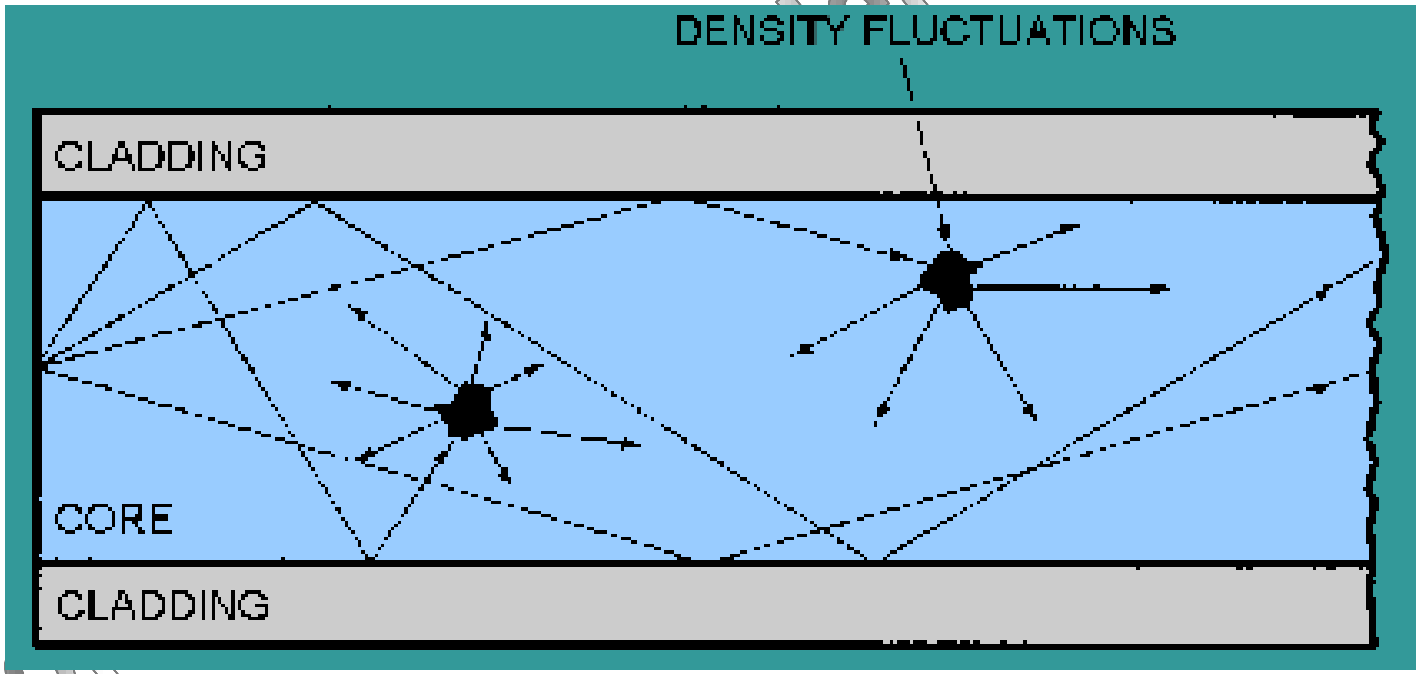
Rayleigh Scattering

- Some of the scattered light will be scattered forward and conserved.
- Some will be scattered backward and lost.
- Some will be scattered into the cladding and lost.

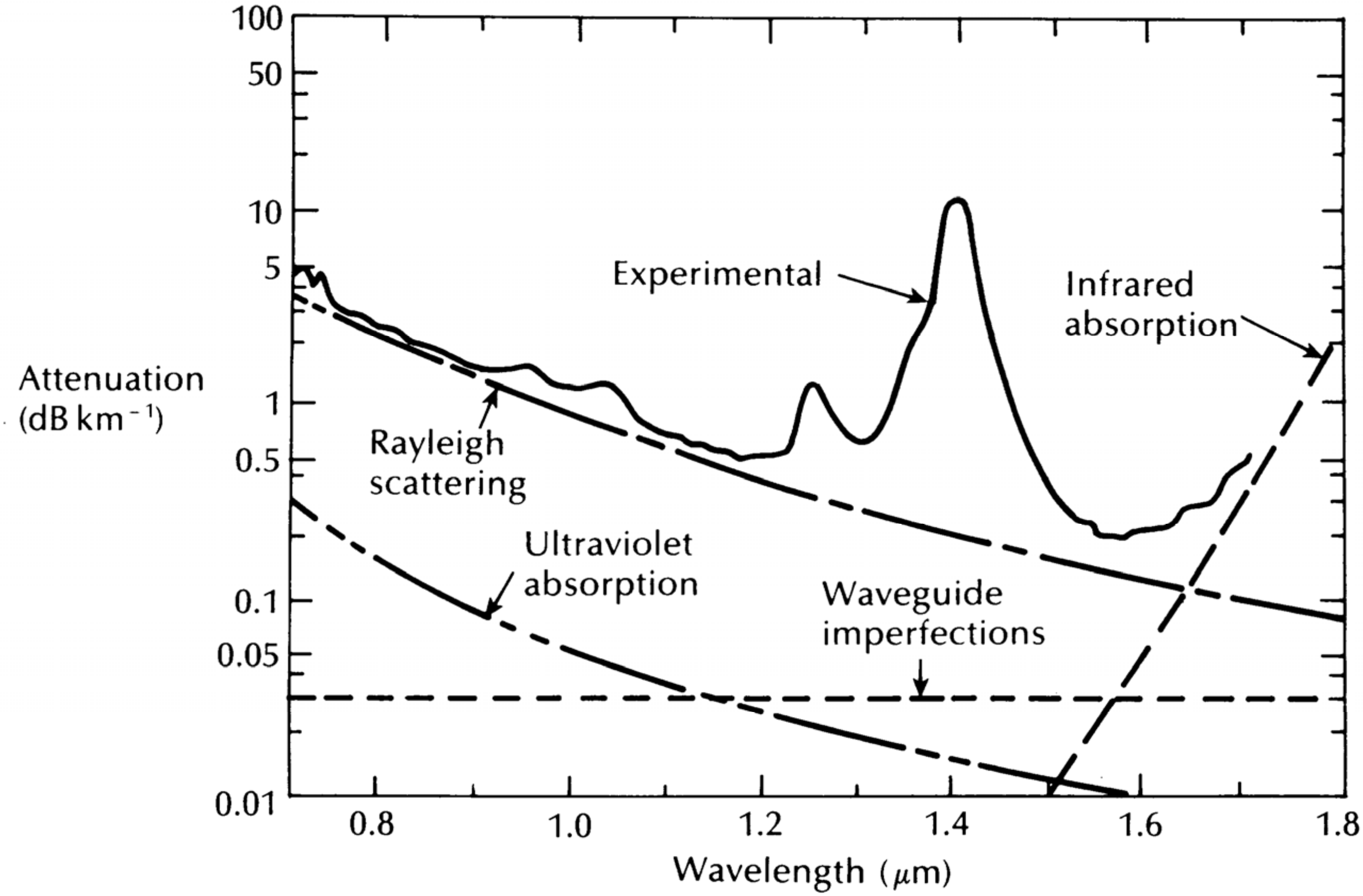


Rayleigh Scattering

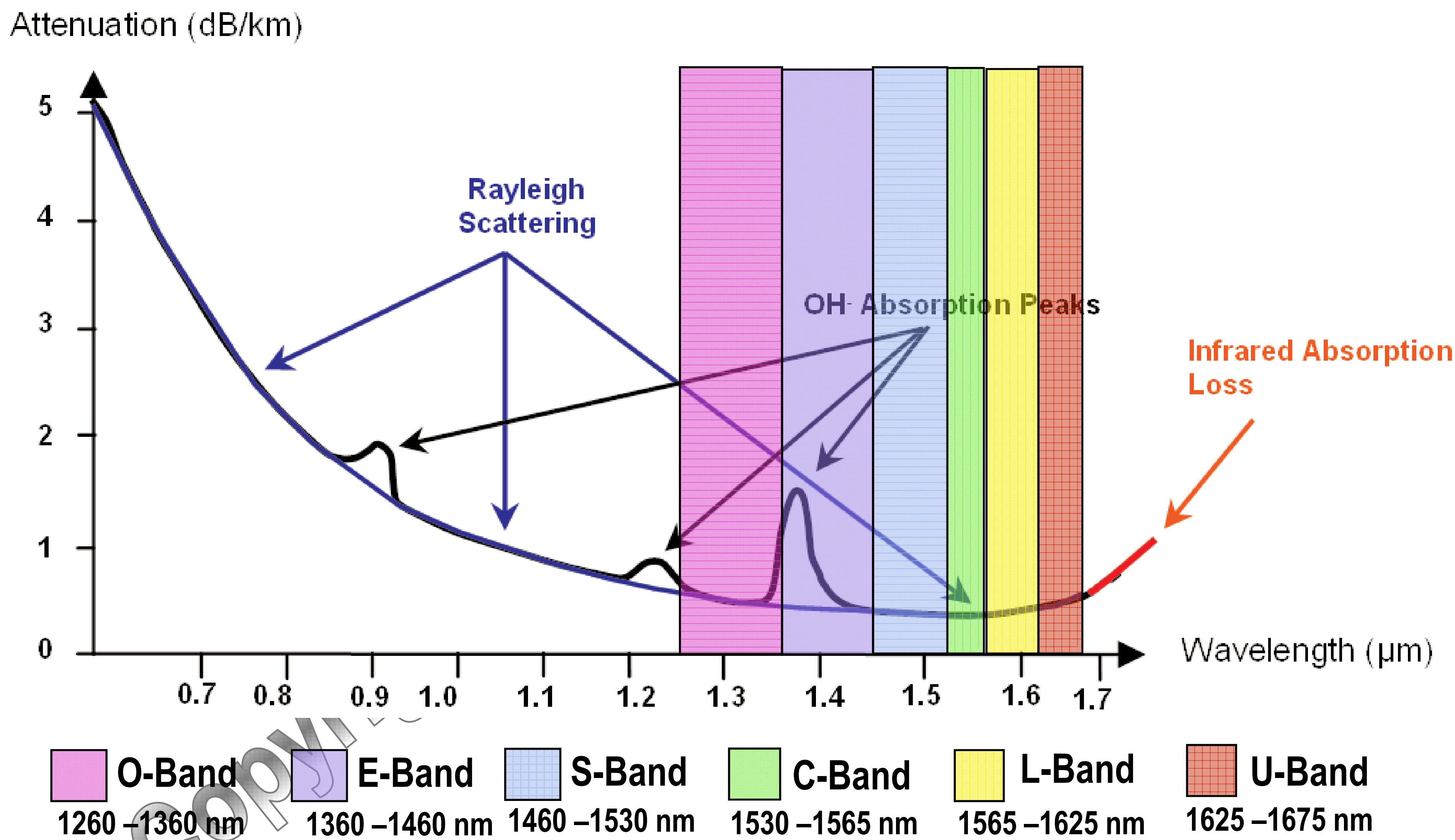
- **Another diagram demonstrating Rayleigh scattering.**
- **This diagram shows the re-radiation occurring in random directions but shows the inhomogeneities as being much too large.**



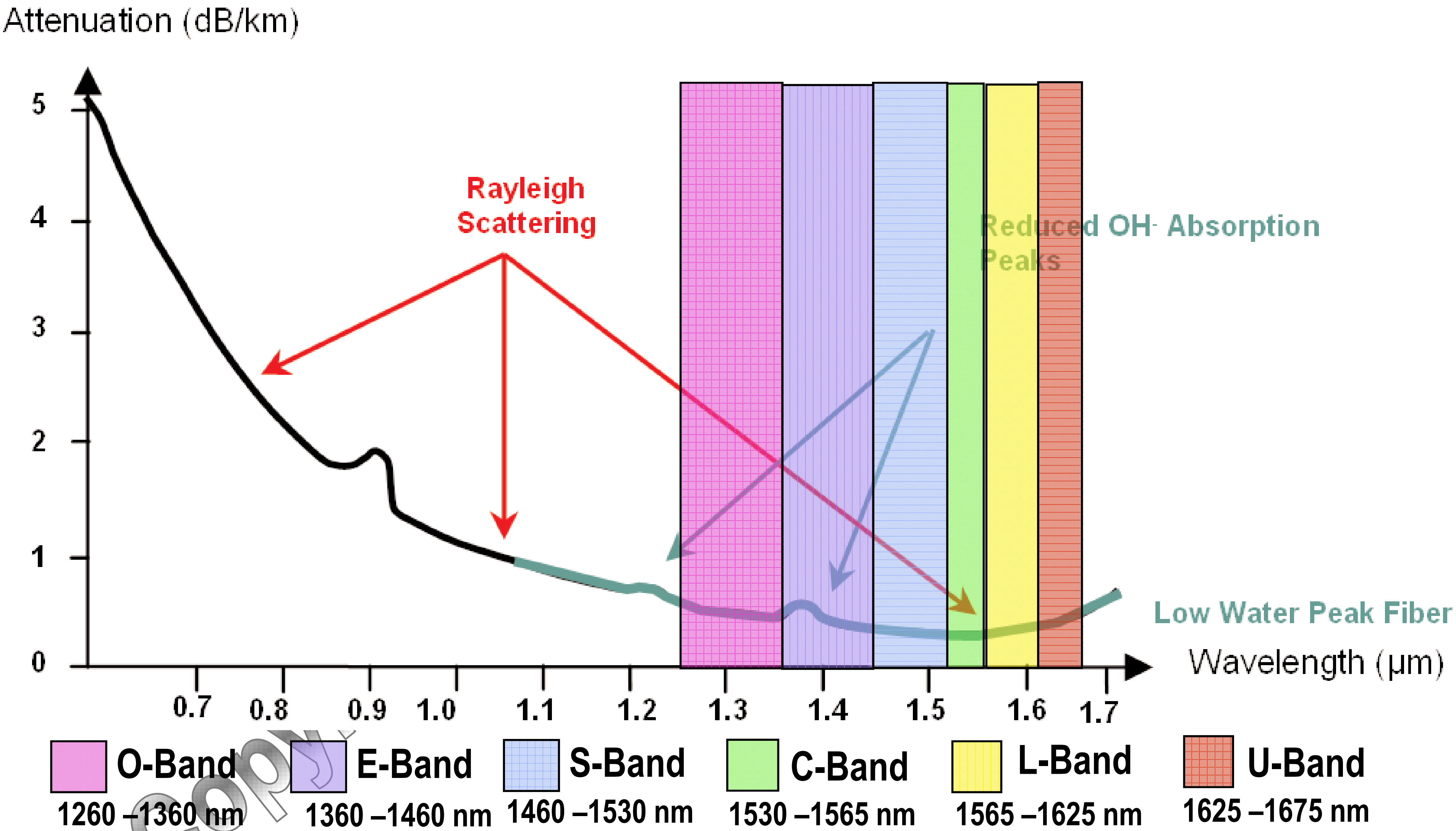
Attenuation in Optical Fiber



Optical Spectrum – Standard Fiber

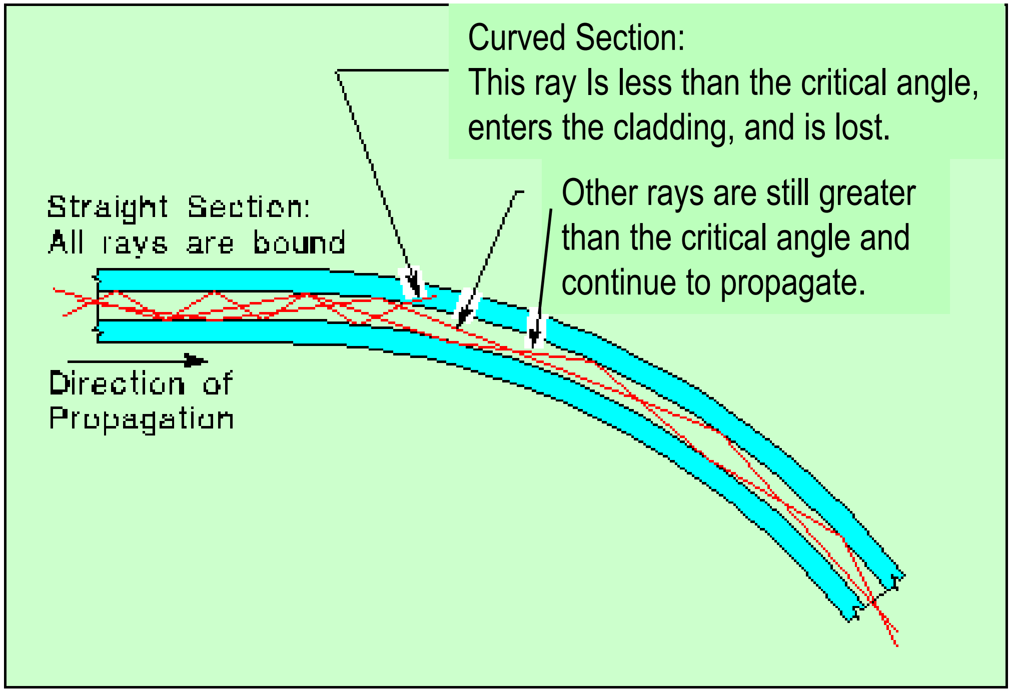


Low Moisture Optical Fiber



Bending Loss

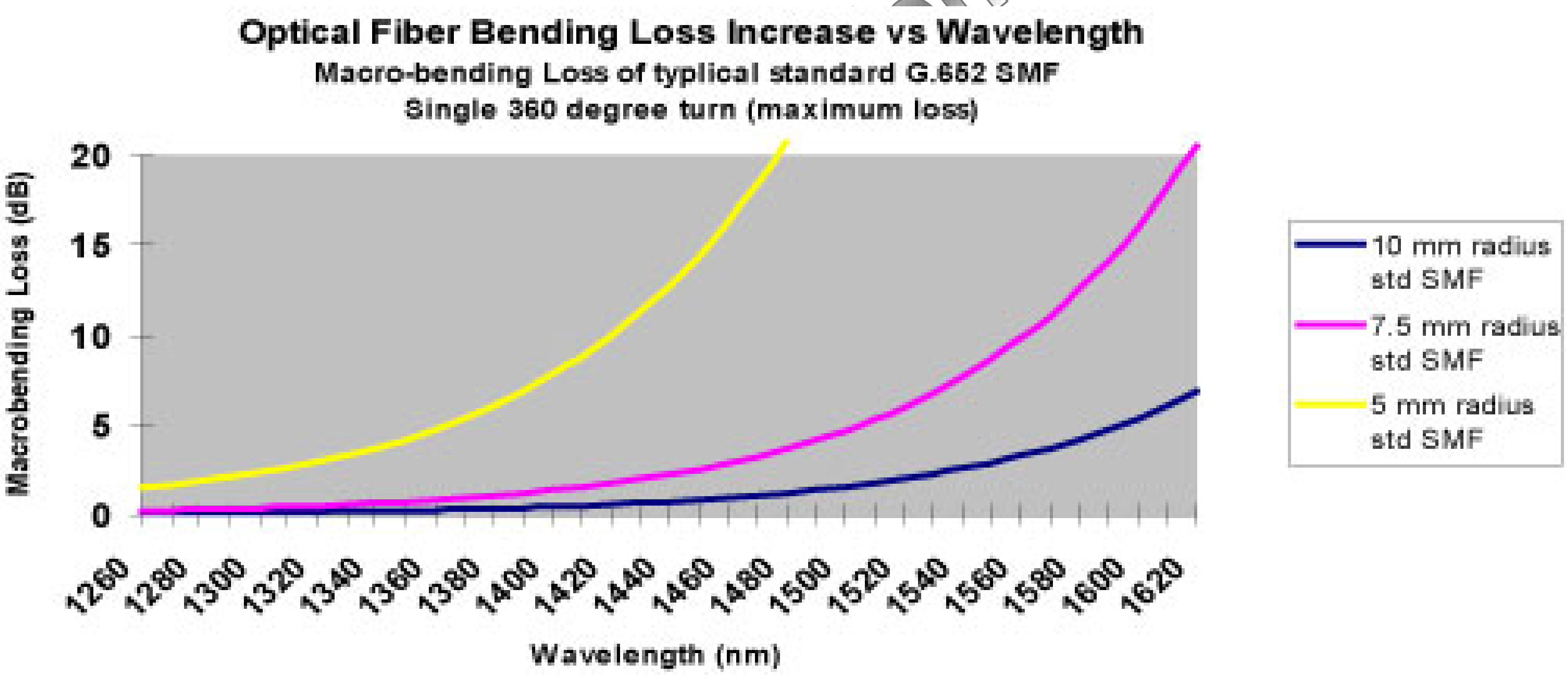
- **If a fiber is bent sufficiently, the light ray can strike the boundary with the cladding at less than the critical angle, be refracted, and lost.**



macrobend loss

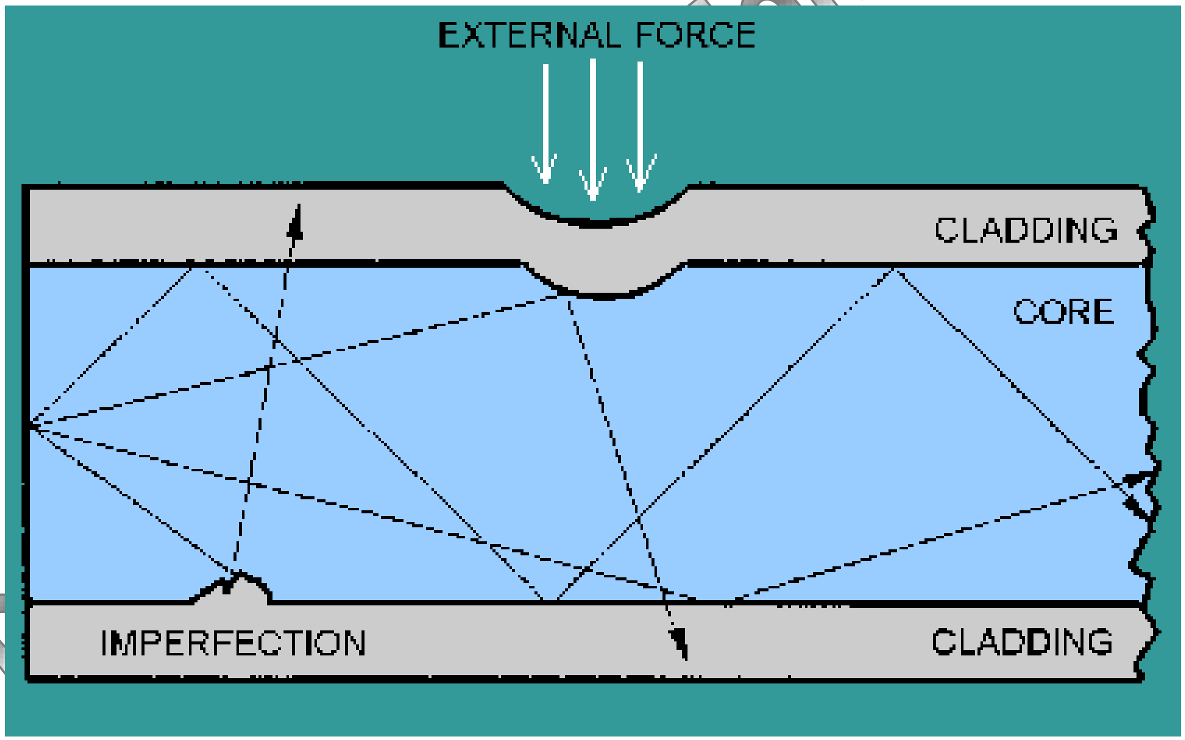
Bending Loss

- Bending loss increases with wavelength.
- The examples given here are fairly tight bends and produce very high attenuation.



Bending Loss

- Bending loss can also occur on “pinched” fibers, or from manufacturing imperfections.

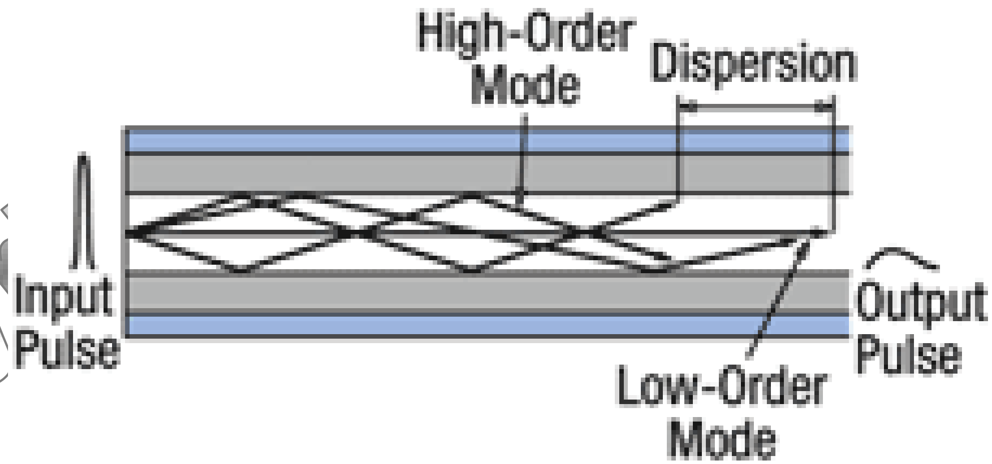


Linear Impairments in Fiber - Dispersion

- **Dispersion – spreading of the optical pulse in time**
  - Material Dispersion – caused by differences in the refractive index with frequency.
  - Waveguide Dispersion – Fields extend into the cladding and travel faster because of lower refractive index. Source of negative dispersion.
  - Polarization Mode Dispersion – Caused by differences in refractive index because the fiber is anisotropic (not perfect – e.g., core may be elliptical).
  - Dispersion is measured in ps/(km-nm).

Model Dispersion

- Characteristic only of multimode fiber. The light travels down the fiber in many modes, some directly down the fiber, some reflecting off the boundary between the core and the cladding.
- The "reflecting" path is longer and thus that light takes longer to travel down the fiber than the straight path. This causes the received signal to be broadened.

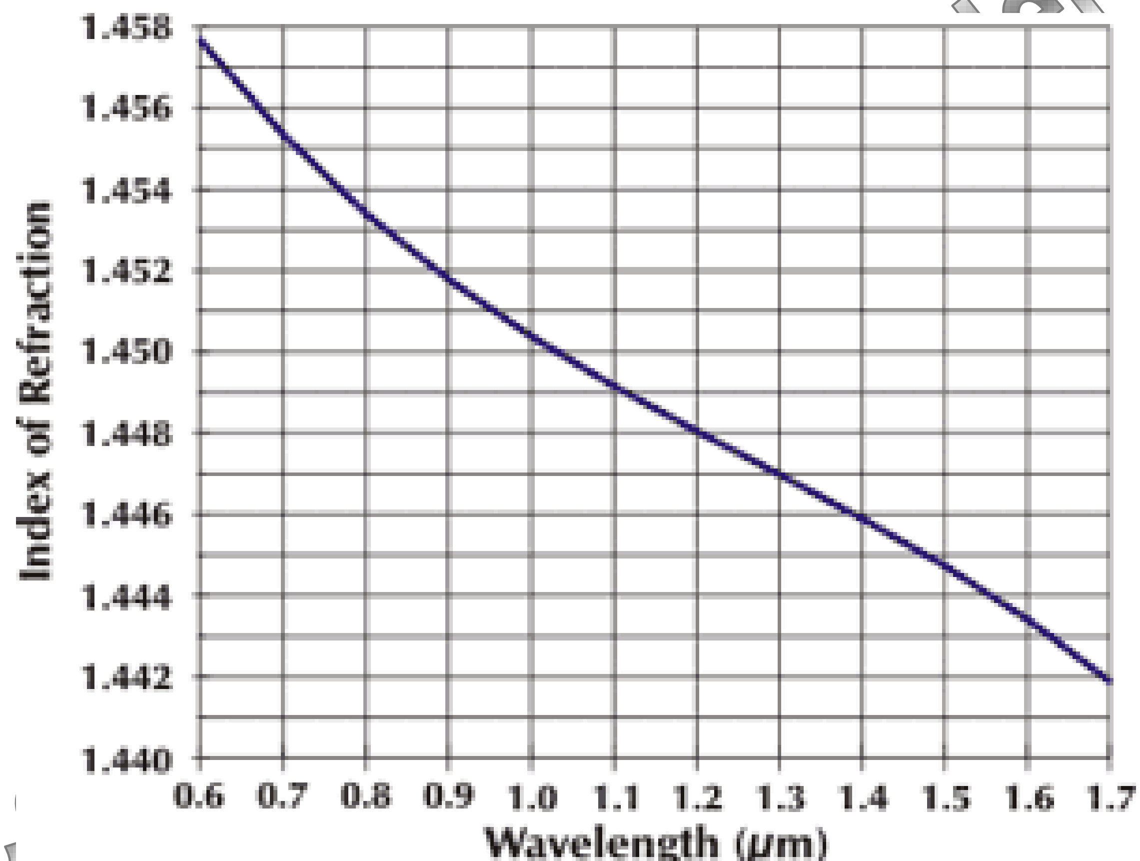


Material Dispersion

- The refractive index of both the core and cladding are frequency dependent.
- Remember that $v = \frac{c}{n}$ where v is the speed of light in the medium, n is the refractive index and c is the speed of light in a vacuum.
- Shorter wavelengths see a higher refractive index and thus travel slower.
- Longer wavelengths see a lower refractive index and travel faster.

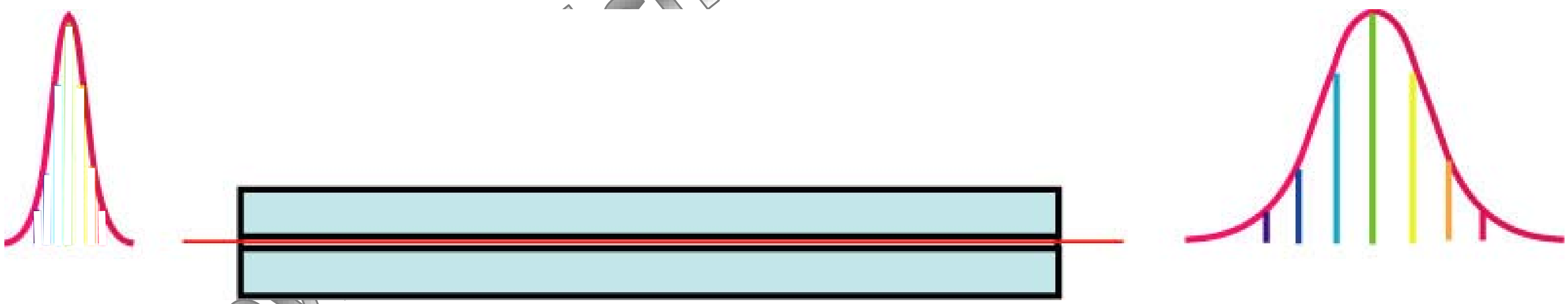
Index of Refraction for Optical Fiber

- **The chart shows the relationship of refractive index and wavelength. Shorter wavelengths (higher frequency – bluer light) see a higher refractive index and thus travel slower.**



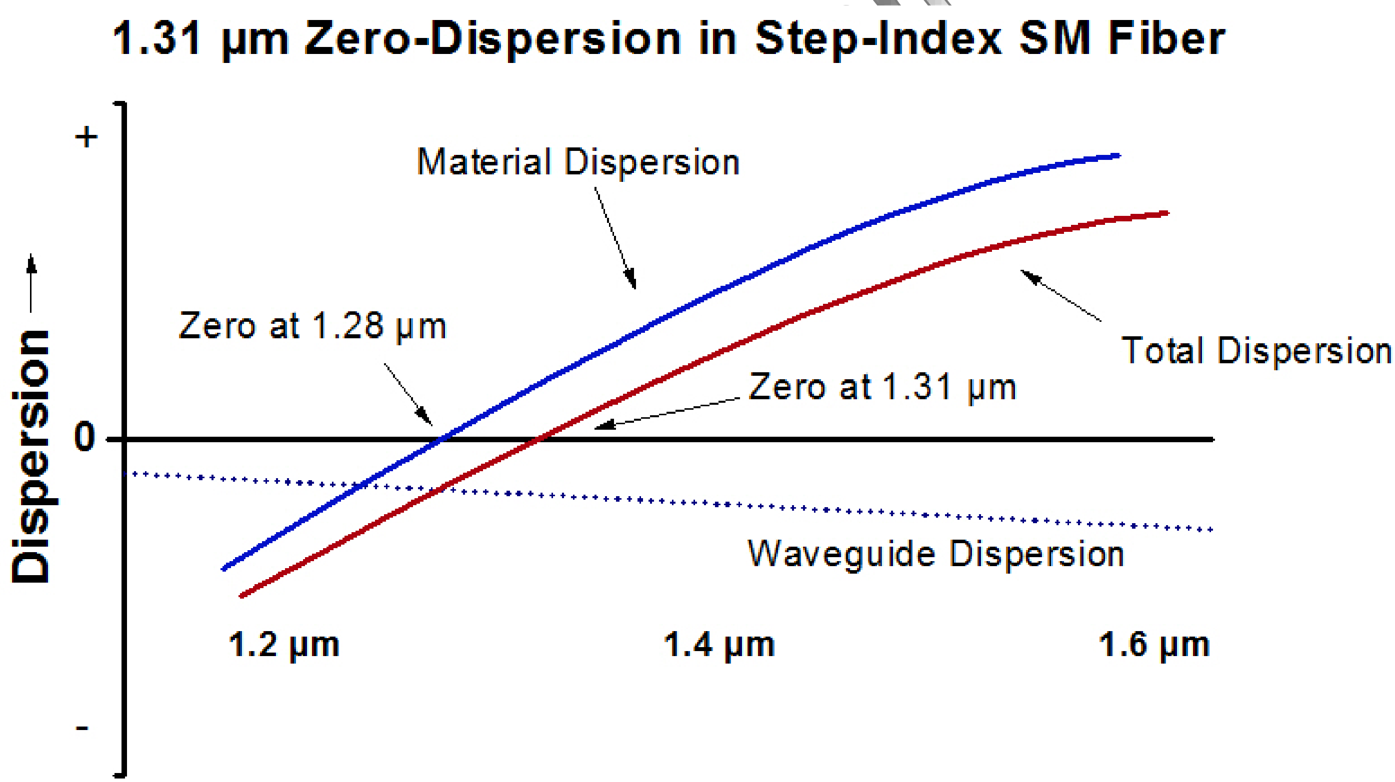
Material Dispersion

- This figure shows how the pulse is broadened because of the differences in speed of different frequencies.
- The input pulse is "white" because all of the components are together.
- Note that "red" (long wave) arrives earlier than "blue" (shorter wave).



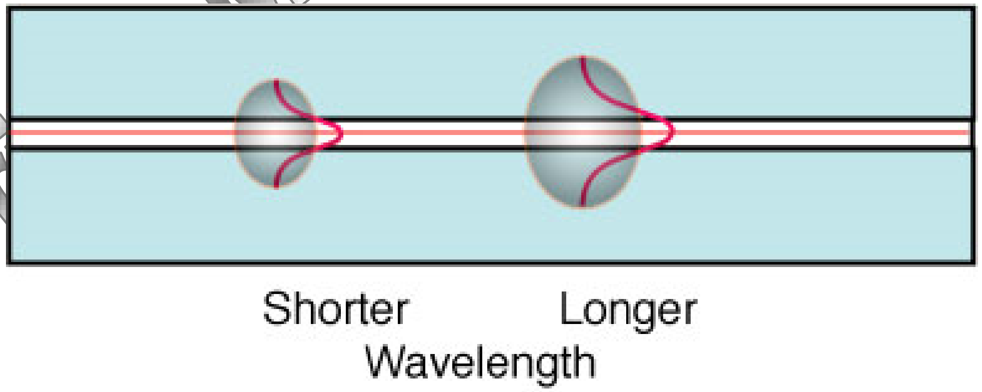
Material Dispersion

- Here's a chart showing the curve of material dispersion (blue line). The zero dispersion point for fused silica is 1276nm.
- The total dispersion is the red line, but I have to discuss waveguide dispersion before it makes sense.



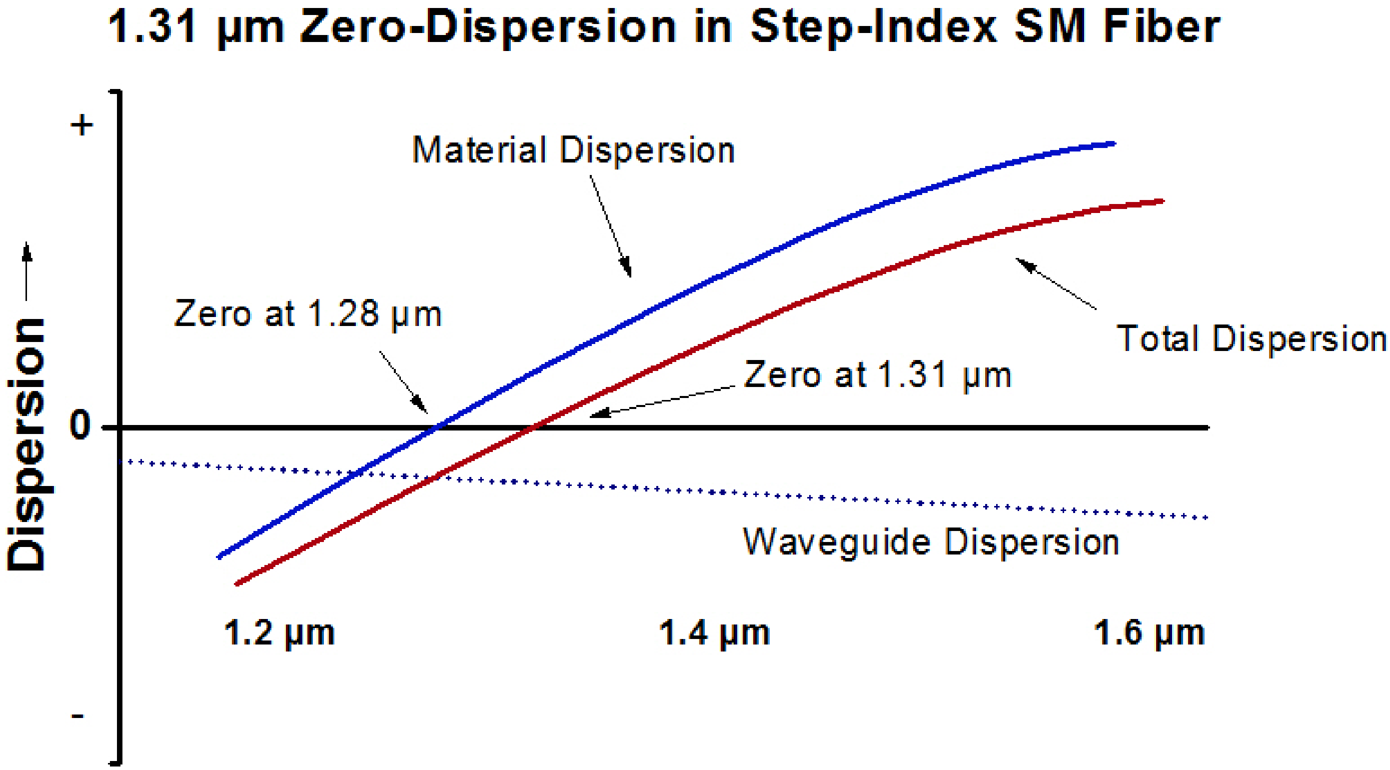
Waveguide Dispersion

- In single mode fiber, the wavelength of the light is larger than the core of the fiber.
- As a result, the electric and magnetic fields of the light actually travel in an area that exceeds the diameter of the core - so part of the "light" is traveling in the core of the fiber and part is traveling in the cladding.
- Because of the differences in refractive index, the part of the signal in the cladding arrives first, causing dispersion.
- Longer wavelengths extend further into the cladding.



Chromatic Dispersion

- Chromatic dispersion is the sum of the material dispersion and the waveguide dispersion.
- The waveguide dispersion is negative and so shifts the chromatic dispersion curve to the right from the material dispersion curve, with a zero dispersion point near 1310nm.

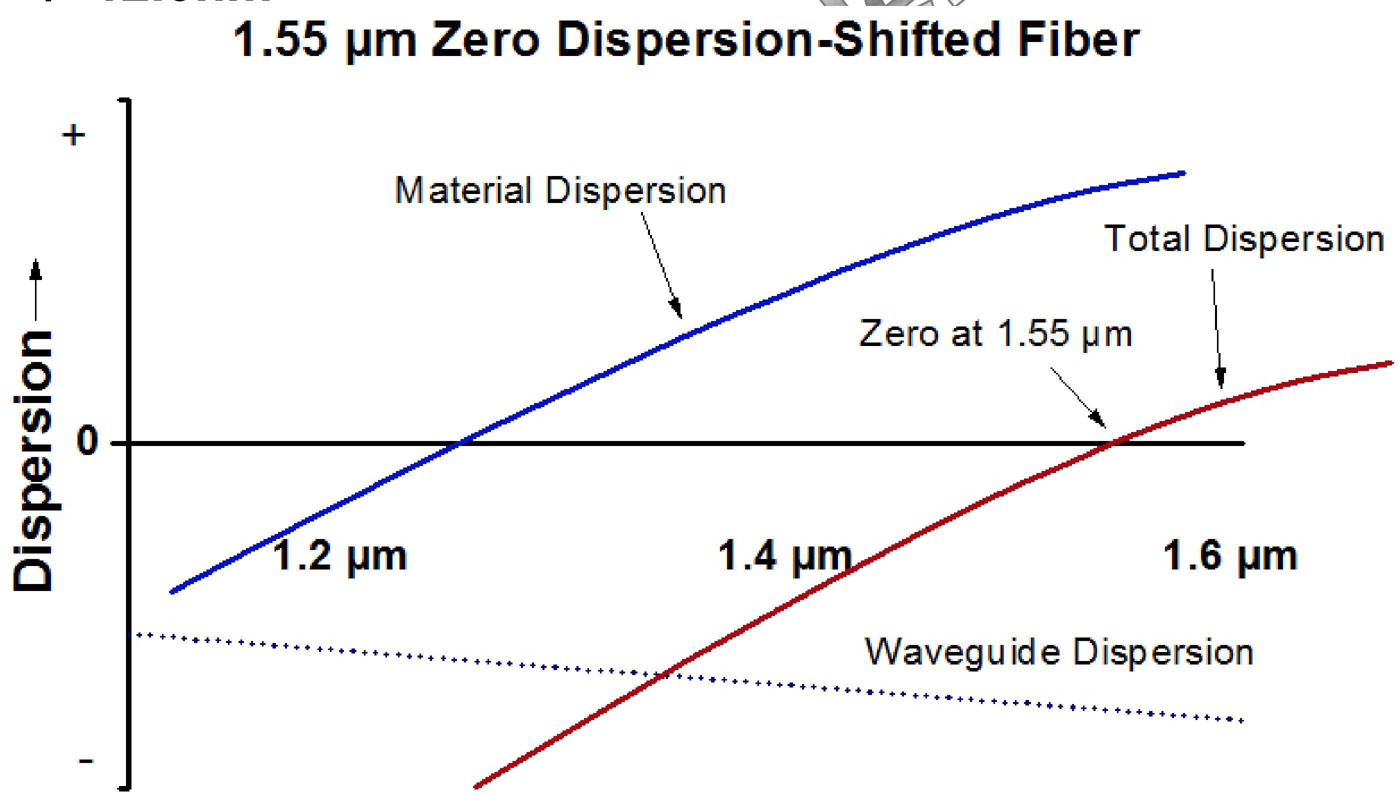


Chromatic Dispersion

- Chromatic Dispersion can be positive, zero, or negative.
- If positive, shorter wavelengths travel slower than longer wavelengths (higher refractive index for shorter wavelengths).
- If negative, shorter wavelengths travel faster than longer wavelengths.
- The effect of chromatic dispersion increases with the square of the bit rate. So a doubling of the bit rate causes a four-fold impact of dispersion.

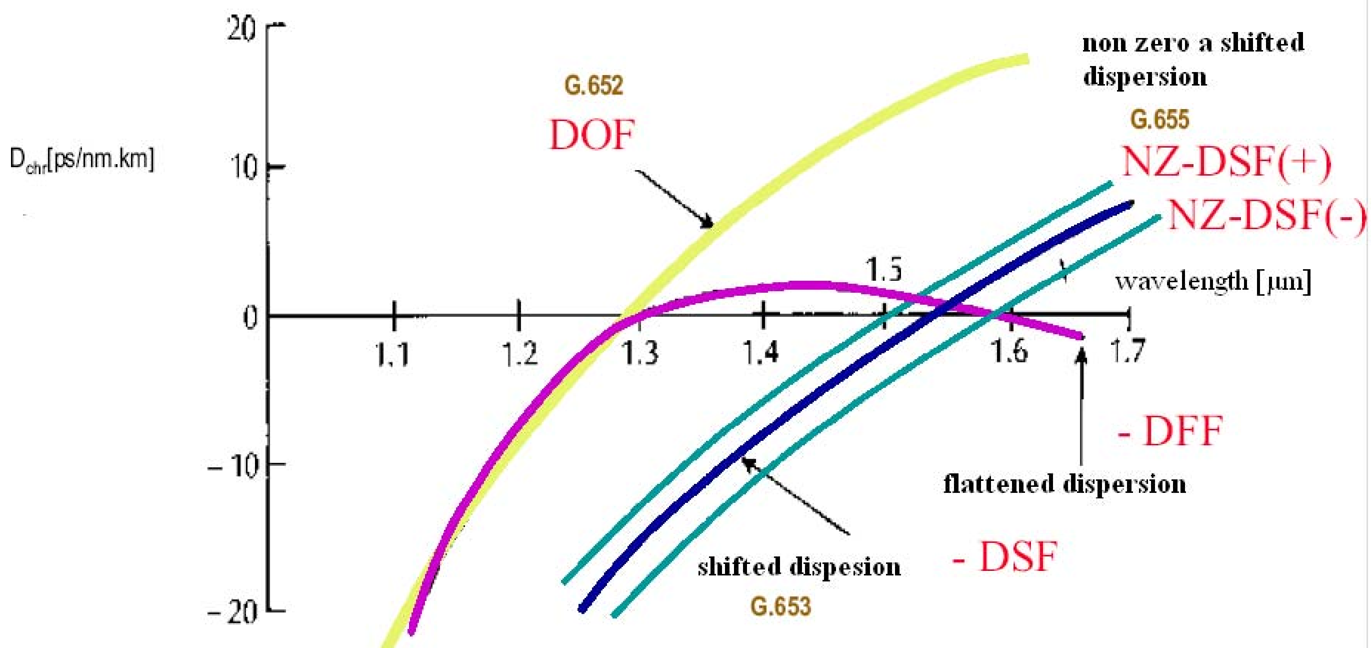
Dispersion Shifted Fiber (DSF)

- Through doping, the waveguide dispersion can be shifted to a more negative direction.
- This shifts the chromatic dispersion to a longer wavelength, as shown in the diagram below. Standard is 1557.5nm, +/- 12.5nm



Non-Zero Dispersion Shifted Fiber (NZ-DSF)

- **When DWDM systems (to be explained soon) came into being, it was found that having the zero dispersion point at 1550nm did not work well.**
- **This led to the development of fiber with a zero dispersion point just off of 1550nm, and Dispersion Flattened Fiber (DFF).**

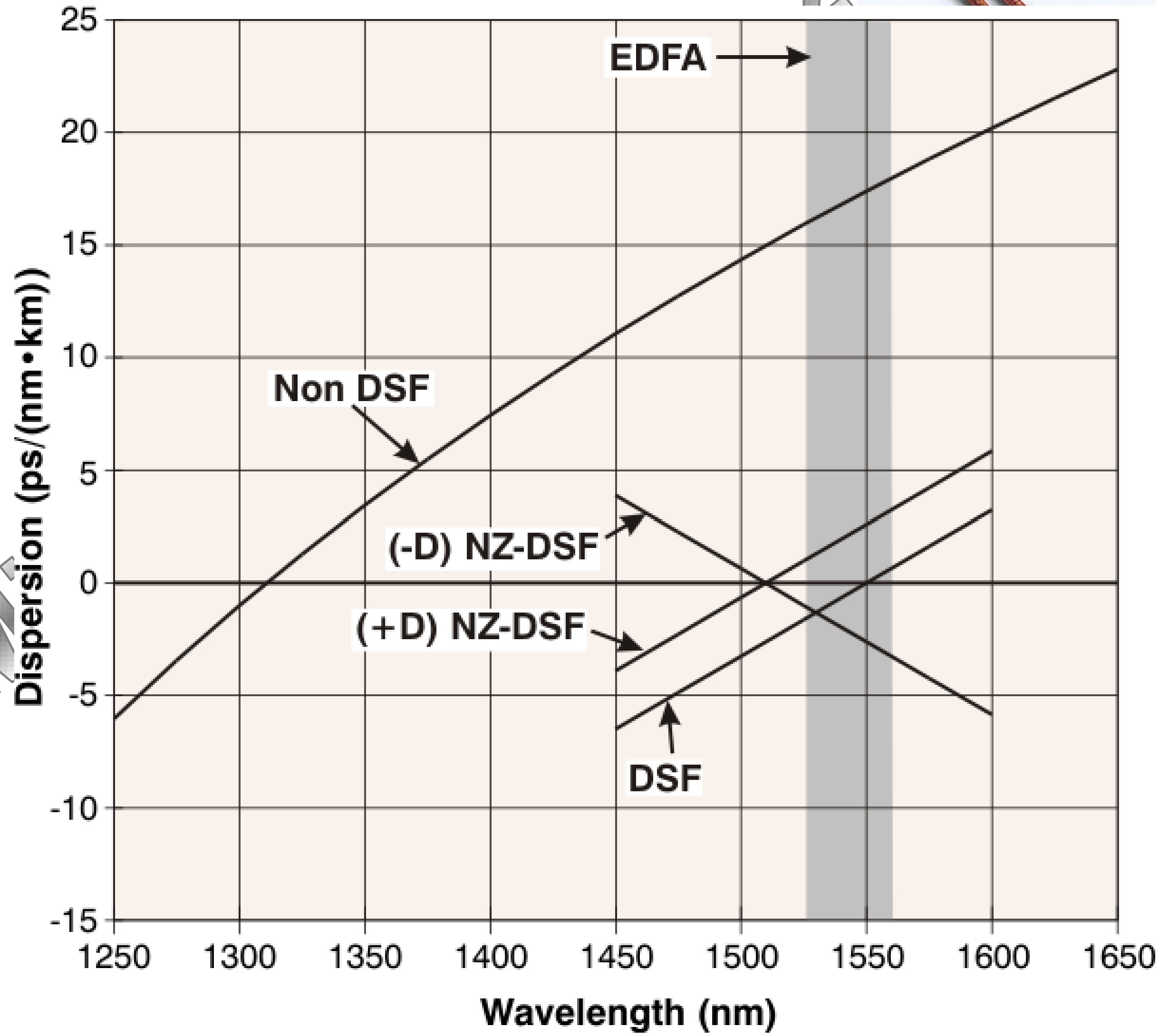


Chromatic Dispersion

- Fiber communication rates typically increase by a factor of four (OC-3 to OC-12 to OC-48, for example).
- The cost and complexity of the electronics are usually less than four times the lower rate.
- However, when increasing the rate by four, the impact of chromatic dispersion is ***16 times*** the lower rate.
- At 10Gbps and, especially at 40Gbps, dispersion compensation is absolutely necessary.

Dispersion Compensating Fiber

- NZ-DSF is available with either a positive or negative total dispersion curve.
- (-D) NZ-DSF can be used for dispersion compensation.
- Other solutions are available for dispersion compensation.

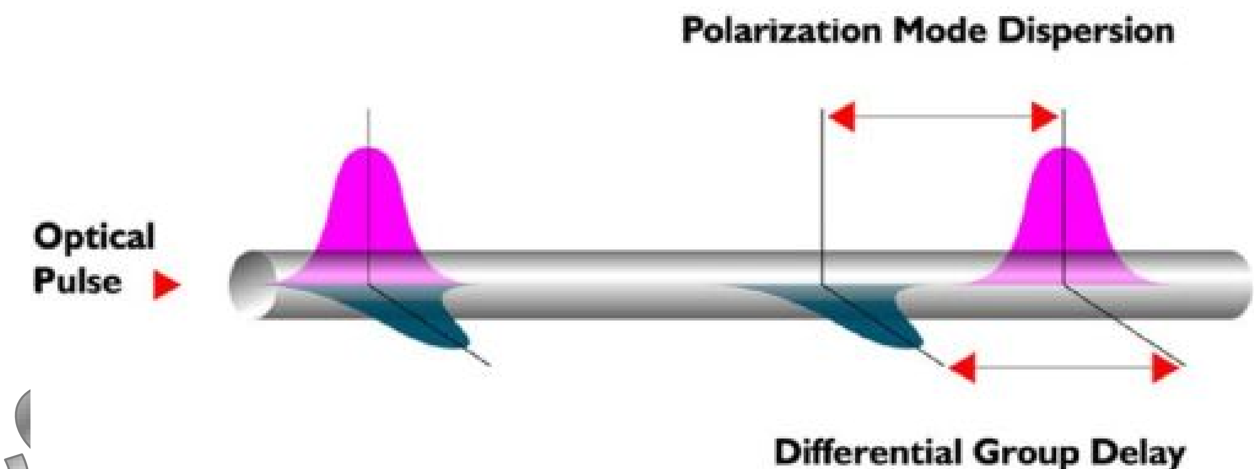


Fiber Standards

- **The ITU has developed several recommendations for fiber.**
  - G.652 for standard fiber with a zero dispersion at 1330nm. Also called standard single mode fiber.
  - G.653 for Dispersion Shifted Fiber.
  - G.655 for Non-Zero Dispersion Shifted Fiber.
  - Several others for specialized applications, including submarine cables.

Polarization Mode Dispersion

- Polarization mode dispersion only becomes important at higher data rates – 10Gbps and above.
- It is caused by irregularities in the fiber – for example, the core not being perfectly round.
- Single mode fiber supports two orthogonal polarizations which travel at different velocities because of these irregularities.



Wavelength Division Multiplexing

- Fiber has tremendous bandwidth.
- Engineers started to ask themselves, “Why limit a fiber to one channel? Why can’t we put multiple channels on a fiber, using frequency division multiplexing?”
- The answer, of course, is that you can do frequency division multiplexing on a fiber, and that’s what Wavelength Division Multiplexing is.
- Terminology – Each wavelength (or signal) is known as a “lambda” because of the Greek letter, λ, used to represent wavelength.
- Note that WDM is a just way to get more capacity out of fiber. It is only a transport mechanism.

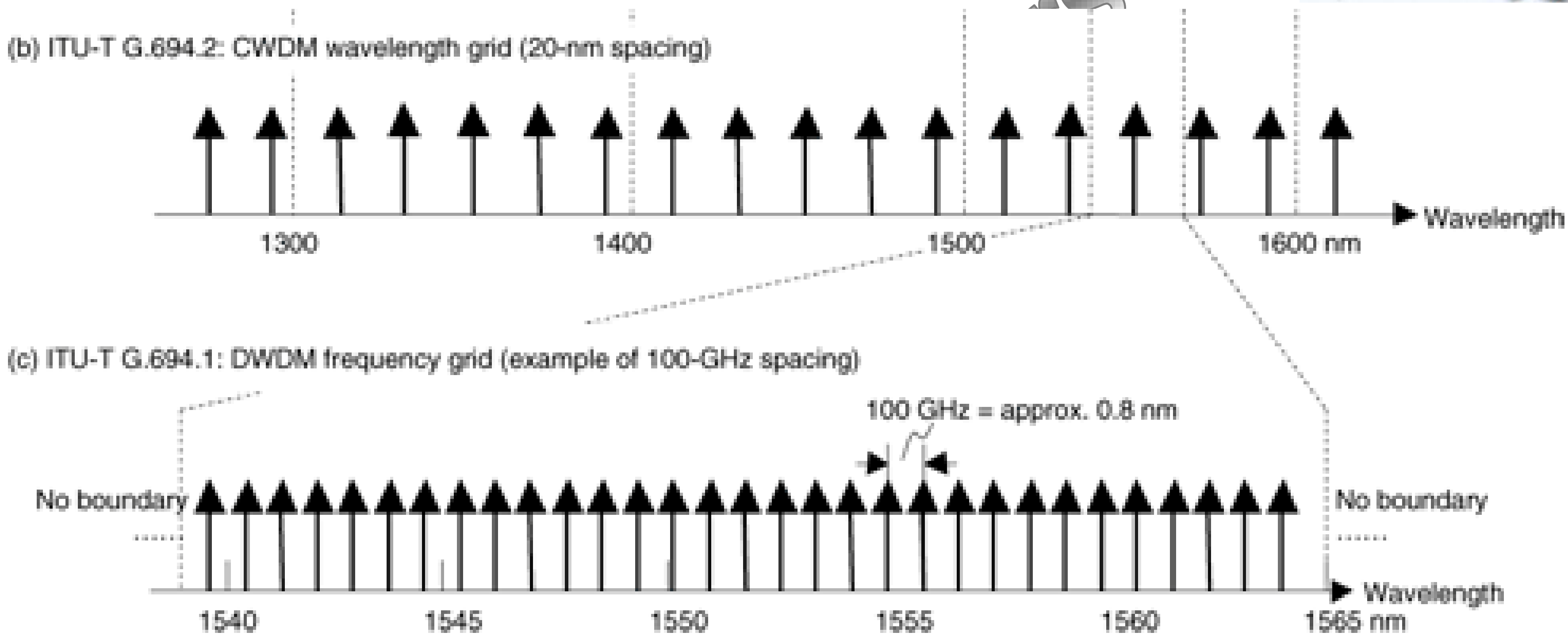
Wavelength Division Multiplexing

- **WDM is the use of multiple wavelengths of laser light (lambdas) to communicate multiple channels of information across optical fiber.**
- **There are two major categories of WDM:**
  - Coarse Wavelength Division Multiplexing (CWDM) with wide spacing between the lambdas. Intended for low cost, short range applications, usually on mulitmode fiber.
  - Dense Wavelength Division Multiplexing (DWDM) with relatively narrow spacing between lambdas. Intended for backbone applications. Limited to the S, C, and L bands because of the bandwidth limits of certain optical amplifiers.

ITU Standards for WDM

- The ITU (G.692.2) defines Coarse WDM as operating between 1271nm and 1611nm with 20nm spacing, resulting in 18 channels.
- Dense WDM is defined (G.694.1) as being centered at 193.10 THz (1552.52nm), with 100 GHz spacing. No defined number of channels, depends on optical bands used.
- Note that CWDM is spaced in nm while the DWDM is spaced in Hz.
- We're going to concentrate on DWDM in this discussion.

The ITU Grids for WDM

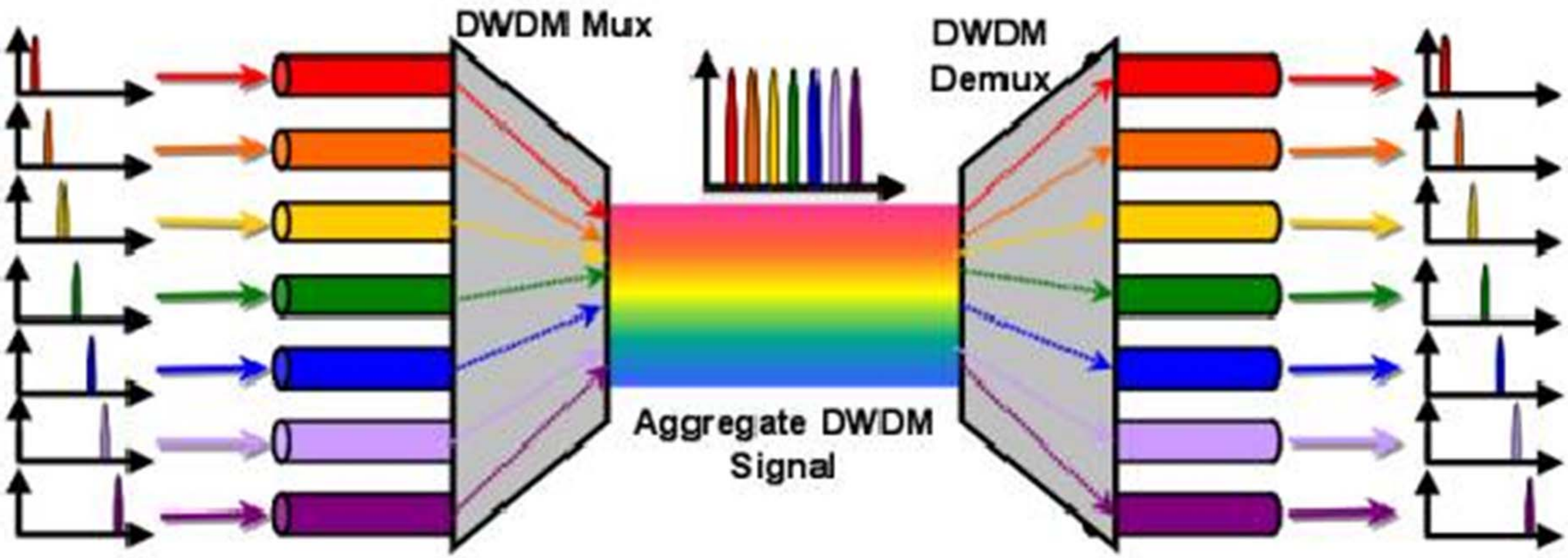


Coarse Wavelength Division Multiplexing

- CWDM was defined to allow low cost WDM, usually for short distance transmission, perhaps within a server facility or between buildings on a campus.
- Can use LEDs or lower cost Fabry-Perot lasers for each lambda because of the wide lambda spacing (not exact frequency and may not be stable with temperature and age).
- Can use lower cost components for muxing and demuxing the lambdas onto the fiber.
- Will usually be used with mulitmode fiber – bigger core – for ease of inserting the lambdas onto the fiber.

DWDM

- Multiple signals are input, each on a different wavelength, called a lambda.
- The lambdas are combined on the fiber and transmitted to the receiver, where they are demuxed to separate the lambdas.



Dense Wavelength Division Multiplexing

- **Note that the ITU centered their recommendation for DWDM in the 1550nm band.**
- **This is because DWDM is primarily used on single mode fiber for long distance transmission.**
- **And the most common optical amplifier, the erbium doped fiber amplifier, operates in the 1550nm band.**
- **The ITU recommendation specifies 100GHz spacing between lambdas but companies may use more or less. Some use as little as 25GHz, while others may use 200GHz.**

Lambda Spacing

- A modulated signal requires bandwidth for transmission.
- For On/Off Keying (OOK), the information signal is modulated onto the laser frequency. OOK is a form of Amplitude Modulation (AM).
- The NRZ signal carries two bits per Hz.
- This produces two sidebands, each with bandwidth equal to the information signal.
- So a 10Gbps signal requires at least 10 GHz spacing, preferably more for guard bands.
- A 40 GHz signal would require at least 40 GHz of bandwidth.

Impairments in DWDM - Linear

- **Adjacent channel crosstalk - Lambdas have bandwidth because the laser is modulated. Depending on how well the signal is filtered before being put on the fiber, signals may extend into adjacent lambdas which will degrade their signals. Also depends on the spacing of the lambdas.**

Impairments in DWDM – Non Linear

- Usually important at higher bit rates, 10Gbps and above.
- Stimulated Brillouin Scattering (SBS) – within a single channel, shorter wavelengths transfer power to longer wavelengths, causing spreading of the pulse.
- Stimulated Raman Scattering (SRS) – Power is transferred from a higher frequency (shorter wavelength) lambda to a lower frequency (longer wavelength) lambda.
- Four Wave Mixing (not bit rate dependent) – Interaction between lambdas causes additional frequencies to appear in the fiber. Affected by channel spacing and fiber dispersion. Reduced by spacing lambdas further apart and by irregular spacing of lambdas.

Impairments in DWDM – Non Linear

- **Self Phase Modulation (SPM) – Occurs even when there is only one lambda in a fiber. Causes dispersion spreading of the pulse.**
- **Cross Phase Modulation (XPM) – Similar to Self Phase Modulation but caused by interaction between the signals on different lambdas. Causes dispersion spreading of the pulse.**
- **As the power into an optical fiber increases, the fiber begins to act like a non-linear device. This limits the number of lambdas and the amount of power for each. This is especially true for SPM and XPM**

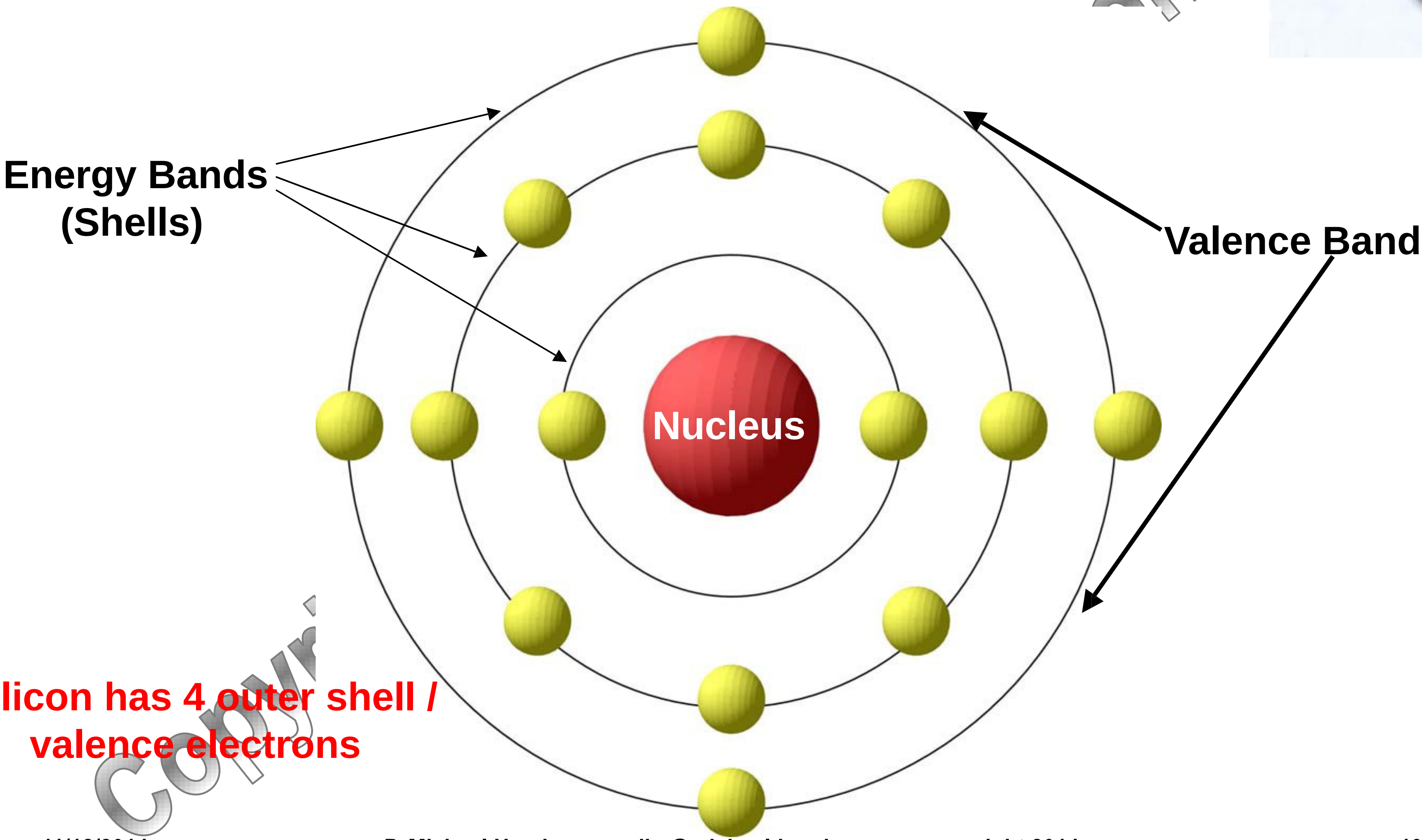
Fiber Optic Components

- **The components which we'll examine are:**
  - **LEDs (Light Emitting Diodes)**
  - LASERs (I'll write this as "laser" from here on).
  - Photodetectors
  - Semiconductor optical amplifier
  - Erbium doped fiber amplifier
  - Raman amplification
  - DWDM mux and demux

First, Some Physics

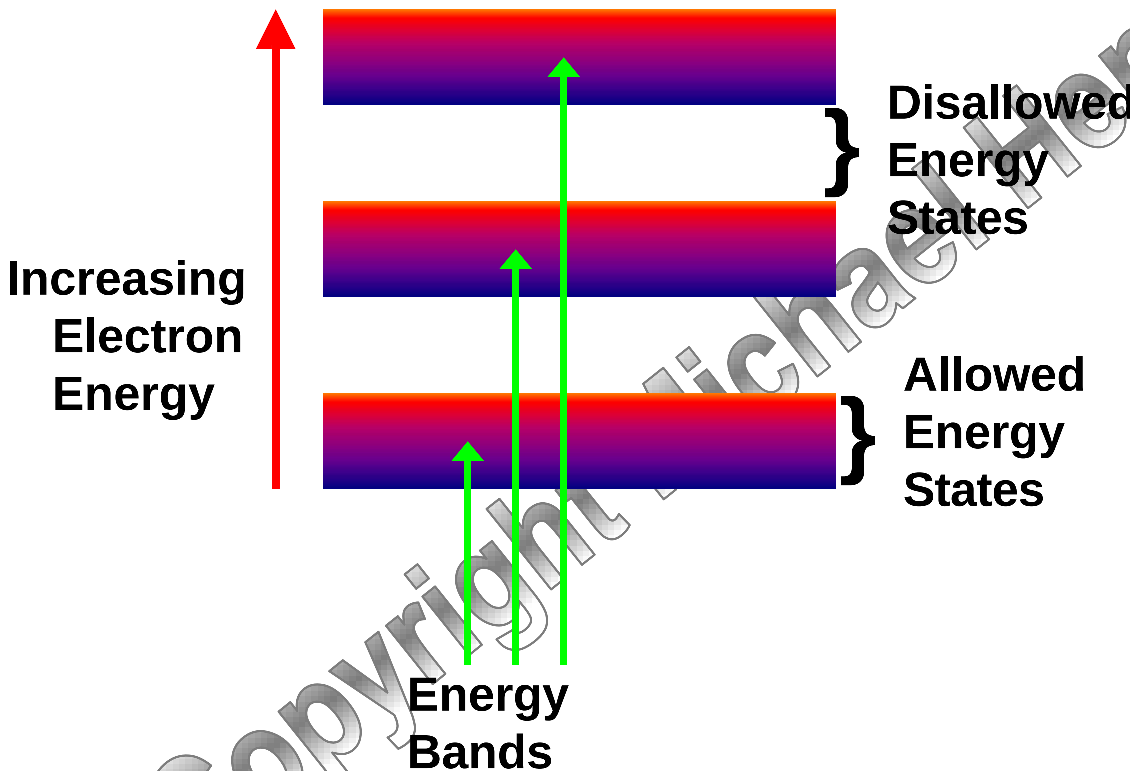
Silicon is the primary semiconductor material used in VLSI systems

Silicon has 14 electrons

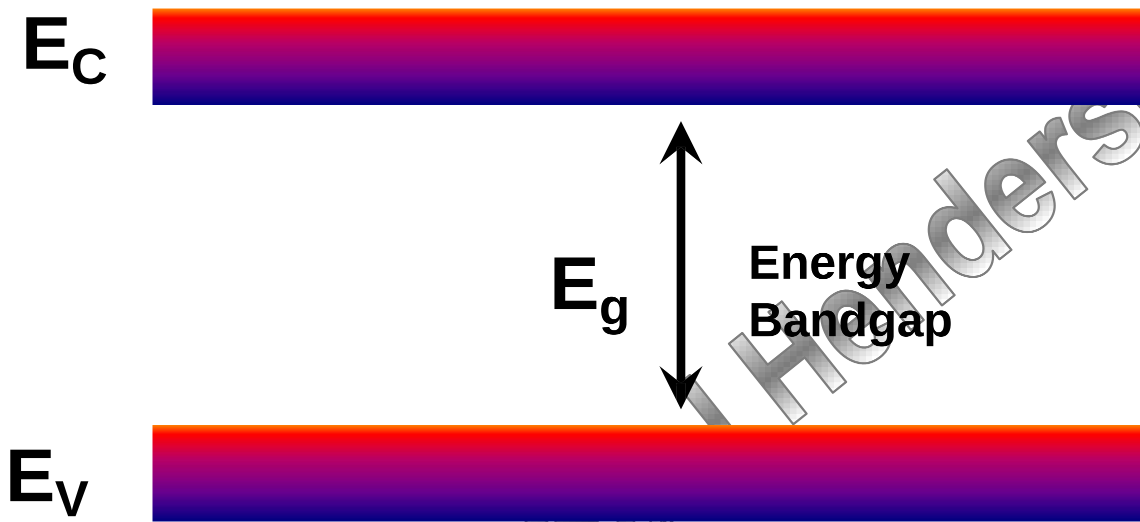


Silicon has 4 outer shell / valence electrons

Energy Bands



- Electrons try to occupy the lowest energy band possible
- Not every energy level is a legal state for an electron to occupy
- These legal states tend to arrange themselves in bands



Conduction Band

First unfilled energy band at T=0K

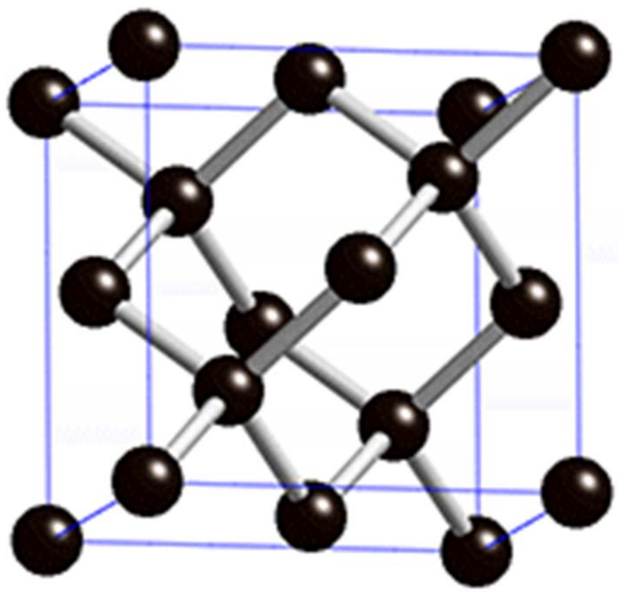
Valence Band

Last filled energy band at T=0K

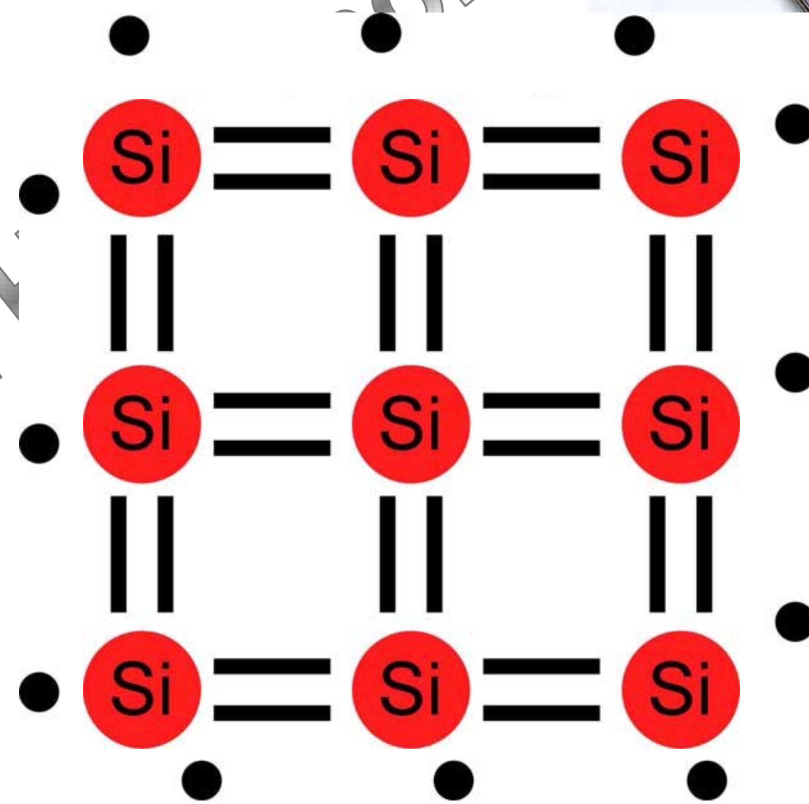
- E_c is the lowest energy state of a free electron.
- E_v is the highest energy state for filled shells.
- E_g is the difference in energy between the E_c band and the E_v band.

Silicon Crystal Structure

- **Silicon, with 4 valence electrons in the valence shell, form crystal structures as shown.**

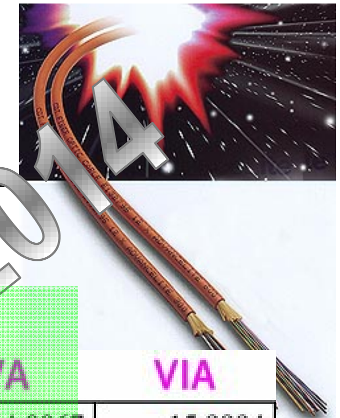


3-dimensional view



2-dimensional view

Doping

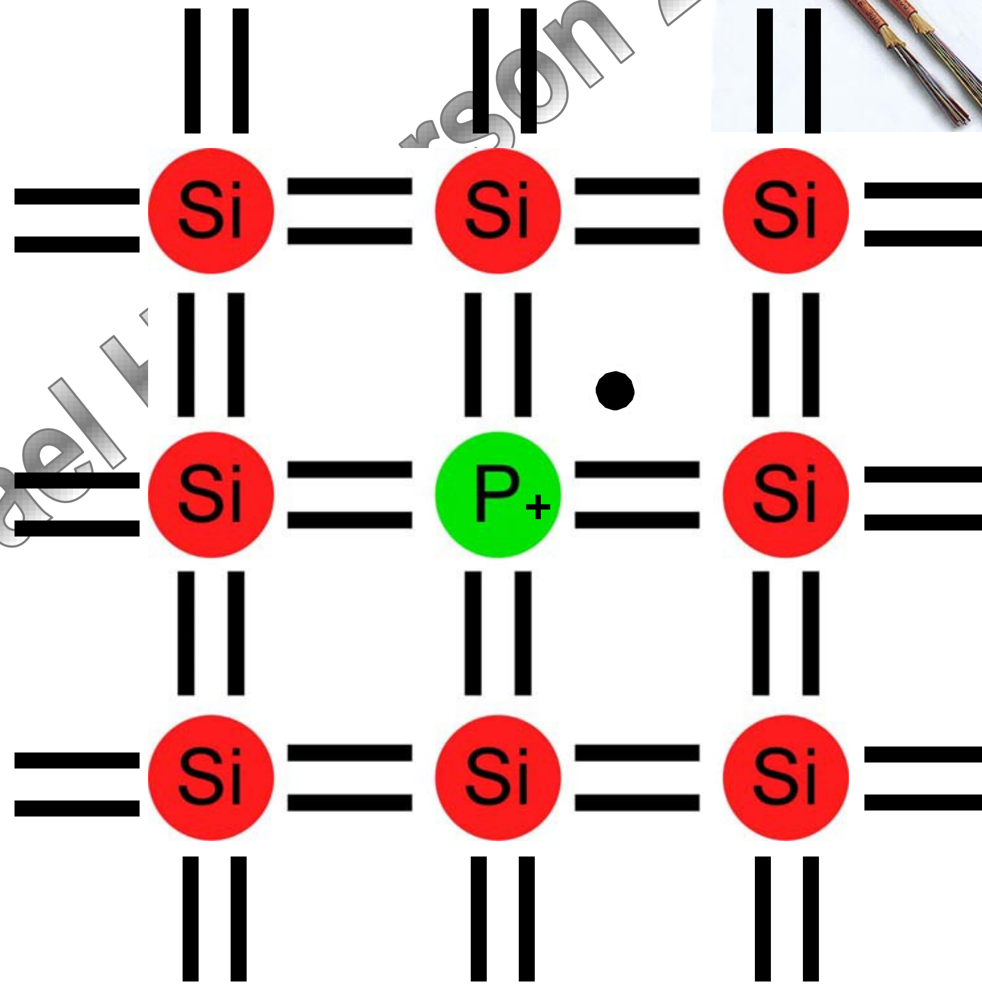


- Silicon can be made into a semiconductor by doping it with certain elements.
- Elements from the IIIA column have 3 electrons in the valence shell.
- Elements from the VA column have 5 electrons in the valence shell.

IIB	IIIA	IVA	VA	VIA
	5 10.811 **B** Boron	6 12.01115 **C** Carbon	7 14.0067 **N** Nitrogen	8 15.9994 **O** Oxygen
	13 26.9815 **Al** Aluminum	14 28.086 **Si** Silicon	15 30.9738 **P** Phosphorus	16 32.064 **S** Sulfur
30 65.37 **Zn** Zinc	31 69.72 **Ga** Gallium	32 72.59 **Ge** Germanium	33 74.922 **As** Arsenic	34 78.96 **Se** Selenium
48 112.40 **Cd** Cadmium	49 114.82 **In** Indium	50 118.69 **Sn** Tin	51 121.75 **Sb** Antimony	52 127.60 **Te** Tellurium
80 200.59 **Hg** Mercury	81 204.37 **Tl** Thallium	82 207.19 **Pb** Lead	83 208.980 **Bi** Bismuth	84 (210) **Po** Polonium

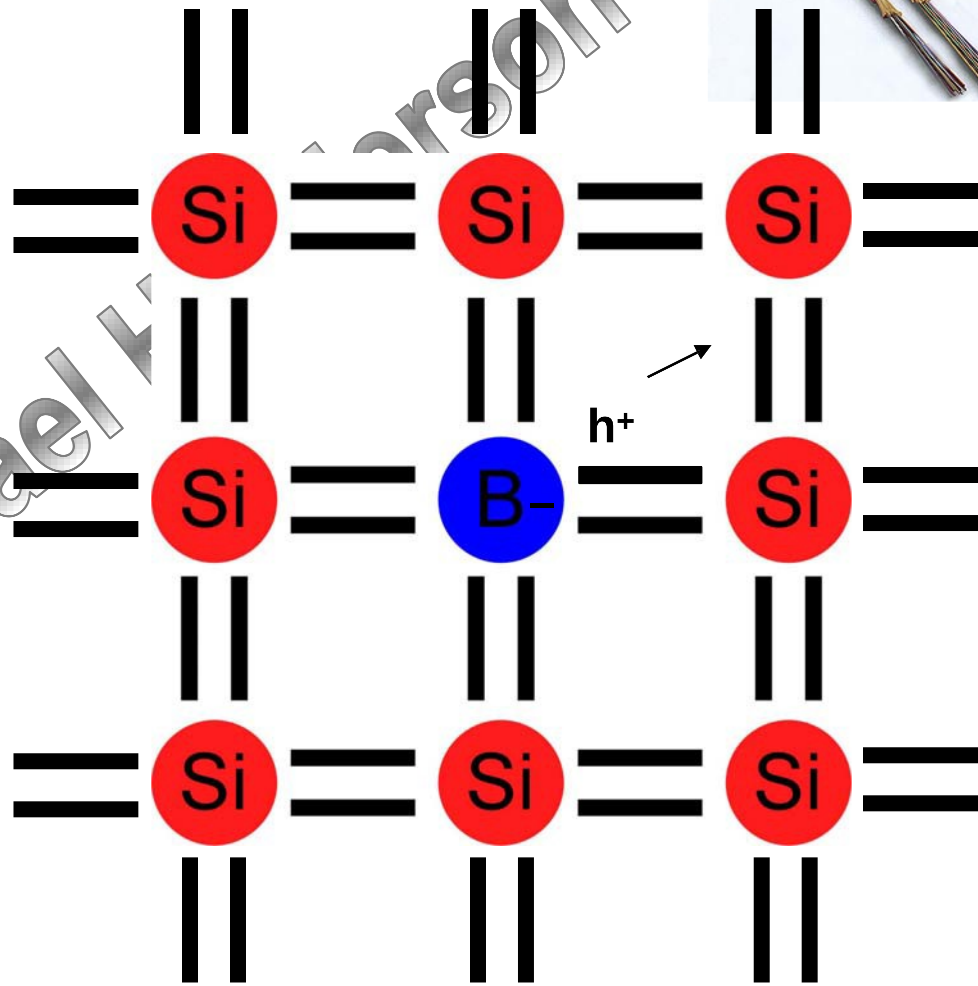
N-type Material

- If phosphorus is added to the silicon, it will take the place of a silicon atom in the crystal, but an electron will be left over.
- There will be free electrons in n-type material.
- When the electron migrates away, what's left is a positively charged atom.



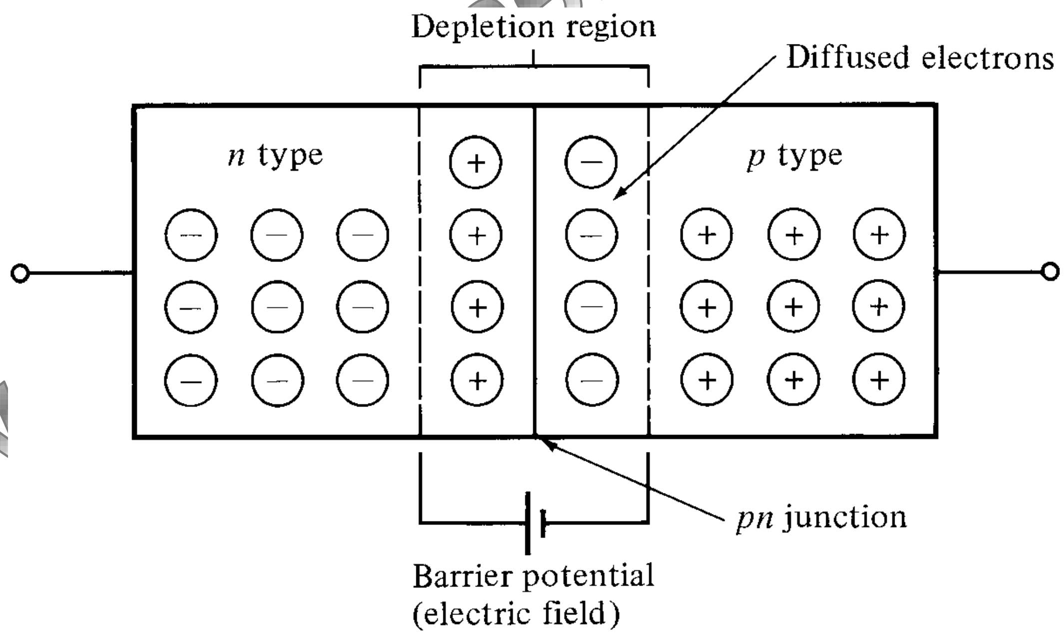
P- Type Material

- If boron is added to the silicon, it will take the place of a silicon atom in the crystal, but an electron will be missing – there will be a hole.
- There will be free holes in p-type material.
- When an electron fills the hole, what's left is a negatively charged atom.



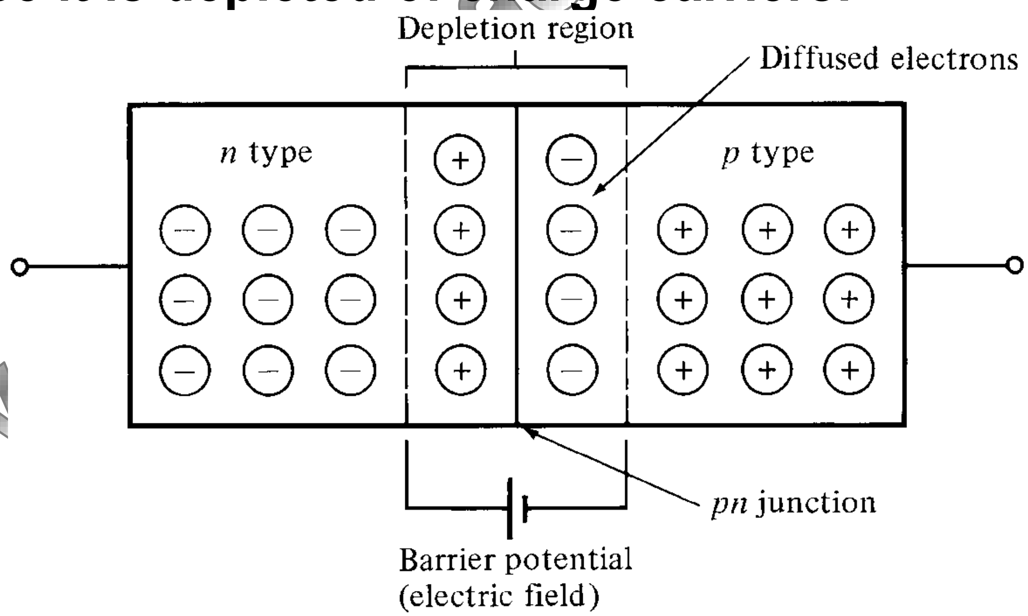
Junctions

- When silicon is alternately doped, we form a pn junction between the n-type material and the p-type material.
- The extra electrons in the n-type material migrate to the p-type material and fill the holes in the valence bands.
- When they do, the n-type atoms are left with a positive charge and the p-type atoms have a negative charge.



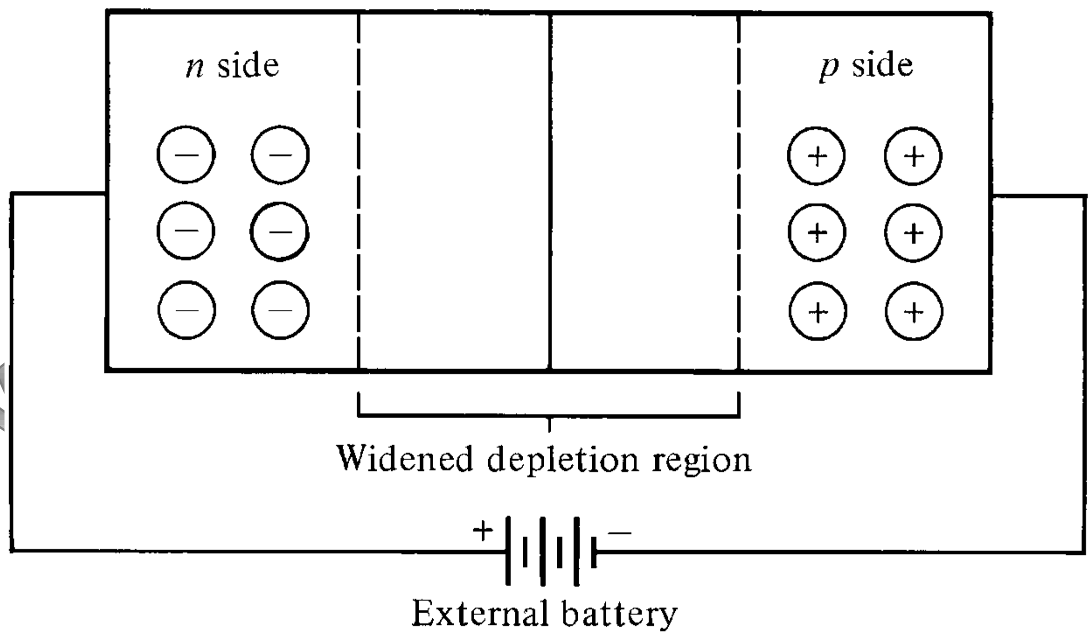
Junctions

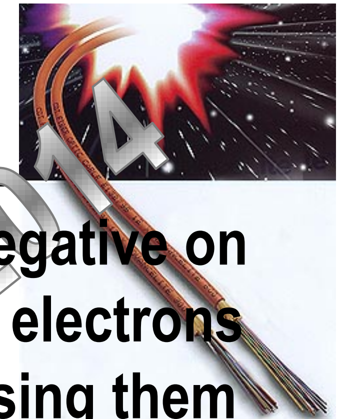
- Because of the charges on the atoms, an electric field is set up between the n-type region and the p-type region.
- The electron migration will continue until the electric field between the charged atoms is sufficient to block any further migration.
- The area of charged atoms is known as the depletion region because it is depleted of charge carriers.



Reverse Bias

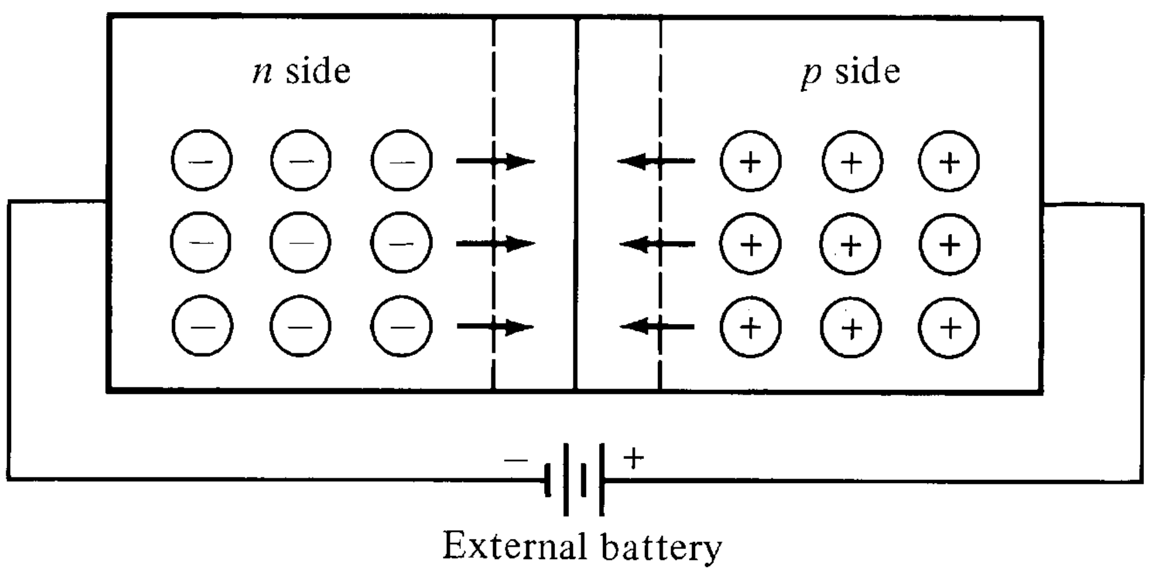
- If we apply a voltage across the semiconductor, with positive on the n-type side and negative voltage on the p-type side, we will pull the electrons from the n-type material and the holes from the p-type material.
- This will widen the depletion region until the electric field across the depletion region balances the voltage applied.





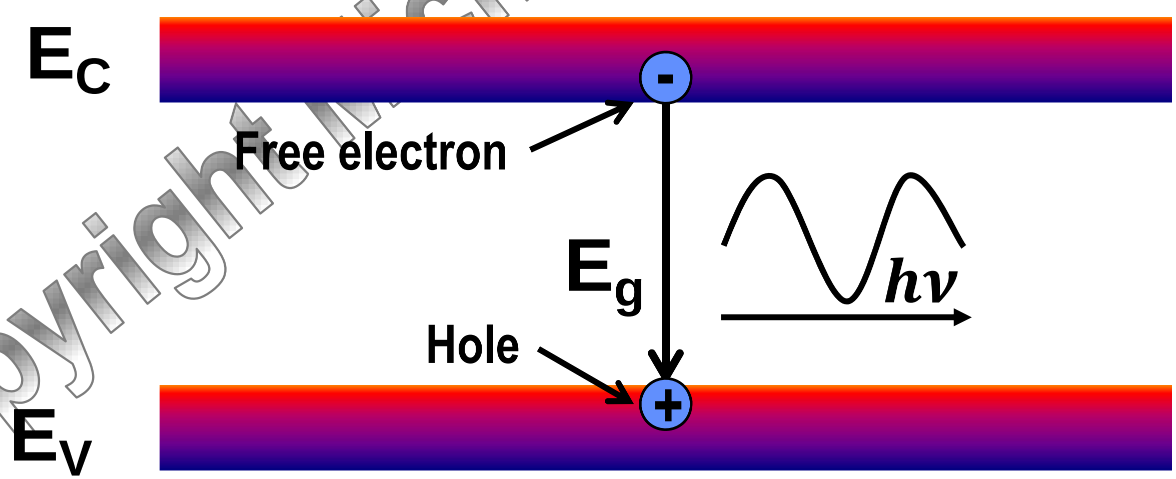
Forward Bias

- If we apply a voltage across the semiconductor, with negative on the n-type side and positive voltage on the p-type side, electrons will flow to the positive ions in the n-type material, causing them to become neutral atoms.
- Electrons will also be pulled from the negative ions in the p-type material.
- Electrons will continue to flow across the junction, falling into the holes in the p-type material.



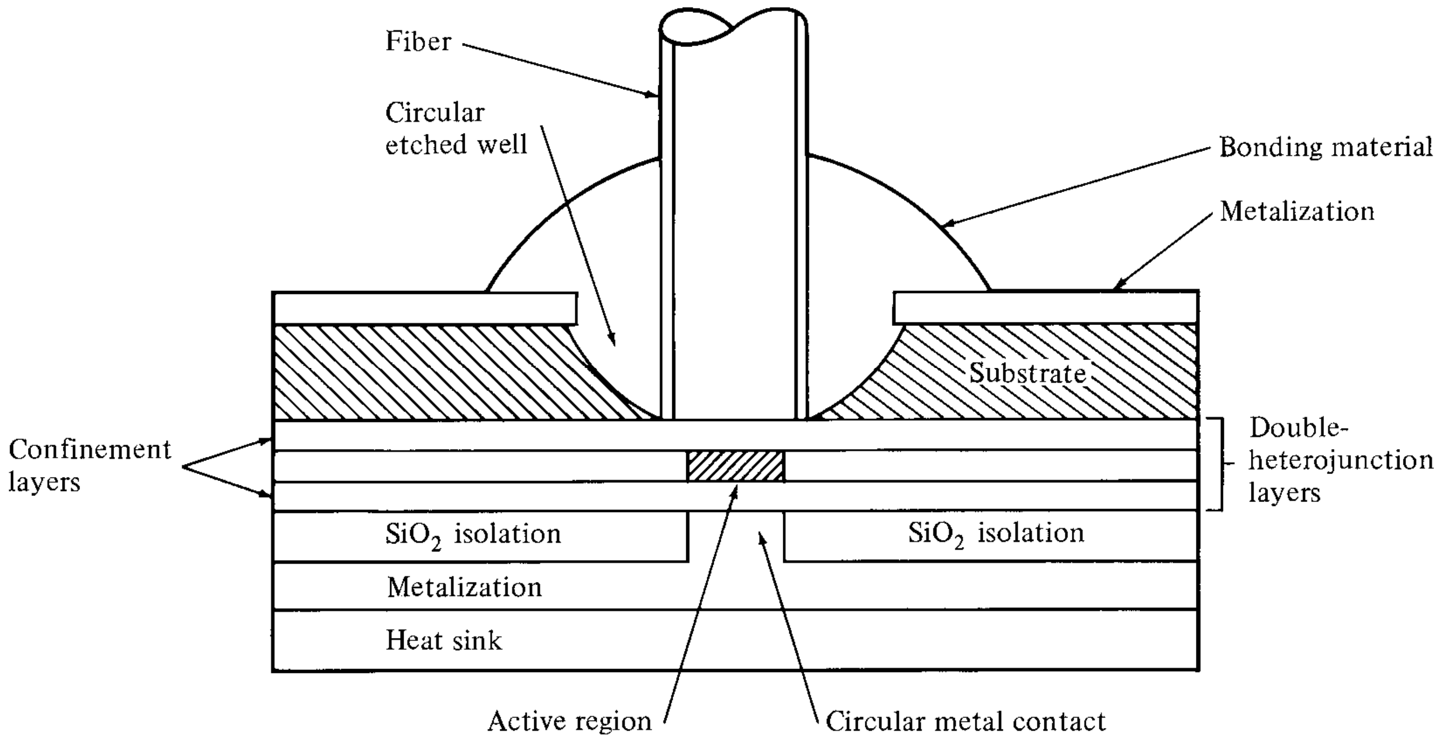
Photon Emission

- When a free electron encounters one of the column III atoms, it will fall to the valance band to fill the hole.
- But the electron has too much energy for that band and must give up E_g, the gap energy.
- It can do this by emitting a photon with that amount of energy, $h\nu$. (h is Planck's constant, and ν is the frequency).
- This is how an LED emits light.



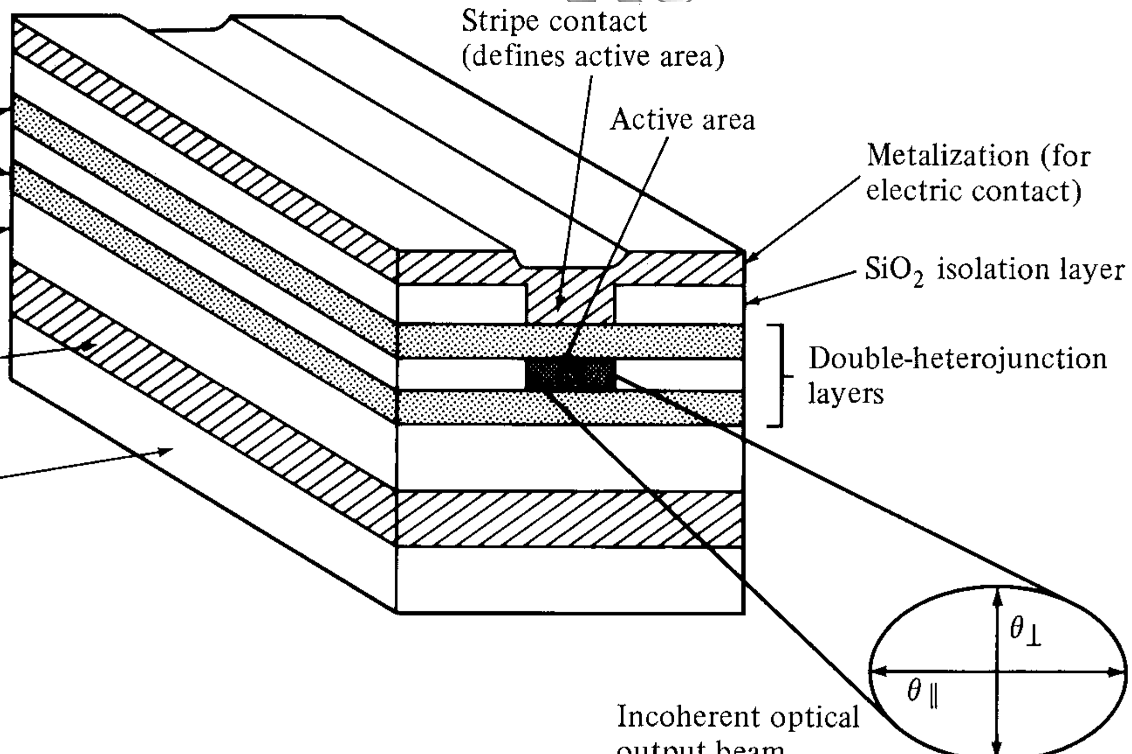
Surface Emitting LEDs (SLED)

- The substrate is etched and a fiber is attached over the active region.



Edge Emitting LEDs (ELED)

- The light exits the semiconductor on the side (edge).
- Edge emitting LEDs can be more powerful than SLEDs.



Advantages and Disadvantages of LEDs

- **LEDs are much less expensive than laser diodes.**
- **Can be easily modulated by controlling the injection current.**
- **But… they have a broad bandwidth, about 35nm in the 800nm band, and between 70nm to 180nm in the 1300nm to 1600nm band.**
- **Mostly used for short distance applications on mulitmode fiber.**
- **Would be difficult to couple to single mode fiber.**

Fiber Optic Components

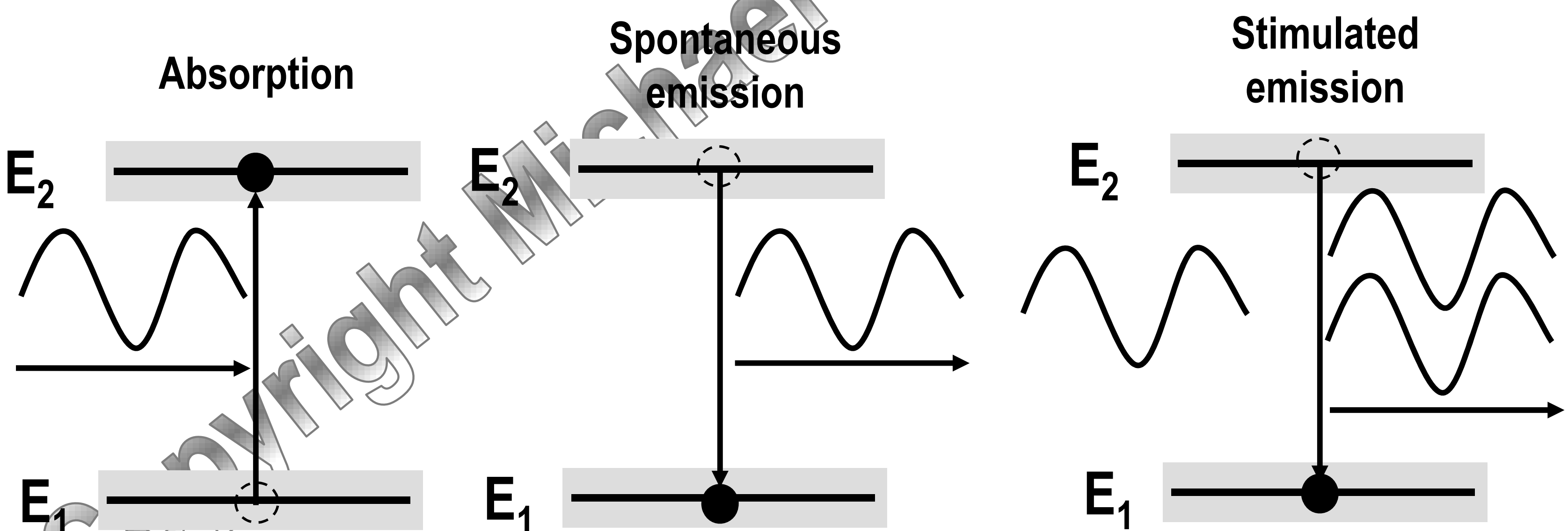
- **The components which we'll examine are:**
  - LEDs (Light Emitting Diodes)
  - **LASERs (I'll write this as "laser" from here on).**
  - Photodetectors
  - Semiconductor optical amplifier
  - Erbium doped fiber amplifier
  - Raman amplification
  - DWDM mux and demux

Laser Diodes

- **For long distance transmission, and for launching an optical signal into a single mode fiber, laser diodes are preferred.**

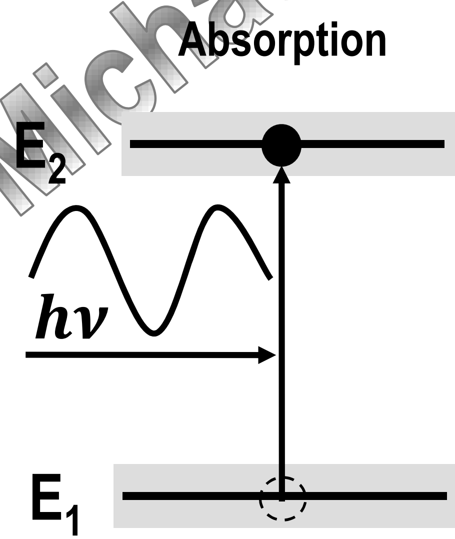
First, Some Physics

- **There are three important atomic concepts related to Lasers**
  - Absorption of a photon.
  - Spontaneous emission of a photon.
  - Stimulated emission.



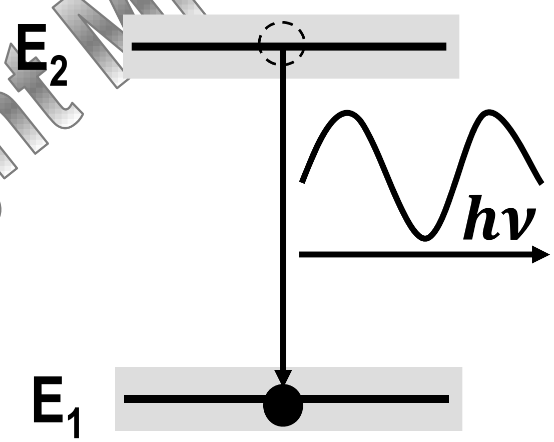
Absorption of a Photon

- Atoms have discrete "orbital shells" which electrons can inhabit.
- There is an energy level associated with each orbital.
- For an electron to move to a higher orbital, it must absorb a photon which contains that much energy (that quanta of energy).



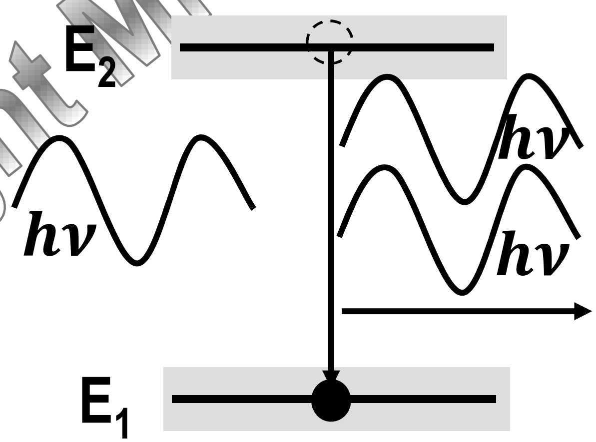
Spontaneous Emission

- Electrons will generally only stay in a higher energy shell for a limited amount of time.
- When they return to the "ground state" they emit a photon that has the energy equal to the difference between the higher orbital and the "ground state" orbital.
- In doing so, the photon emitted will be emitted in a random direction.



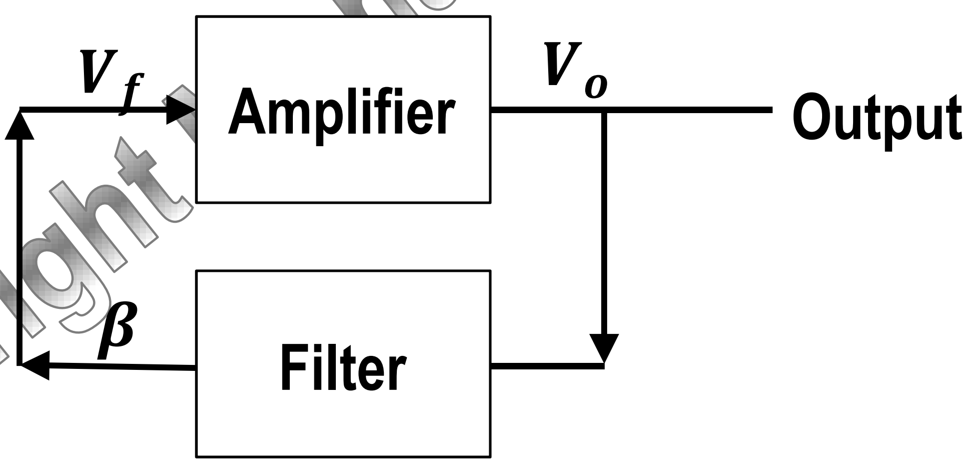
Stimulated Emission

- When an electron is in an excited state, it can be "stimulated" to emit a photon by interaction with another photon of the proper energy.
- When it does so, the emitted photon is exactly the same frequency, phase, and direction as the photon that that stimulated the emission.
- Einstein proposed stimulated emission in a paper in 1917.



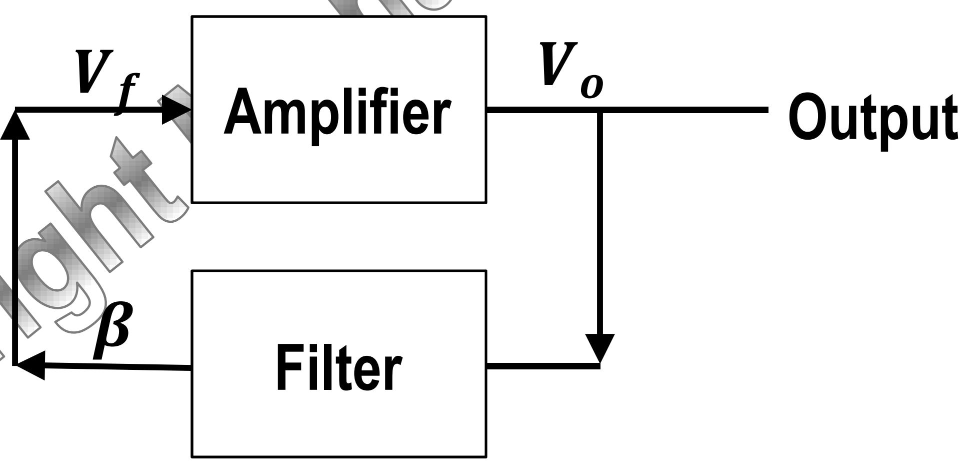
Oscillators

- An oscillator is a system which includes a gain stage and a method of providing positive frequency selective feedback.
- When power it applied, the amplifier will begin amplifying noise, which is broad band.
- A selected frequency will be fed back through the filter, amplified and fed back again.



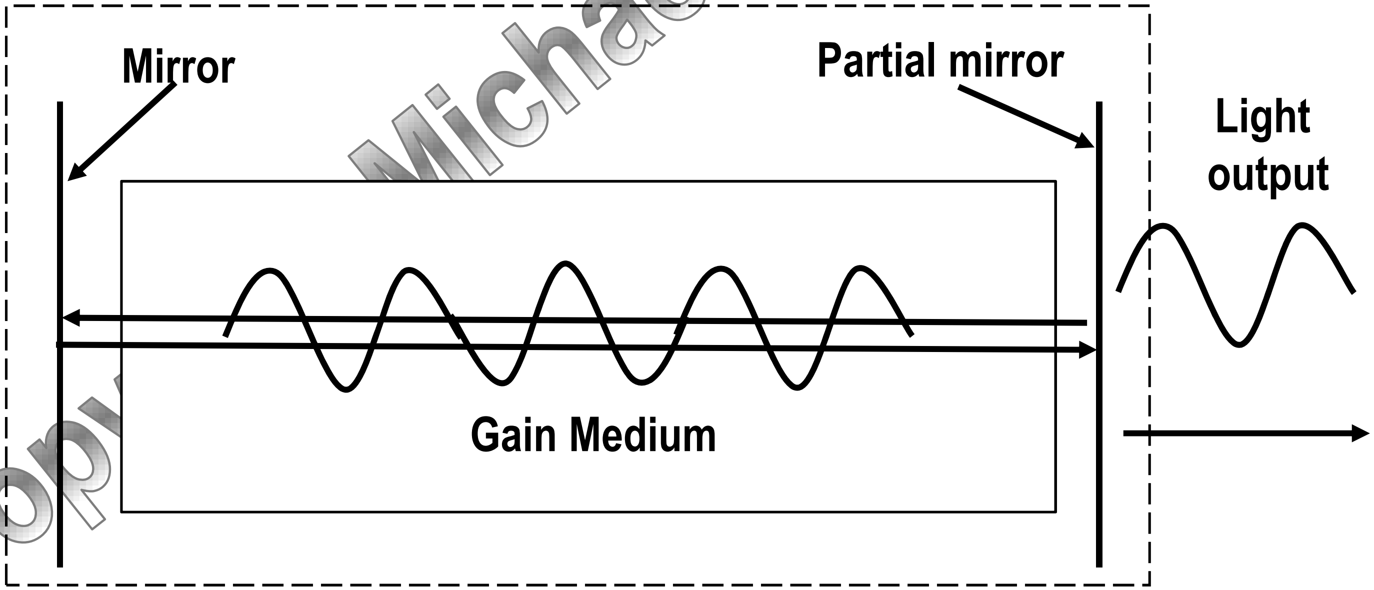
Oscillators

- The output will become a sinusoidal wave of the frequency passed by the filter.
- The amplifier will eventually saturate which will limit the amplitude of the output signal.



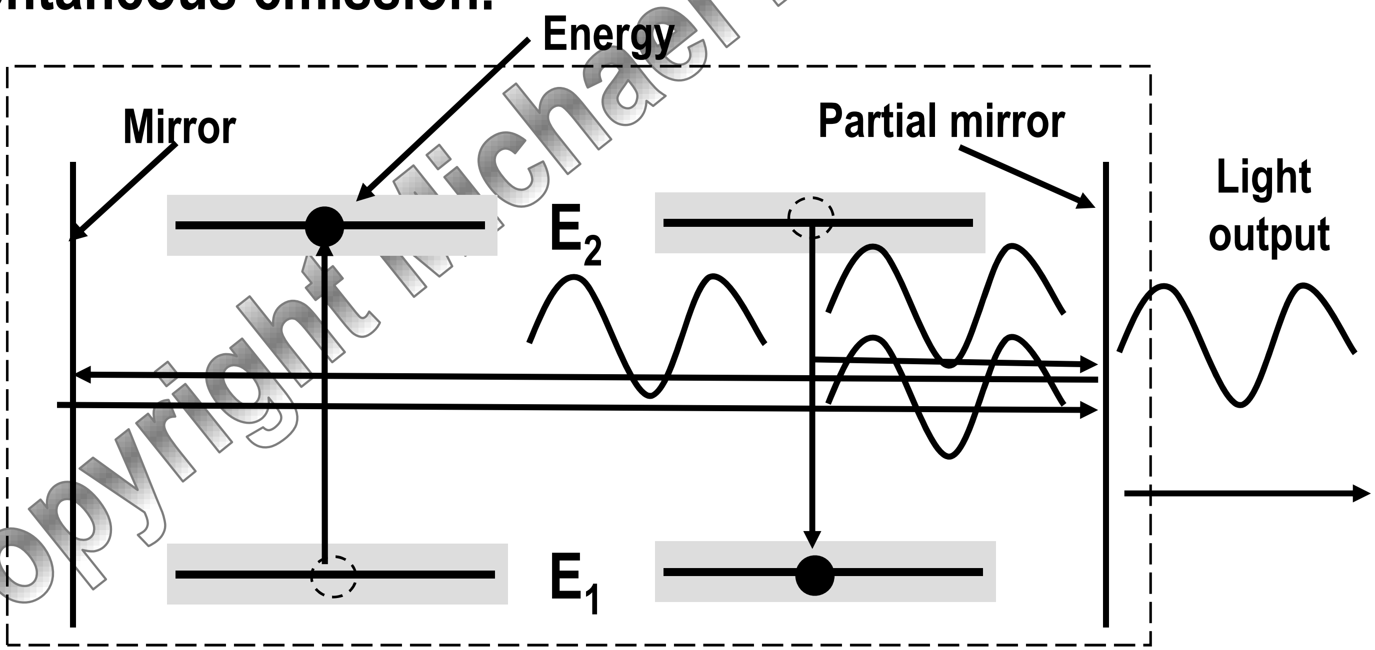
Laser

- A Laser is an optical oscillator consisting of a gain medium and a frequency selective feedback mechanism.
- Photons travel through the gain medium, and bounce back and forth, being amplified each time.
- The length of the cavity (the distance between the mirrors) determines the frequency.



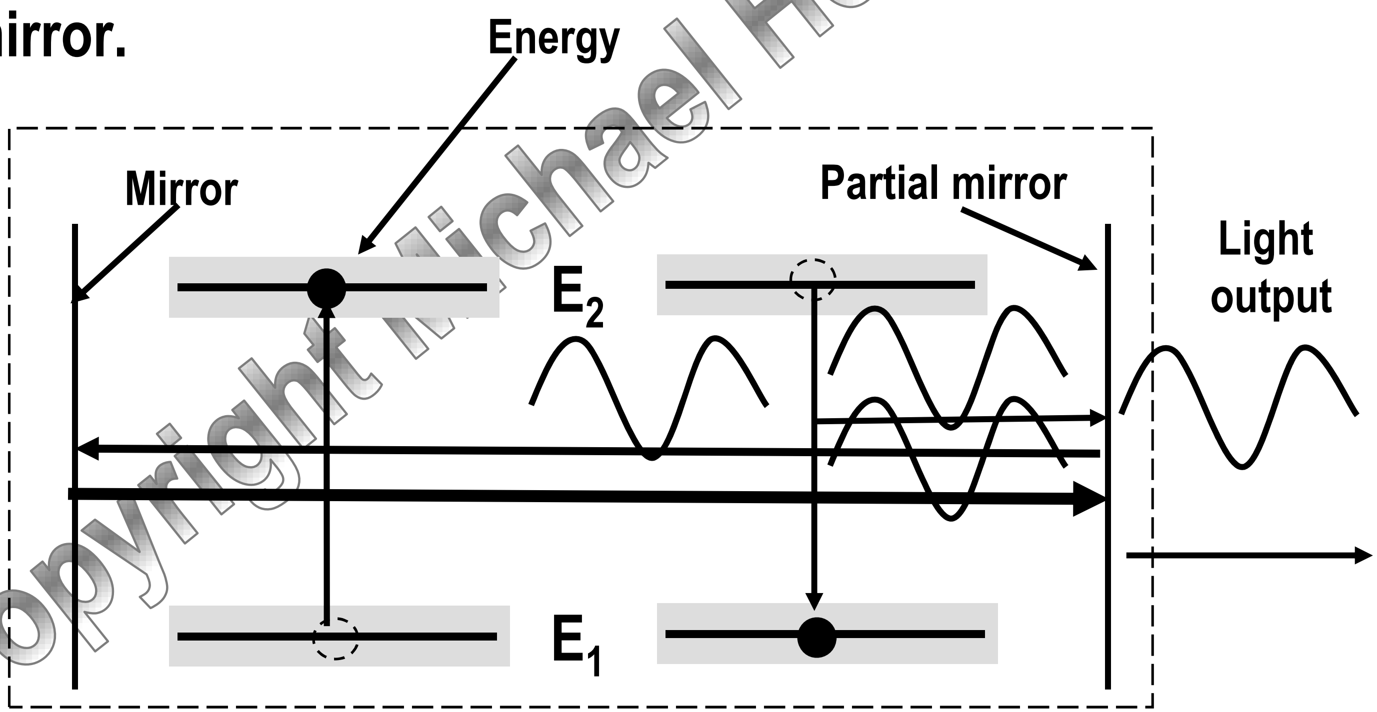
A Laser Oscillator

- Combining these ideas, we can see how a Laser works.
- First, energy is put into the gain medium, which raises the electrons in atoms to a higher orbital state. It creates a "population inversion".
- Then some of the atoms will release a photon through spontaneous emission.



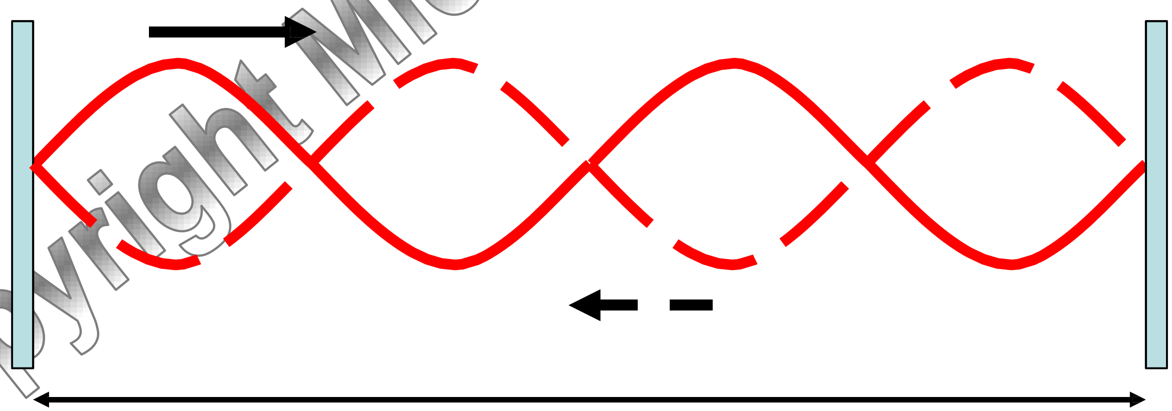
A Laser Oscillator

- Those photons that are emitted directly towards the ends will be reflected back and forth through the gain medium, causing stimulated emission.
- The cavity length will determine the optical frequency.
- Some of the light leaves the gain medium through the partial mirror.

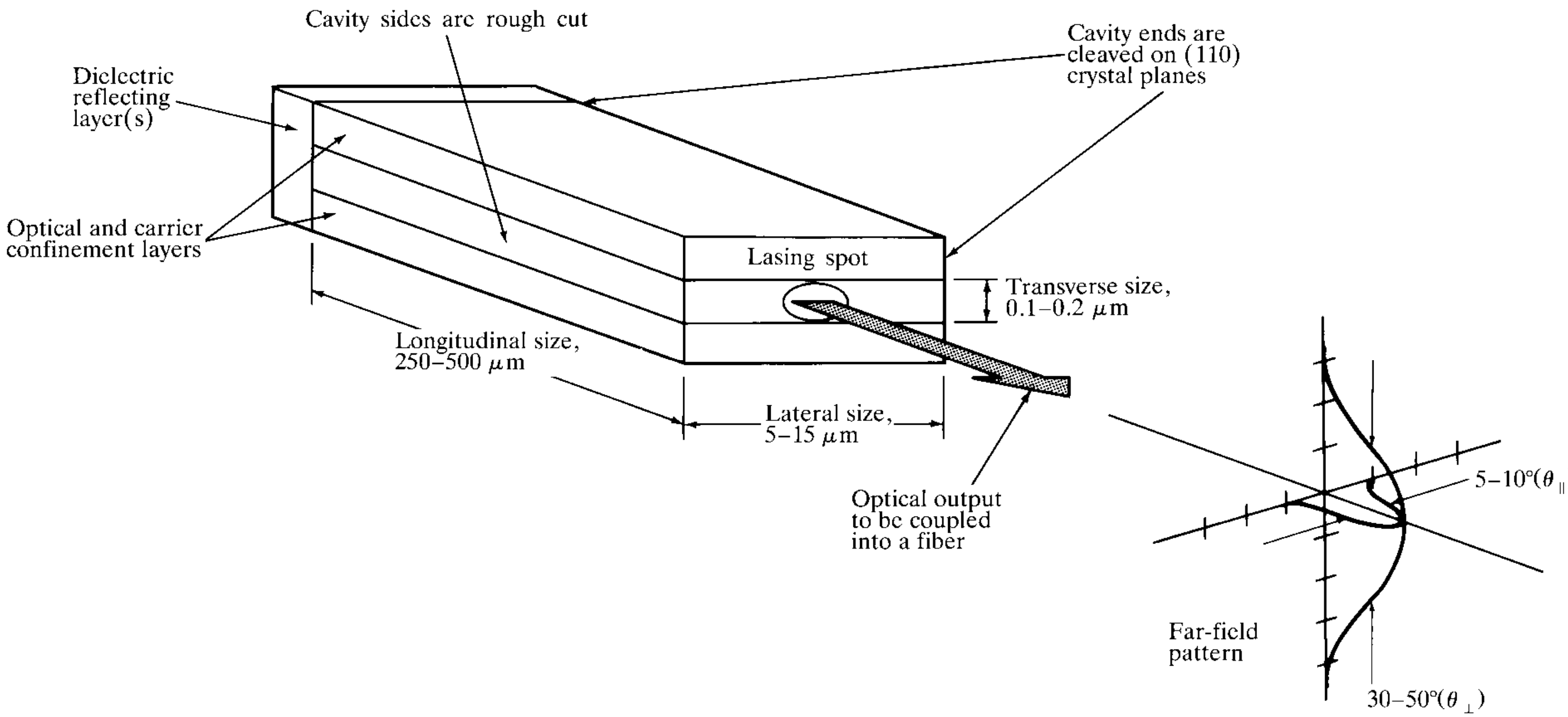


Fabry-Perot Laser

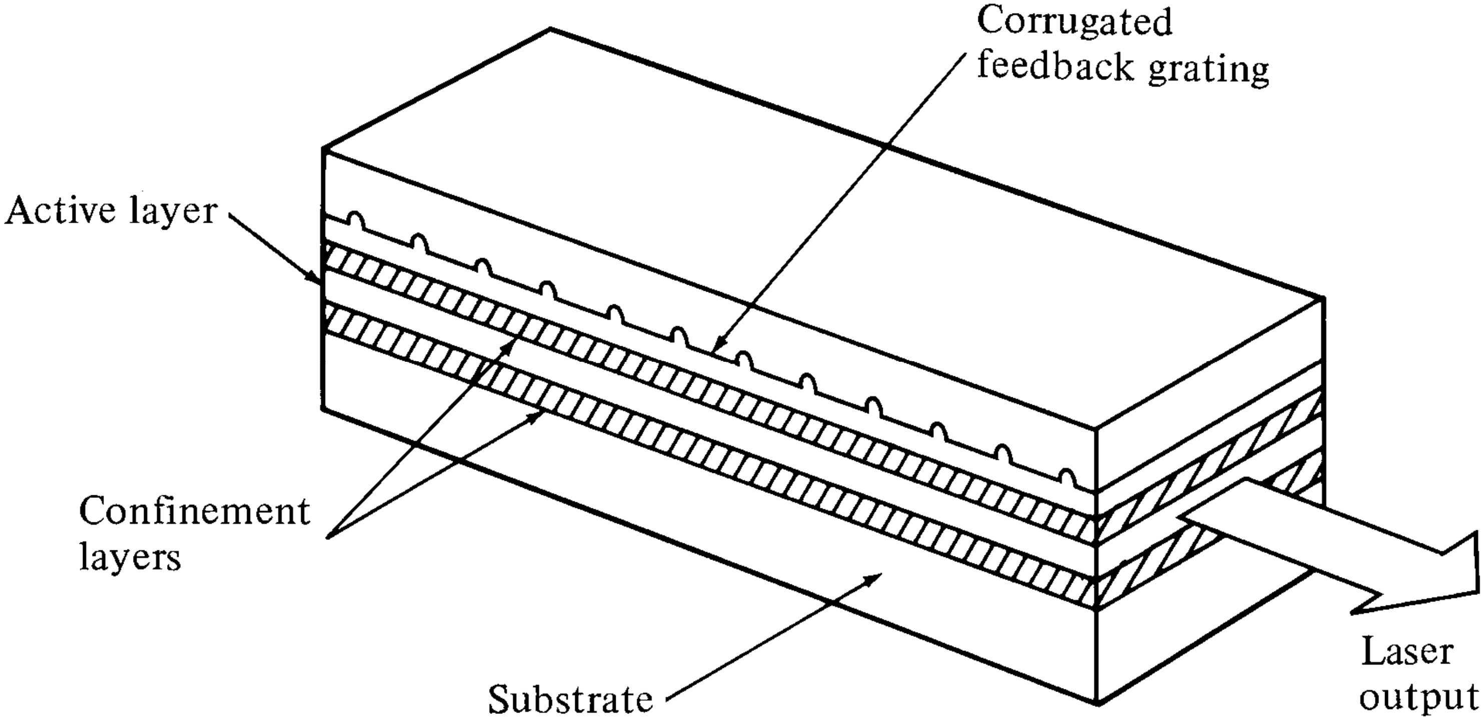
- Only certain wavelengths will be amplified in a laser.
- Those wavelengths will form standing waves in the resonator cavity.
- Multiple wavelengths, of course, can form standing waves.
- The wavelength centered on the gain curve of the active medium will grow the fastest while the others will fade out. Only a single mode will survive.



Fabry-Perot Laser



Distributed Feedback Laser (DFB)



Modulation of the Optical Signal

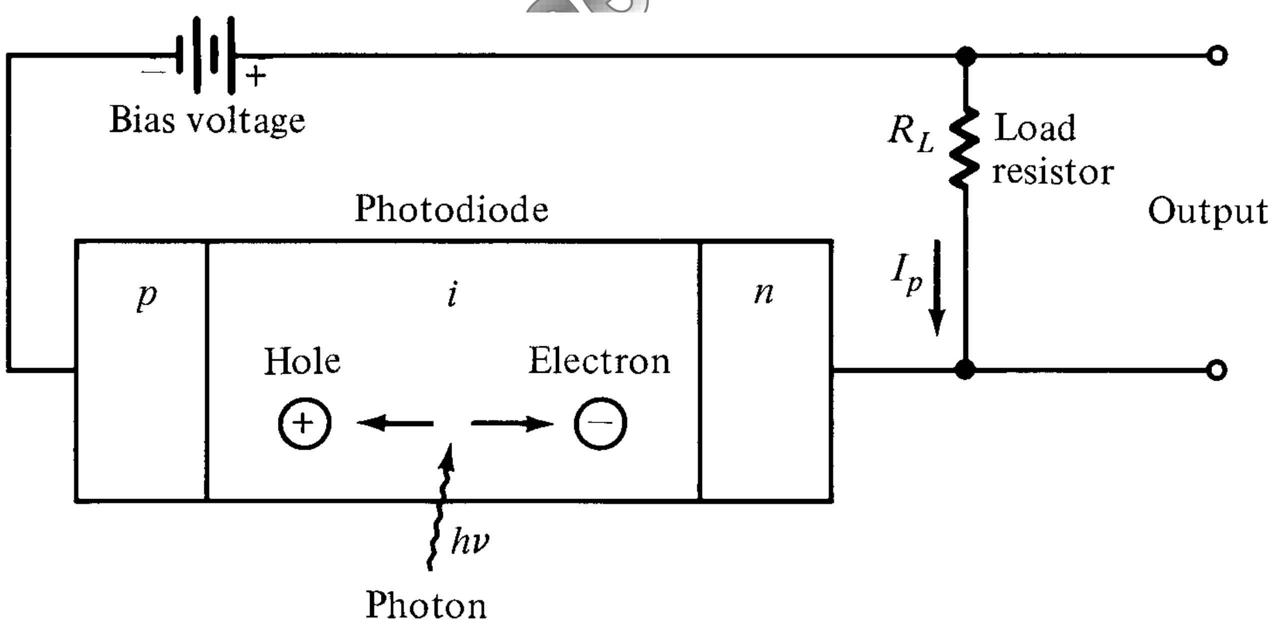
- The diode laser can be modulated via the laser drive current.
- This is called direct modulation.
- Only practical for lower speed links, 2.5Gbps or less.
- For higher data rates, “chirp” – widening of the laser line width – is a problem, so external modulation of a steady laser output is used.
- These external modulators can be electro-optical or electro-absorption.

Fiber Optic Components

- **The components which we'll examine are:**
  - LEDs (Light Emitting Diodes)
  - LASERs (I'll write this as "laser" from here on).
  - **Photodetectors**
  - Semiconductor optical amplifier
  - Erbium doped fiber amplifiers
  - Raman amplification
  - DWDM mux and demux

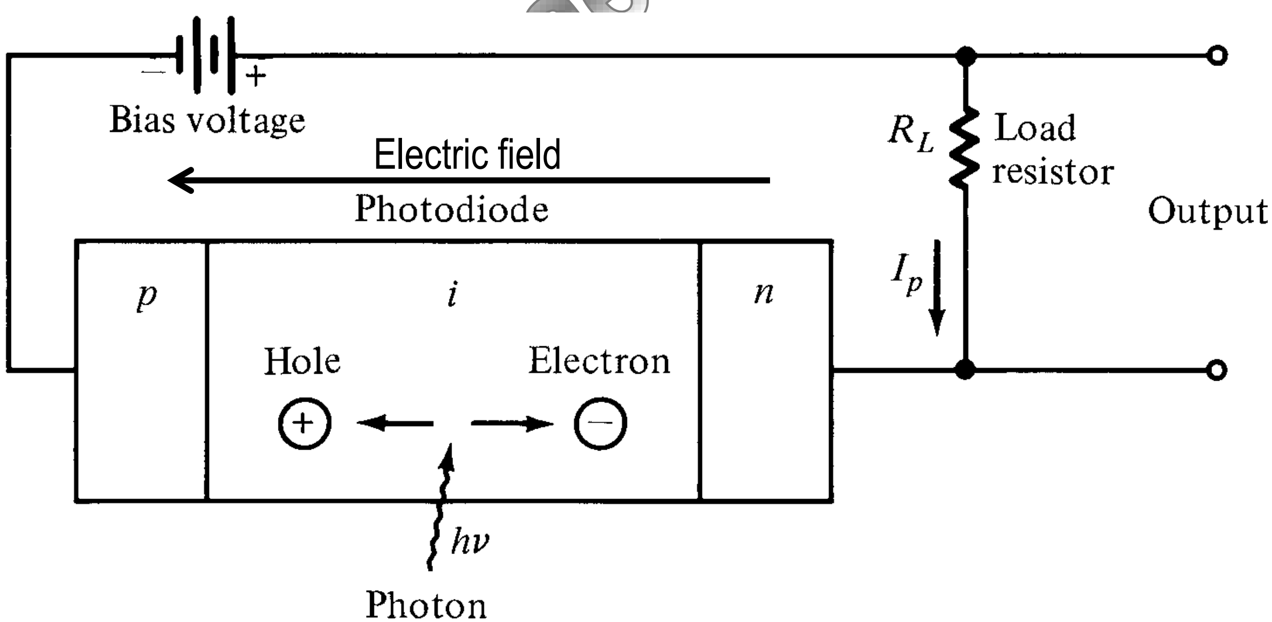
Photodetectors

- The most common photodetector is the PIN diode.
- It consists of a highly doped p-type material, a very lightly doped region, called the intrinsic layer, and a highly doped n-type material.
- A reverse bias is applied sufficient to deplete the intrinsic layer of any charge carriers.



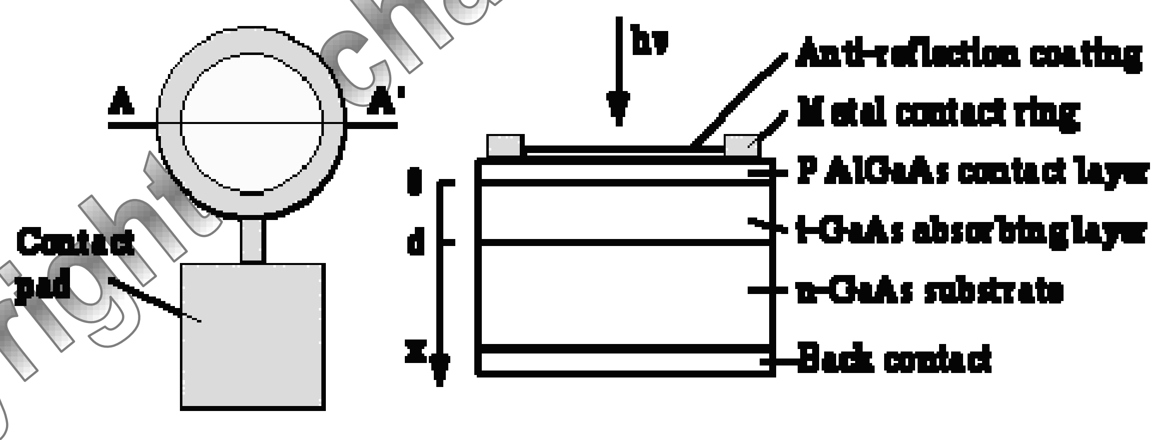
PIN Diode

- When a photon with energy $\geq$the band gap energy is incident on an electron, it can cause the electron to transition from the valence band to the conduction band.
- This creates a hole and an electron which migrate to the ends under the influence of the electric field in the intrinsic layer.



PIN Diode

- The top layer is p-type and transparent.
- There is a metal ring around the P-type layer to provide electrical contact.
- The intrinsic layer is where the photons are captured.
- Two important characteristics are quantum efficiency and response speed.

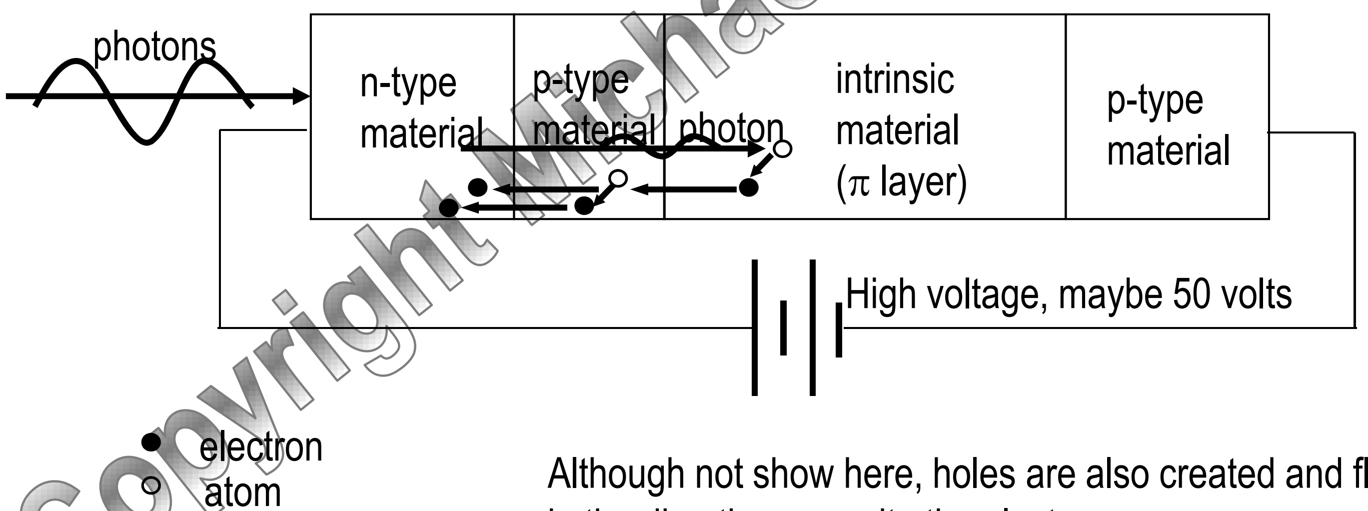


Top view

Vertical structure

Avalanche Photodiodes (APD)

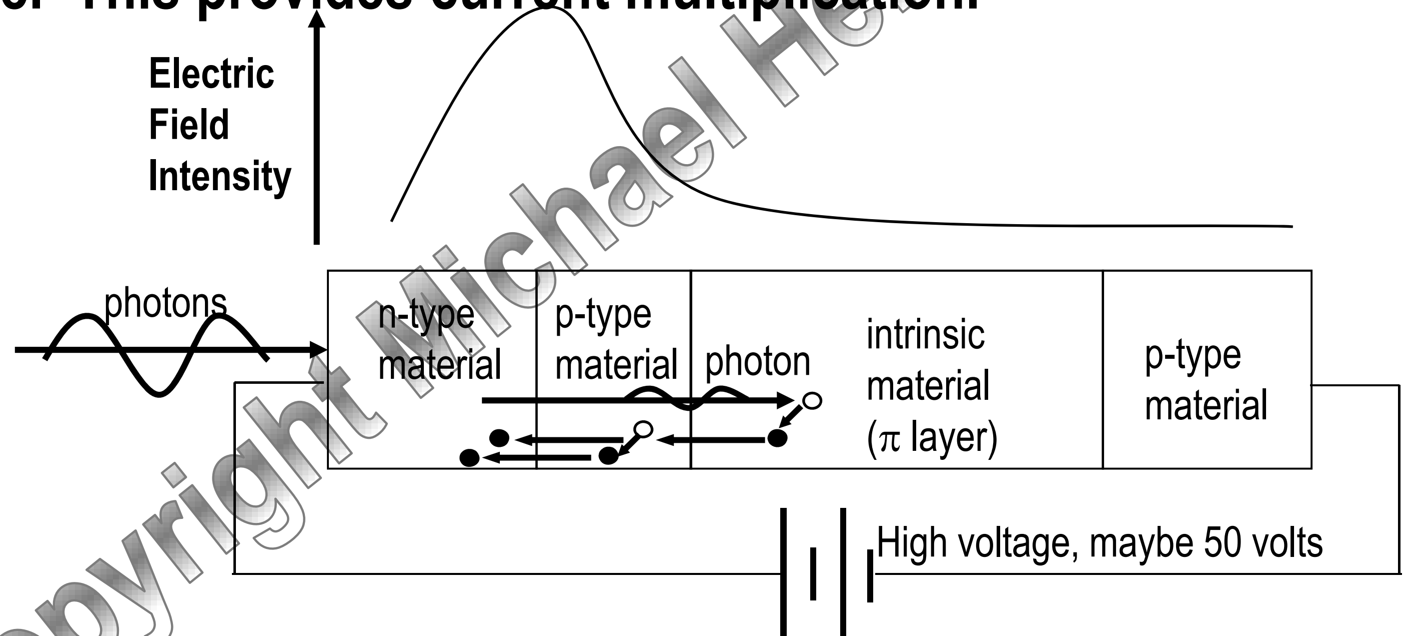
- An avalanche photodiode (APD) can be thought of as a photodetector with a build in amplification stage.
- It is similar to the PIN diode but with the addition of an additional p-type layer between the n-type layer and the intrinsic layer.
- The ADP is operated with a high reverse bias, as high as 50 volts.



Although not show here, holes are also created and flow in the direction opposite the electrons.

Avalanche Photodiodes

- **A high electric field is developed between the n-type layer and the p-type layer.**
- **When an electron is promoted by a photon, the electron accelerates in the electric field, knocking other electrons free. This provides current multiplication.**



● electron
○ atom

Although not show here, holes are also created and flow in the direction opposite the electrons.

Avalanche Photodiodes

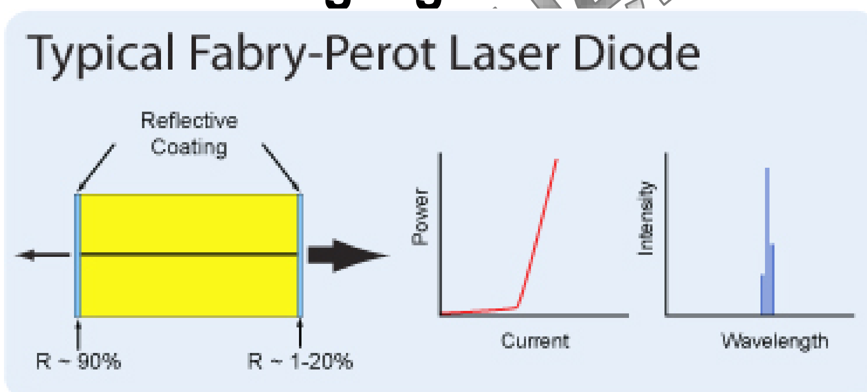
- **Avalanche Photodiodes have certain problems:**
  - The amount of amplification is limited. If too much is attempted, it will "run away" because of thermal electrons.
  - It's noisy because thermal electrons get amplified, also.
  - It takes time for the current to build so the response time is longer than for the PIN diode.
- **For high speed applications, the PIN diode is preferred.**

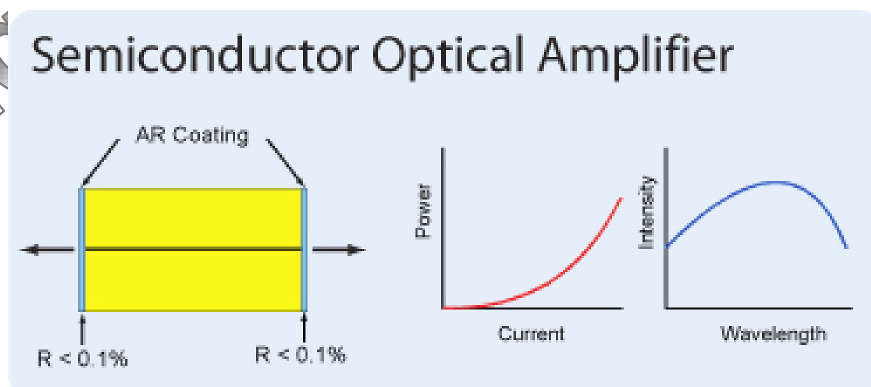
Fiber Optic Components

- **The components which we'll examine are:**
  - LEDs (Light Emitting Diodes)
  - LASERs (I'll write this as "laser" from here on).
  - Photodetectors
  - **Semiconductor optical amplifies**
  - Erbium doped fiber amplifier
  - Raman amplification
  - DWDM mux and demux

Semiconductor Optical Amplifier (SOA)

- **The semiconductor amplifier is basically a semiconductor laser without the reflective ends.**
- **Light enters the gain medium and causes stimulated emission.**
- **The amplified light has the same frequency, phase and direction as the incoming signal.**





Semiconductor Optical Amplifier (SOA)

- Can be made to operate from about 1300nm to 1700nm by changing the composition of the InGaAsP.
- Gain of about 20 dB.
- Gain bandwidth usually less than 85nm.
- Higher noise than erbium doped fiber amplifiers.
- But much more compact than an EDFA.
- Often used just before a photodetector as a preamplifier.

Fiber Optic Components

- **The components which we'll examine are:**
  - LEDs (Light Emitting Diodes)
  - LASERs (I'll write this as "laser" from here on).
  - Photodetectors
  - Semiconductor optical amplifier
  - **Erbium doped fiber amplifier**
  - Raman amplification
  - DWDM mux and demux

Erbium Doped Fiber Amplifier (EDFA)

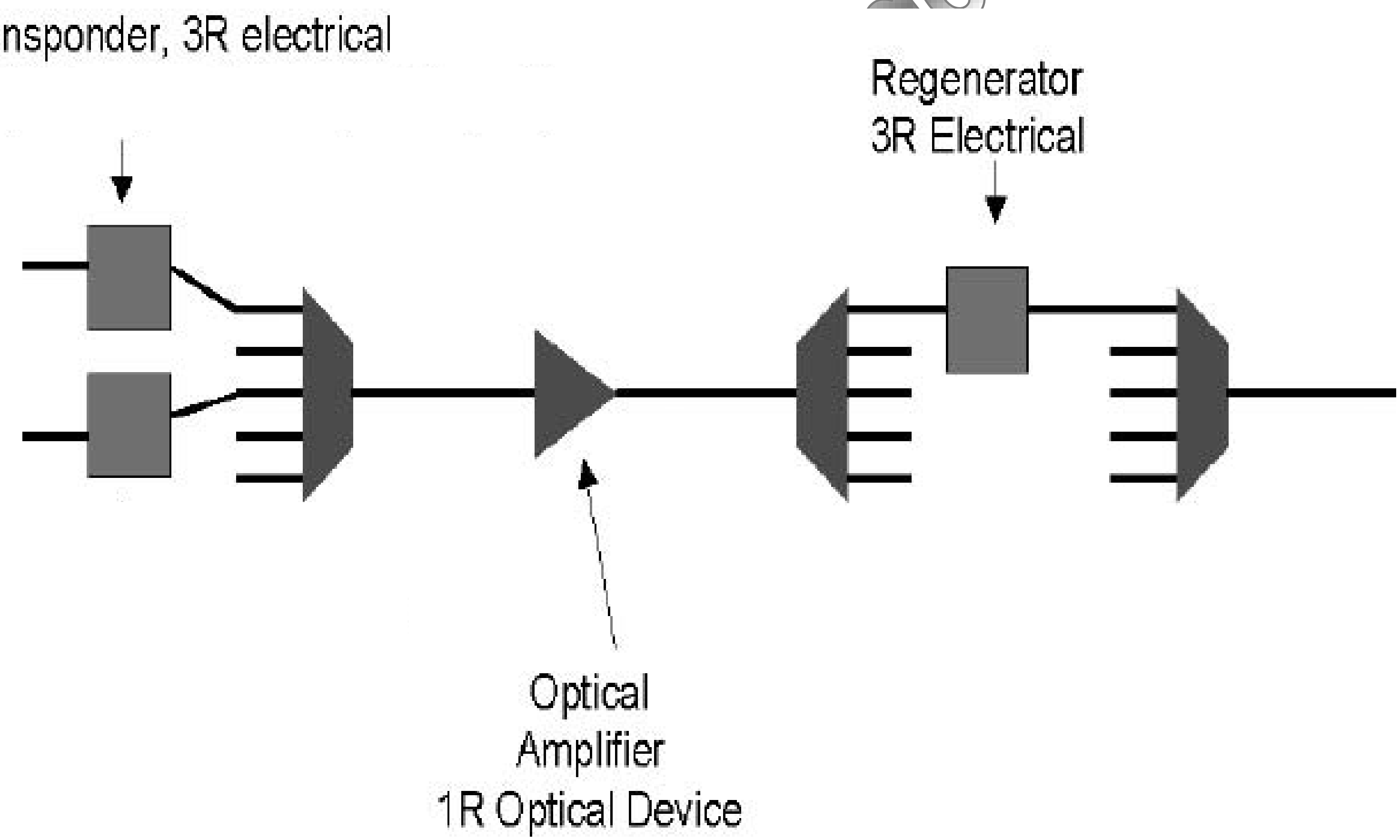
- **Optical fiber has attenuation, which means after the signal has traveled far enough, it will not be able to be recovered.**
- **To achieve long distances, the signal must be “restored”.**
- **One way to “restore” the signal is to convert it back to electronic and then retransmit it.**
- **This is called 3R regeneration:**
  – Re-amplification
  – Re-shaping
  – Re-timing
- **Also called OEO, or Optical, Electronic, Optical.**

Erbium Doped Fiber Amplifier (EDFA)

- **But OEO is expensive.**
  - Lots of expensive equipment, especially for DWDM.
  - Significant amount of power.
- **Better if we could amplify in the optical domian.**
- **And that's where the Erbium Doped Fiber Amplifier (EDFA) comes in.**
- **An EDFA consists of a length of fiber doped with the rare earth Erbium.**

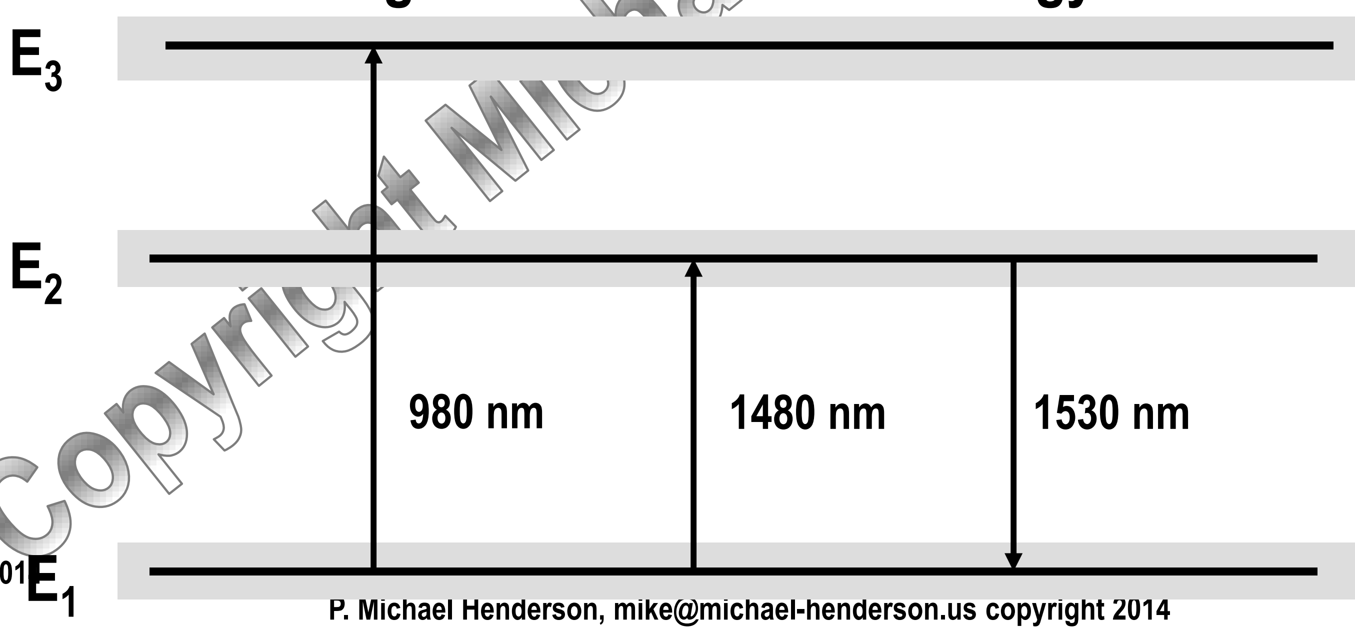
Regeneration

- **Example of OEO regeneration and optical regeneration.**
- **One OEO unit needed for every lambda, plus mux and demux.**



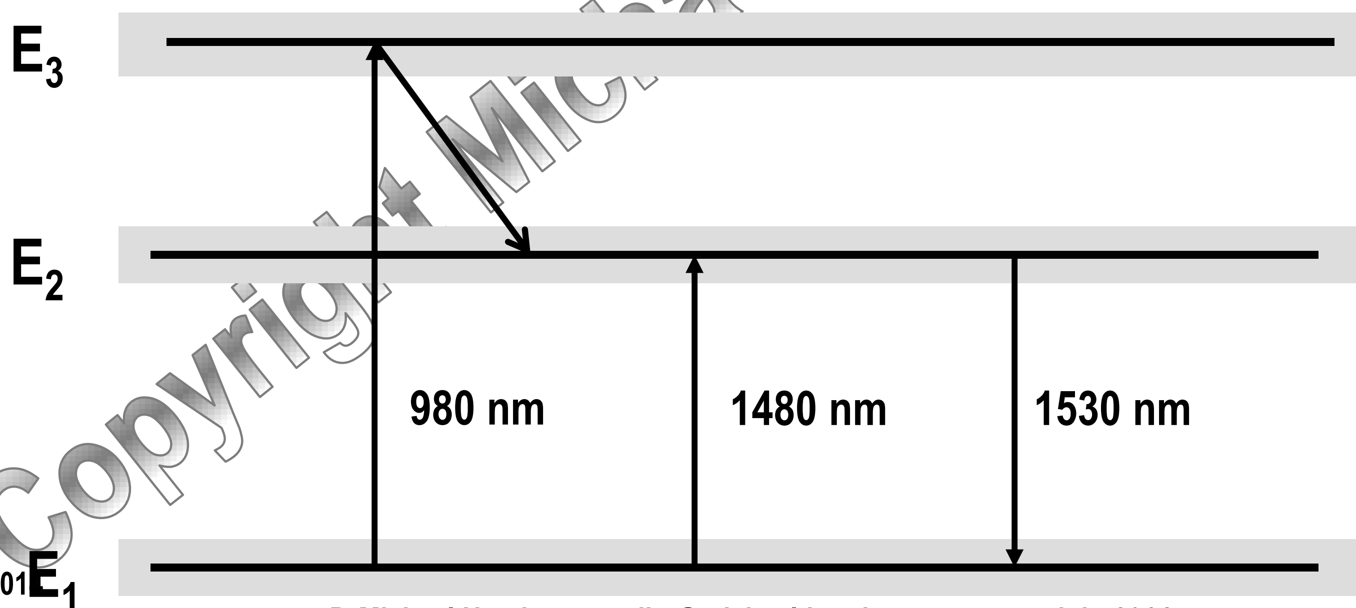
EDFA

- **Erbium has multiple energy levels, three of which are of interest to us, labeled E_1, E_2, and E_3.**
- **Since the Erbium ions are part of the glass fiber, the energy bands are split into multiple energy levels, a process known as *Stark splitting*. The net effect is that at the macroscopic level the Erbium ions appear to have a continuous energy band in the region of each discrete energy band**



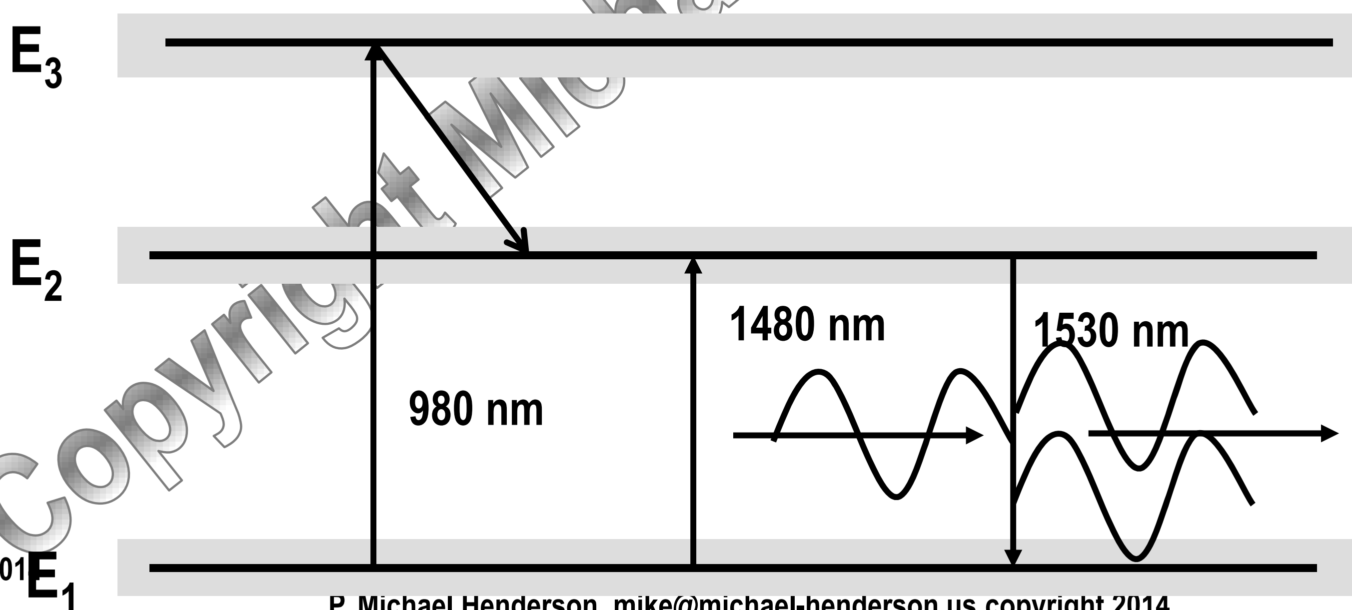
EDFA

- When pumped by a 980 nm laser, electrons that are normally at the rest state E_1 are pumped to state E_3.
- The electron in the E3 state has a short lifespan, about 1 μs, before decaying to the E2 state where it has a lifespan of about 10ms.
- Given sufficient pumping energy, a population inversion will result.



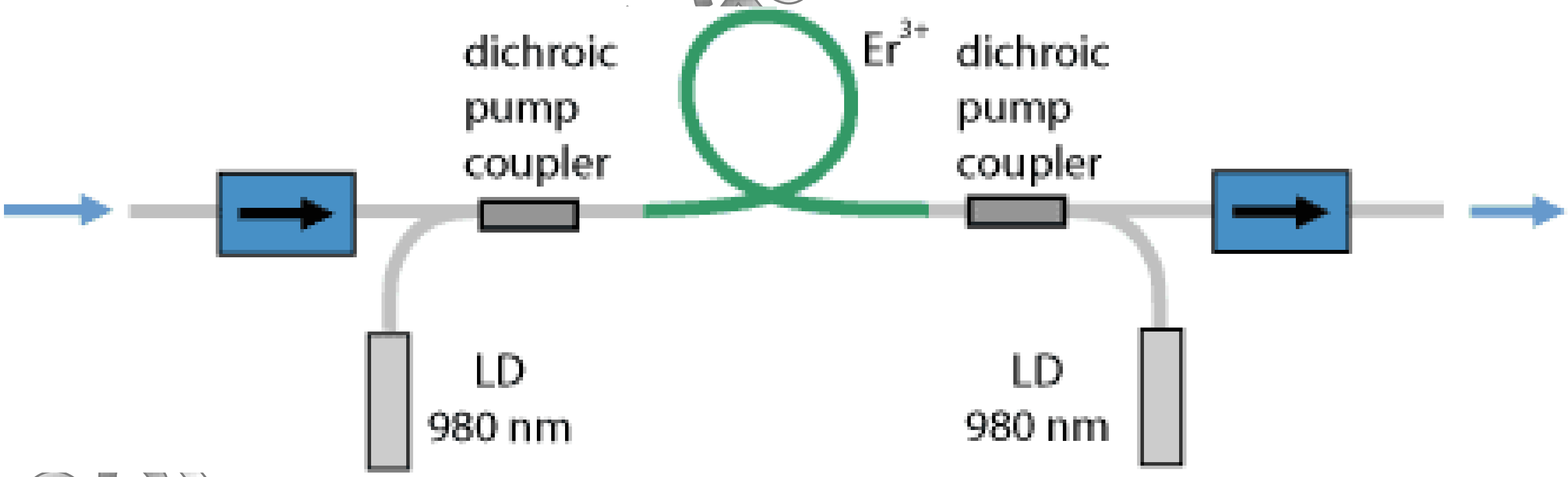
EDFA

- When a photon enters the fiber, it will cause stimulated emission in the 1530nm band.
- The EDFA functions similar to an SOA, except that the doped fiber is much longer than the cavity of the SOA, providing for more stimulate emission photons.
- The EDFA generally provides greater gain than an SOA.



EDFA

- Erbium based amplifiers, in silica, operate best from 1530 to 1560nm in the C band, providing 30dB of gain.
- EDFAs are available which are optimized for other "bands" than 1530 to 1560, but only in the C and L bands.
- There are issues, such as gain flattening, which are not discussed here.



Fiber Optic Components

- **The components which we'll examine are:**
  - LEDs (Light Emitting Diodes)
  - LASERs (I'll write this as "laser" from here on).
  - Photodetectors
  - Semiconductor optical amplifiers
  - Erbium doped fiber amplifiers
  - **Raman amplification**
  - DWDM mux and demux

Raman Amplification

- A higher frequency (shorter wavelength) signal will transfer energy to a lower frequency (longer wavelength) signal.
- To provide amplification, a shorter wavelength signal is launched into the fiber close (10 to 15 THz) to the signal to be amplified.
- It's usually launched counter to the propagation of the information carrying signal.
- Often used in conjunction with an EDFA to provide amplification on long haul links.
- Can be used in any optical band.

Fiber Optic Components

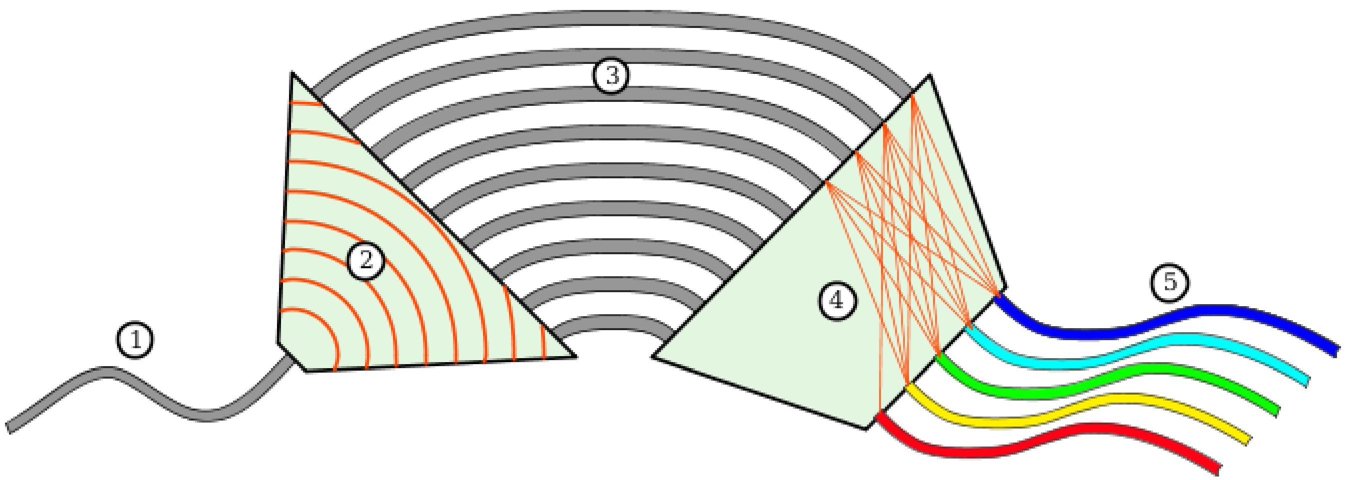
- **The components which we'll examine are:**
  - LEDs (Light Emitting Diodes)
  - LASERs (I'll write this as "laser" from here on).
  - Photodetectors
  - Semiconductor optical amplifiers
  - Erbium doped fiber amplifier
  - Raman amplification
  - **DWDM mux and demux**

DWDM Mux and Demux

- **How do we get all those lambdas into a single mode cable, and then get them out again?**
- **That's the job of the Mux and Demux.**
- **There are several technologies used today. A few are:**
  - Arrayed Waveguide Grating
  - Prism Diffraction Multiplexing
  - Waveguide Grating Diffraction
  - Multi-Layer Interference Filters

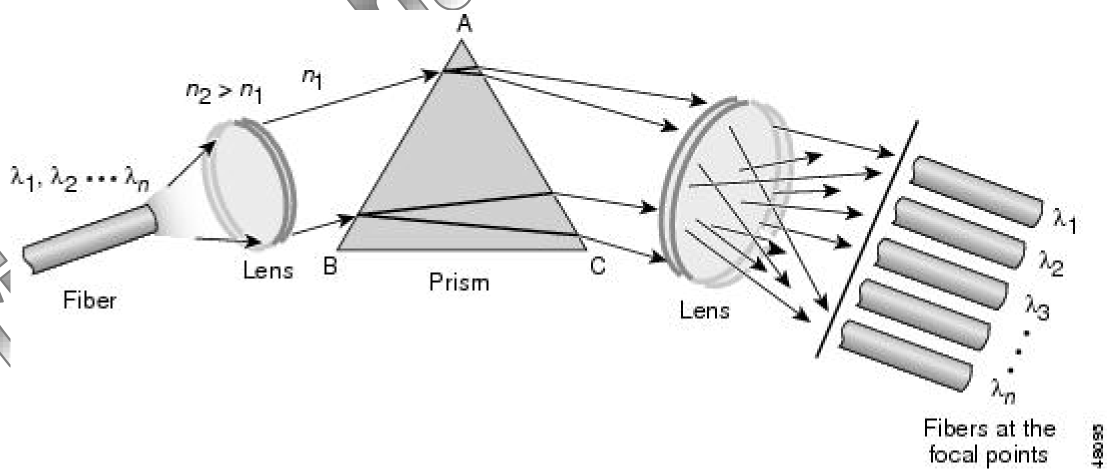
Arrayed Waveguide Grating

- The incoming light **(1)** traverses a free space **(2)** and enters a bundle of optical fibers or channel waveguides **(3)**.
- The fibers have different length and thus apply a different phase shift at the exit of the fibers.
- The light traverses another free space**(4)** and interferes at the entries of the output waveguides **(5)** in such a way that each output channel receives only light of a certain wavelength.
- Can be used for demuxing as well as muxing.



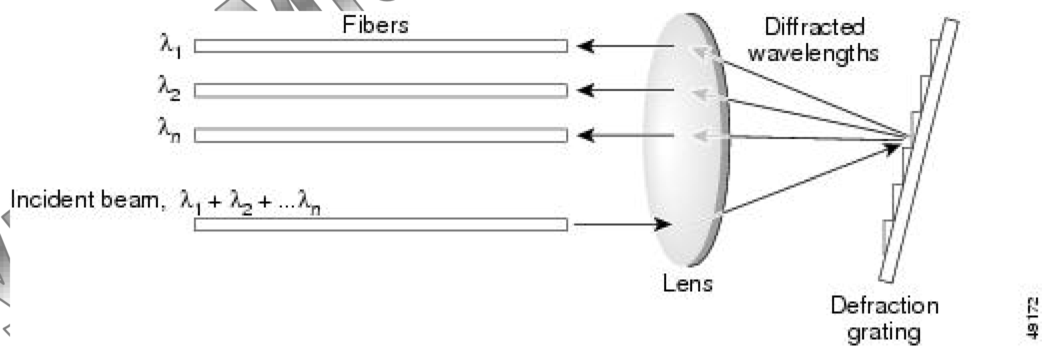
Prism Diffraction Multiplexing

- The light from the fiber is focused to parallel and then put through a prism.
- The prism separates the different lambdas and the second lens focuses the light into an array of fibers.



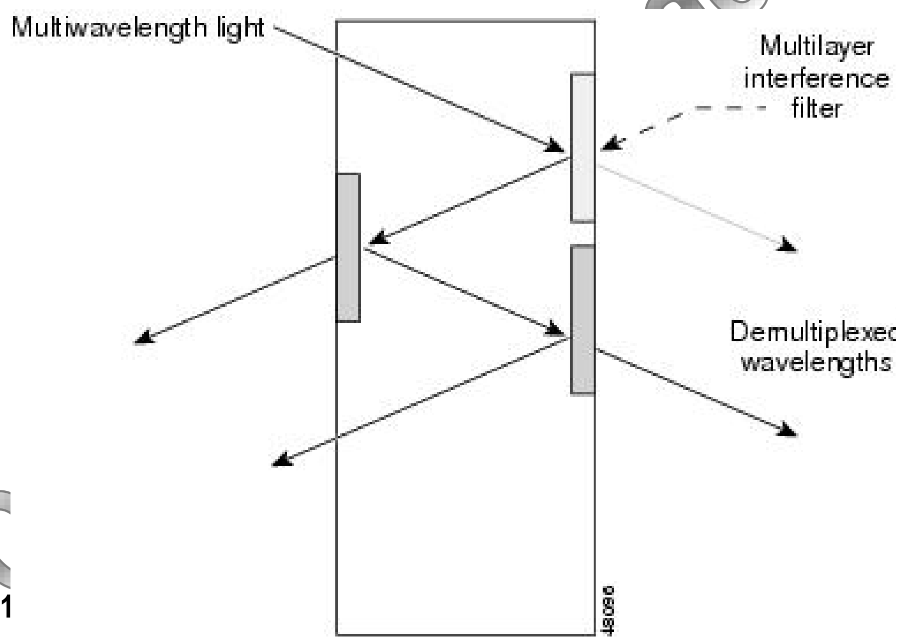
Waveguide Grating Diffraction

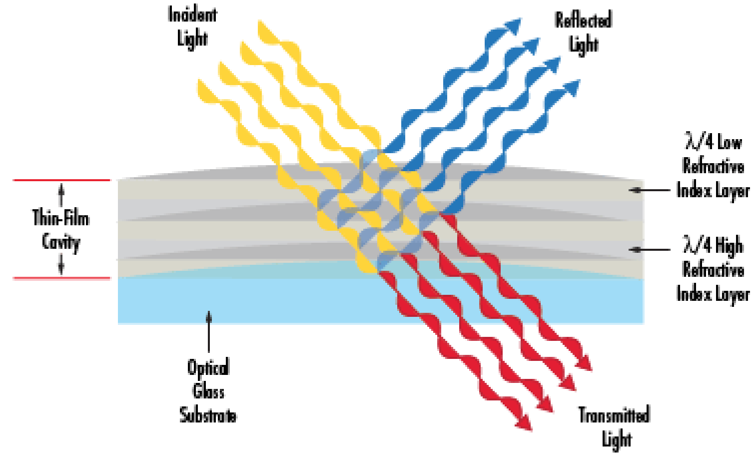
- The light enters through a lens which focuses the light on a diffraction grating.
- The light is reflected and separated by the diffraction grating.
- The lens focuses the light into an array of fibers.



Multi-Layer Interference Filters

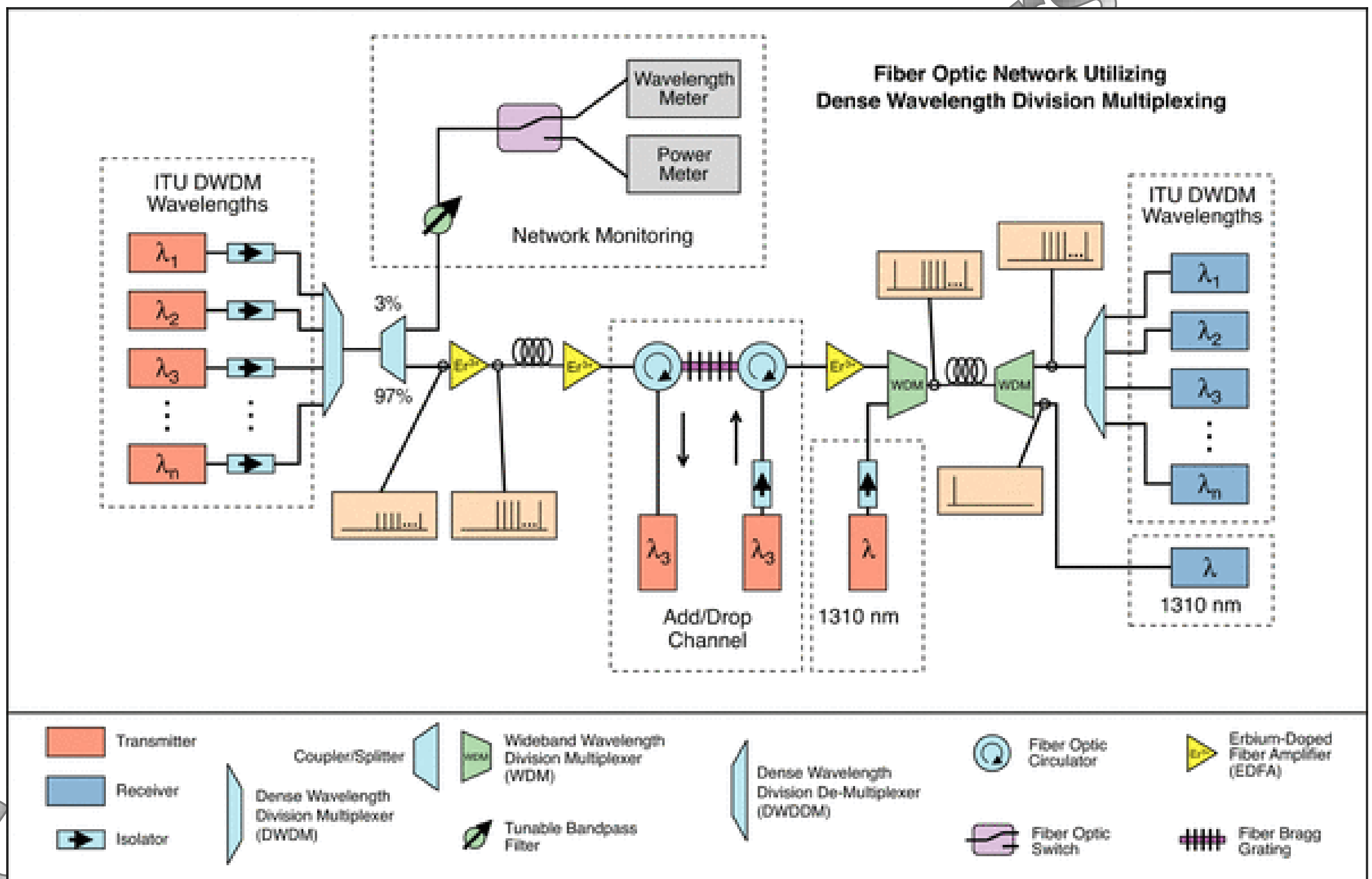
- Dichroic filters can separate certain wavelengths and reflect (or transmit) the rest.
- A sequence of dichroic filters can separate the lambdas in a DWDM fiber.





Putting it all together

- **This diagram shows a DWDM system, with multiplexers, optical amplifiers, an EDFA, and an optical add/drop unit.**



Network Communications

- **Note that we're still in the voice section of this presentation – we'll get to data soon.**
- **Now, we're going to examine how voice is carried over the technologies we've been discussing so far.**
- **There are two major "technologies"**
  - The Plesiochronous Digital Hierarchy (PDH)
  - SONET/SDH
- **After we complete these two subjects, we'll take a look at voice switching.**

PDH and SONET/SDH

Agenda

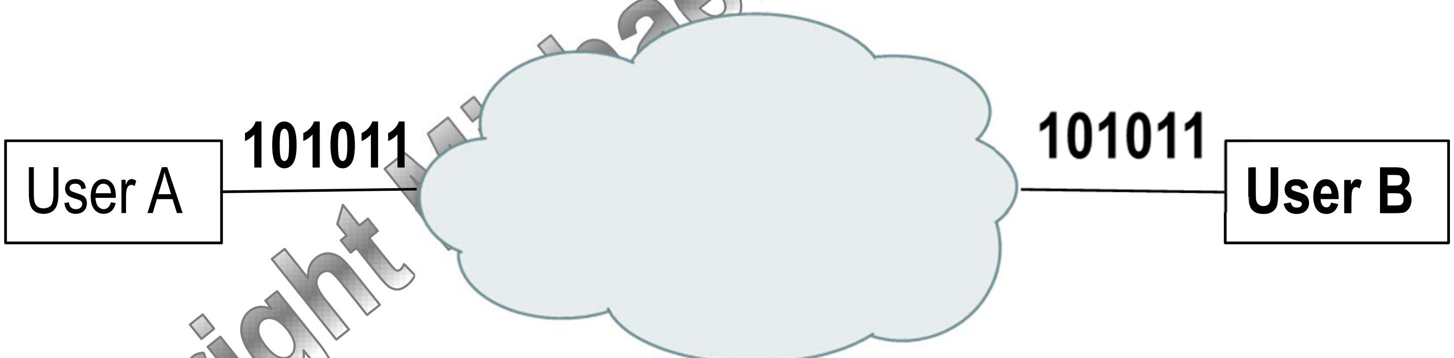
- **Clocks in digital communications**
- **Review of Plesiochronous Digital Hierarchy (PDH)**
  - DS1/DS2/DS3
  - E1/E2/E3
  - Multiplexing of PDH
- **Why SONET/SDH**
- **Historical background**
- **Basic structure of the SONET/SDH frame**
- **Transport overhead**

Agenda (continued)

- **Payload pointer processing**
- **Payload overhead**
- **Mapping of SONET payloads**
  - Virtual tributaries
  - Handling ATM, POS, and GFP
- **Automatic protection switching**
- **Summary**

Basic Digital Communications Concepts

- **For purposes of this presentation, we'll define digital communications as the transfer of information in binary form between two points.**



Clock

- To accurately receive the data, the receiver must know when to look for a bit. This requires the concept of clocking.
- Example: If the sender is sending 10 bits per second, the receiver must know to look for a bit every 1/10 second.
- And the receiver must know when – within that 1/10 second – to sample the incoming data to determine whether it's a 1 or a 0.
- To accomplish this, we use clock signals.
- Most of the complexity in network protocols comes from the need to accommodate clocks at different rates.

What is a "Clock?"

- In electronics, a clock is a signal, usually a square wave, with a frequency equal to the transmit bit rate.
- The incoming data stream is sampled on either the rising or falling clock edge to determine whether the information is a 1 or a 0.

Clock

- **How can the receiver generate a clock that "matches" the transmitter clock?**
- **Normally, the receiver generates a clock from the received data.**
  - The receiver will look at the transitions in the incoming data and generate clock based on how often the transitions occur.
  - A phased lock loop (PLL) is generally used.
- **This means that the receive clock should wind up with the same frequency as the transmit clock, but with a different phase because of the propagation delay.**

Clock Problems

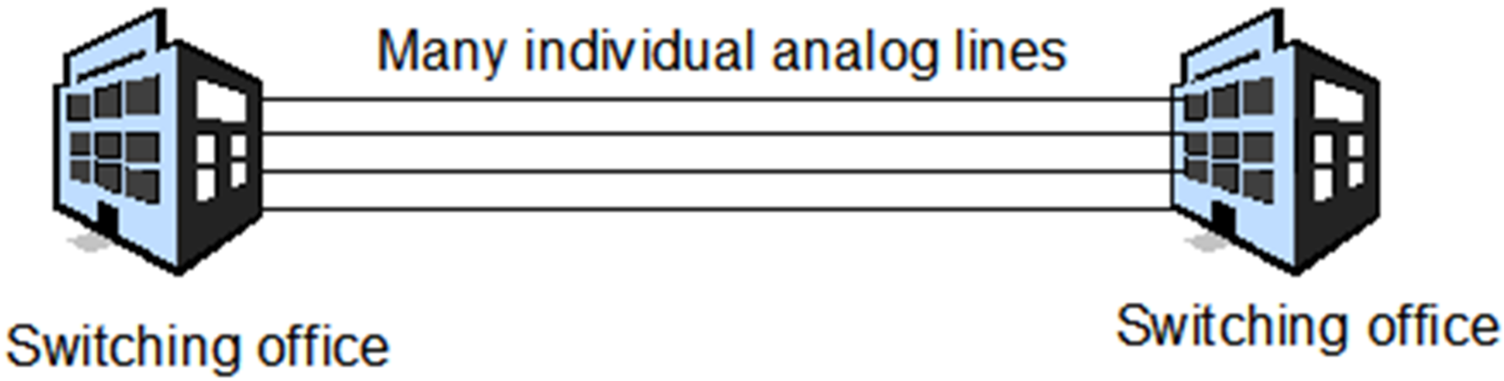
- **Clock instability**
  - Wander – the transmit clock's frequency varies over time. The ITU defines wander as changes that occur less than 10 times per second (<10Hz).
  - Jitter – The clock's frequency changes over time at a rate that exceeds 10 times per second (>10Hz).
- **Causes**
  - Noise and interference in the transmission medium can "trick" the PLL.
  - For mobile systems, Doppller shift and changes in distance.
  - Plain old drift in the transmit clock.

Plesiochronous Digital Hierarchy (PDH)

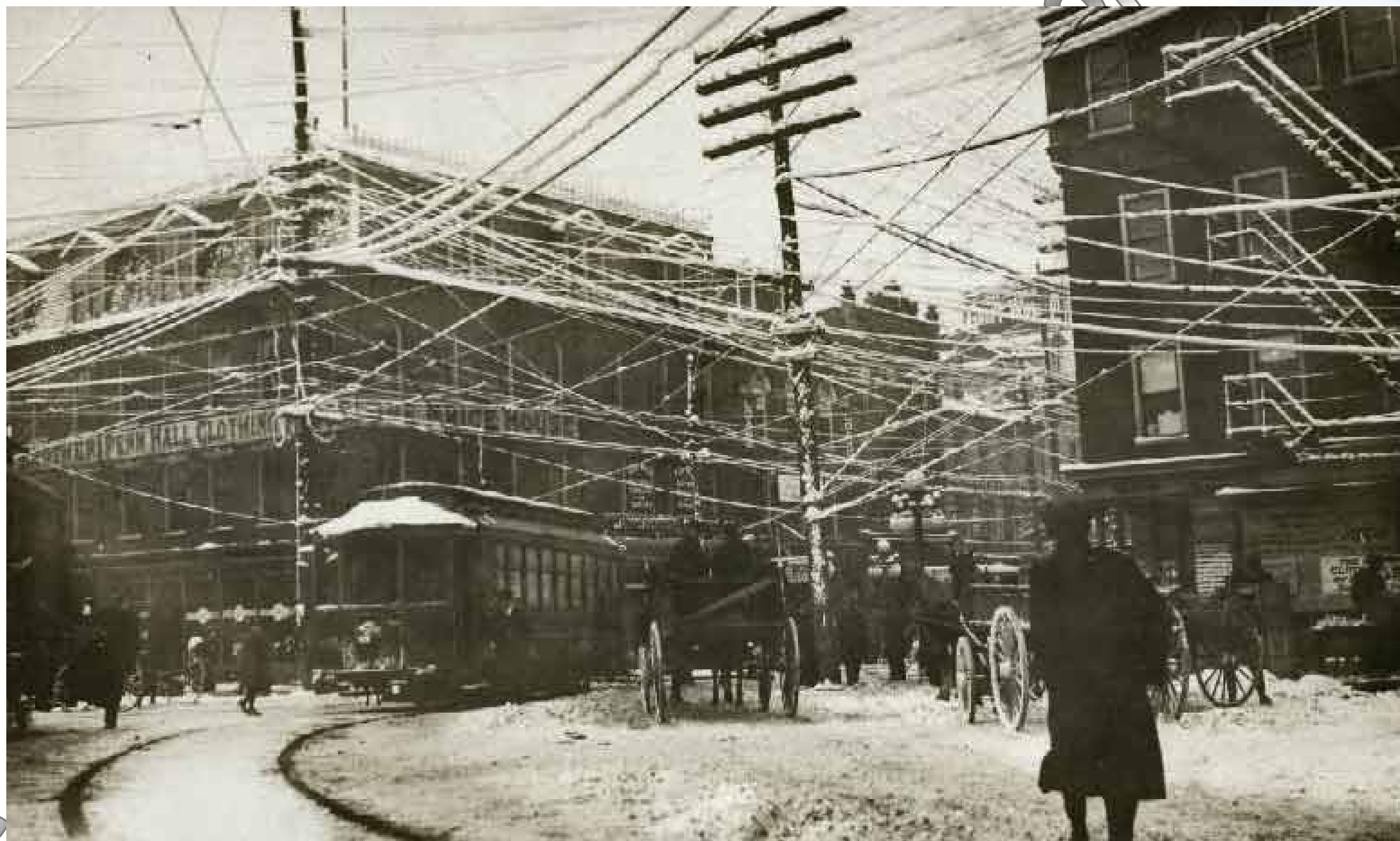
- **Plesiochronous means that the devices in the network have transmit clocks that are close to each other in frequency but are not exactly synchronized.**
- **The services associated with PDH are**
  - DS1 and DS3 in the United States
  - E1, E2 and E3 in most of the rest of the world
- **Question: Why 1.54Mbps in the US and 2.048 in Europe?**

Why PDH

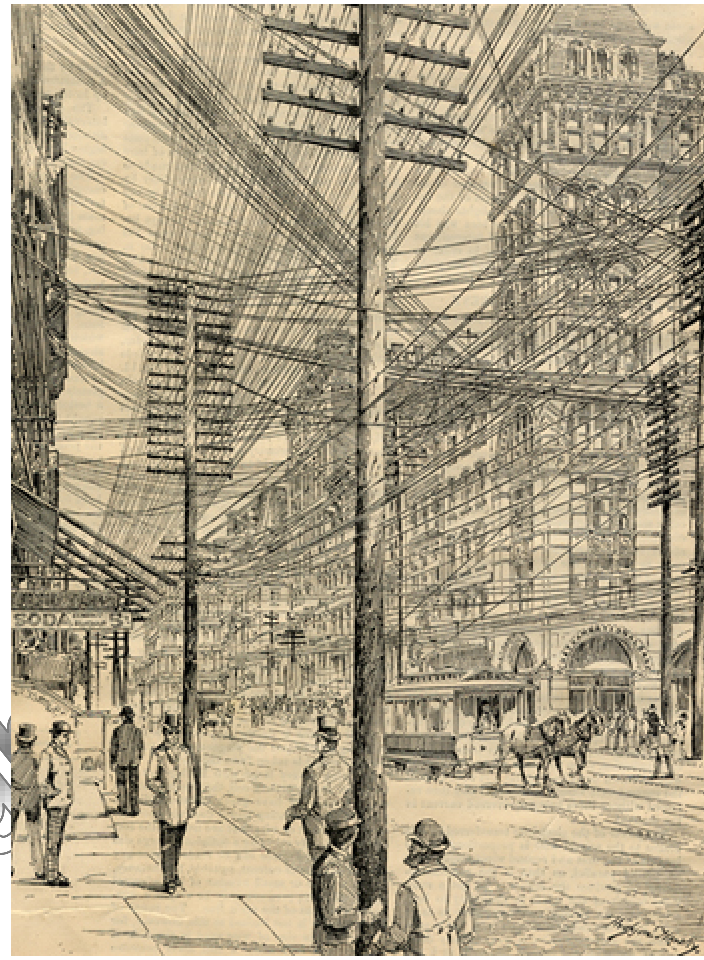
- The PDH developed to support voice communications in the telephone network.
- In the "old days" everything was analog and central offices were connected with analog lines.



Telephone lines in New York City

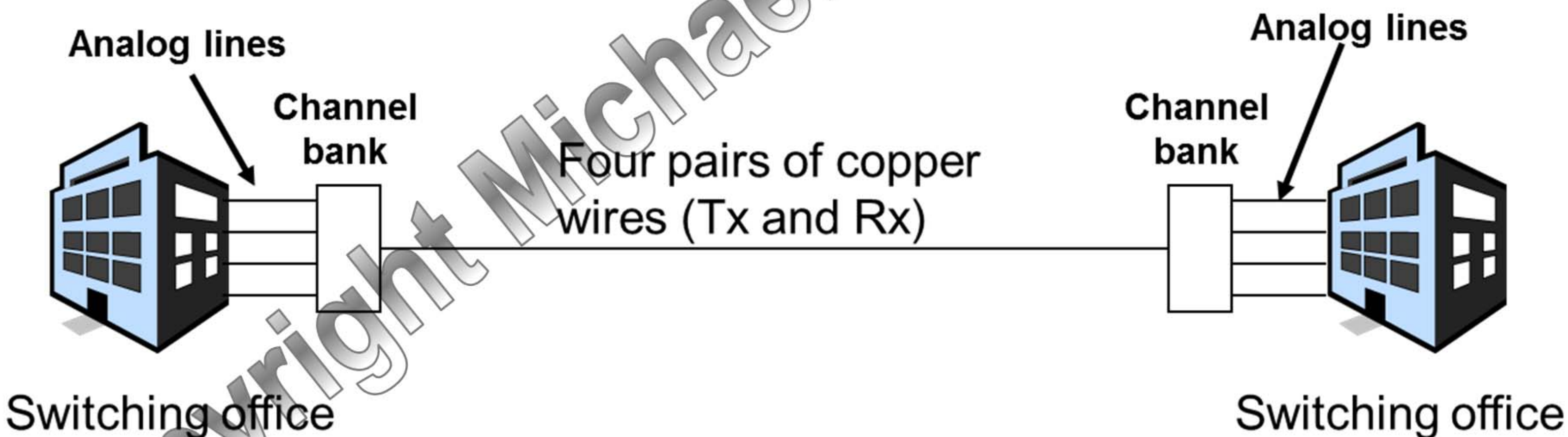


Another view of NYC – about 1900



Why PDH

- **But soon, there were too many analog lines. There had to be a way to concentrate multiple calls into a single communications line.**
- **Enter the channel bank.**



Channel Bank (Analog Switches)

- The channel bank took 24 analog lines, converted the signal on each line to digital at 64Kbps (each voice sample is 8 bits, sampled 8,000 per second).
- Those 24 64Kbps channels were communicated across a data link to the channel bank on the other end of the link, where each channel was converted back to analog.
- This specific link was called a T1 link, and it operated at 1.544Mbps (nominal).
- The 1.544 Mpbs signal could be fast or slow but since the voice channels were changed back to analog on the other end, it didn't matter.
- Thus, nothing was needed to accommodate clock differences.

But Problems Came Up

- **Analog switches were replaced by digital switches.**
  - 64Kbps data had to flow through the switch and out to another digital link.
  - But 64Kbps was only nominal. Actual channels could be slightly more or less than 64Kbps because of differences in the sampling rate.
- **As the network grew, more and more T1 lines were needed between switching centers.**
  - The phone company needed to multiplex many T1 lines into a single line.
  - But that was a problem because the clocks in the T1s were different.

Plesiochronous Digital Hierarchy

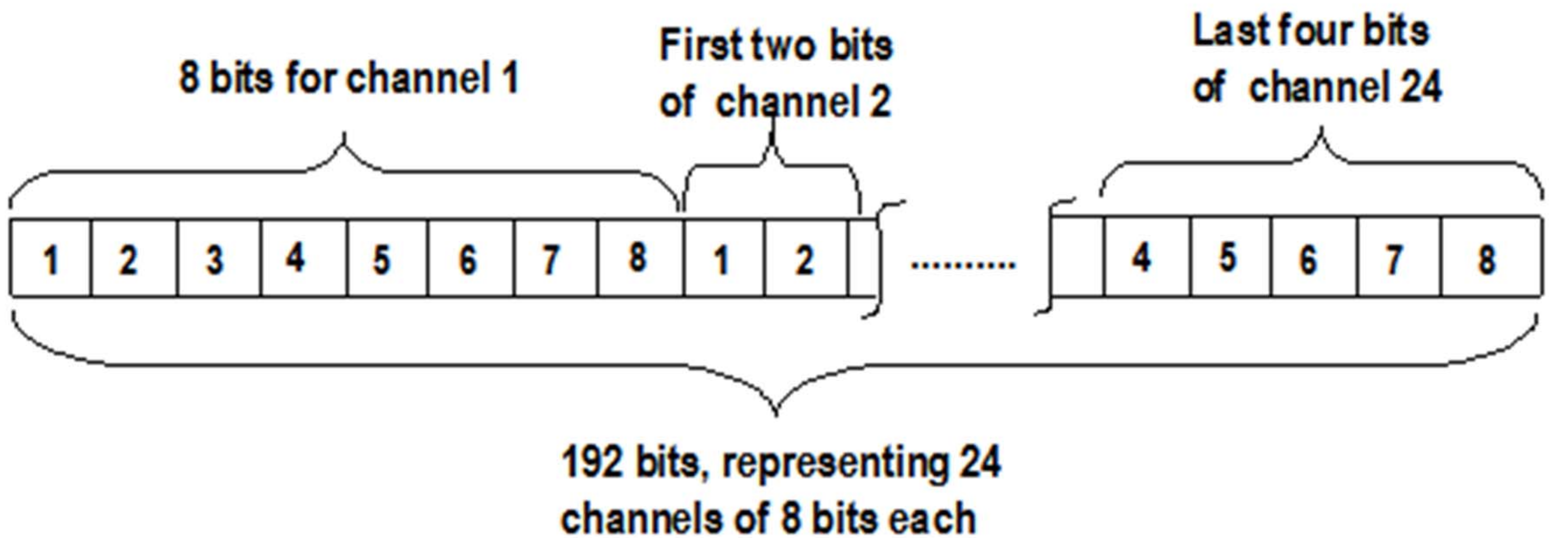
- **PDH is all about how to deal with these different clocks when transmitting data.**
- **Let's look at how this is done, starting with the US system.**
- **The basic building block of the PDH is the voice channel. Voice is sampled 8,000 times per second and each sample is 8 bits, making each voice channel a 64 Kbps bit stream.**

DS1

- The DS1/T1 technology was developed at Bell Labs and introduced in 1962 to carry voice channels.
- The 1.544Mbps signal can be referenced as either a DS1 or a T1. What's the difference?
- Normally, when working with the digital signal prior to putting it on the transmission line, it's referred to as Digital Signal 1 or DS1.
- To transmit a DS1, a line code must be used (Alternate Mark Inversion or AMI). Once the signal is in the AMI domain, it is referred to as T1.
- But in the US, a 1.544Mbps line is usually referred to as a T1 line.

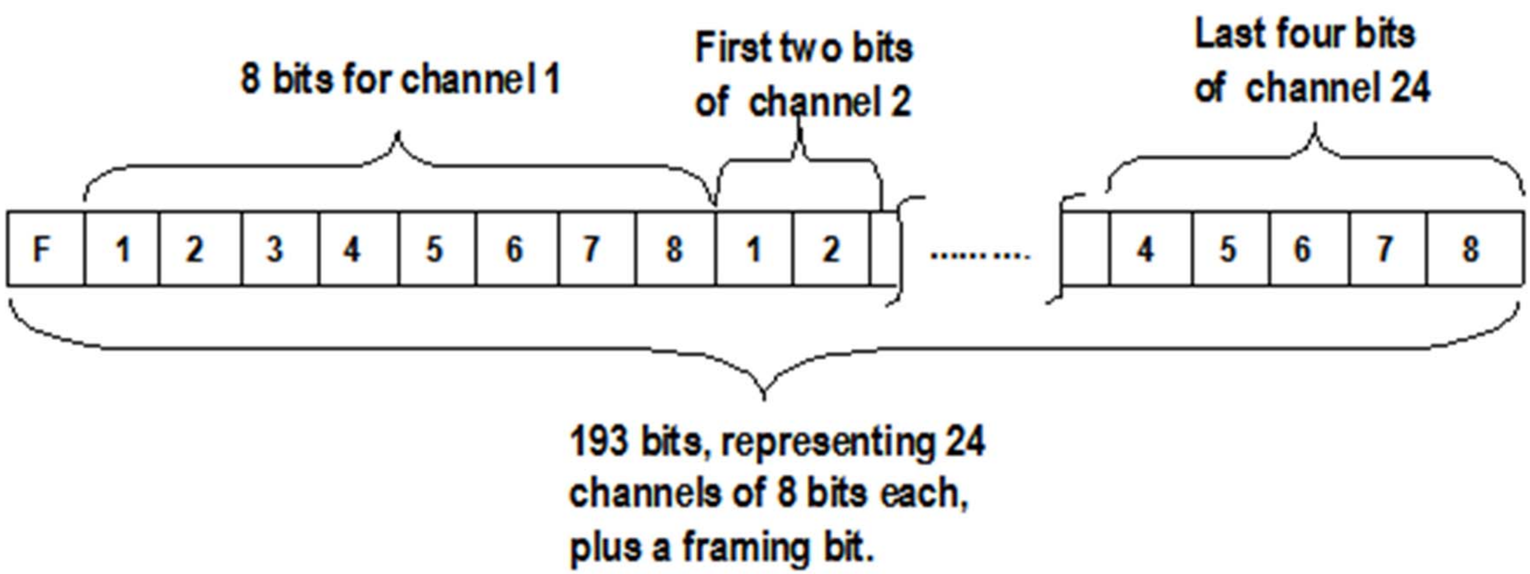
DS1

- The basic structure of the DS1 signal is shown in the picture.
- 8 bits from each of 24 voice channels are put into the bit stream. All 8 bits from each channel are together. The channels are octet multiplexed, not bit multiplexed.



DS1

- **But we can't just send that stream of bits. We have to have some way to find the start of the block so that we can recover the octets of the voice channels.**
- **This is done by the addition of a Framing bit at the front of the frame.**



DS1 Clocks

- 193 bits, 8,000 times per second gives us the DS1 rate of 1.544 Mbps.
- Note that there's no accommodation in the DS1 frame for voice channel clocks that may be fast or slow (a bit more than 8,000 samples per second, or a bit less).
- But we know that clocks that are not synchronized cannot be perfect. So how do we accommodate those voice channels that may be fast or slow when we build the DS1 frame?
- E1 is easier because the rate is simply 32 times 64KHz, so the voice channels and E1 can operate off the same base clock. (use 2048MHz clock and divide down by 256 to 8KHz)

DS1 Framing

- **So how can one bit be used for framing?**
- **Remember that DS1 were developed to carry voice circuits.**
- **Prior to digitization, the voice channel is bandpass filtered rolling off at 3400Hz, with a required attenuation of 14dB at 4KHz.**
- **The framing is an alternating {1,0} sequence. That is, if one frame has a 1 bit in the framing position, the next frame will have a 0 in that position.**
- **Since 4KHz is essentially eliminated as a signal in the channel, it's almost impossible for a voice channel to have a repeating {1,0}.**

Obtaining Frame Sync

- The brute force method is to select a bit position and look ahead 193 bits to see if the bit value changed.
- If not, select the next bit position and look ahead 193 bits, etc.
- Eventually, you'll find an alternating {1,0}.
- The average time to obtain frame sync is

$$\text{Frame time} = N^2 + \frac{N}{2} \quad \text{Bit times}$$

Where N is the number of bits in the frame, including the framing bit. For DS1, it's about 24 ms.

Why just one bit for framing?

- **A reasonable person might ask is, "Why just one bit for framing?"**
- **More bits would make finding frame sync much faster.**
- **I've read several opinions about this but the most likely answer is that bandwidth was precious back in the days when DS1 was developed.**

Call Progress Indications

- **There's something else that the DS1 has to carry besides the voice channels – it also has to carry call progress indications so that a call can be set up and torn down.**
- **This was done with a technique known as "robbed bits".**
  - The low order bit of certain channels were taken for call progress signaling.
  - These bits are known as the "A/B" bits and allow for 4 messages {0,0}, {0,1}, {1,0}, and {1,1}.
- **To implement these call progress bits, the designers had to introduce the concept of the "Superframe" which will be described next.**

DS1 Superframe

- **A superframe consists of 12 of the 193 bit frames.**
- **The {1,0} framing was moved to alternated frames so the average sync time was doubled to about 48 ms.**
  - The first bit of the first frame has a framing character.
  - The next framing character is in the third frame.
- **This allows us to get block sync, but how do we get Superframe sync?**
  - The other six frames have framing bits of {0,0,1,1,1,0}
- **This allows us to get both frame sync and Superframe sync.**

DS1 Superframe

Frame number	Bit number	Frame alignment bit value	Superframe alignment bit value	Signaling bit value in low order data bits
1	0	1	-	
2	193	-	0	
3	386	0	-	
4	579	-	0	
5	772	1	-	
6	965	-	1	A
7	1158	0	-	
8	1351	-	1	
9	1544	1	-	
10	1737	-	1	
11	1930	0	-	
12	2123	-	0	B

DS1 Superframe – A/B bits

- **Note that the A/B bits are in the 6^{th} and 12^{th} frame of the Superframe.**
- **The bits in the 6^{th} frame are the “A” bits and the bits in the 12^{th} frame are the “B” bits. Their meaning is defined in ANSI T1.403.02-1999.**
- **Every voice sample in those blocks has the low order bit “robbed”. What impact does this have on the voice?**
  - 50% of the time, the value of the low order bit will be the same as the sampled value.
  - The other 50% of the time, the output level moves to the next quantization level (up or down) but the overall comprehension is not seriously affected.

DS1 Extended Superframe

- **The Superframe concept worked well for many years.**
- **But when ATT started provisioning T1 to customers, there was the need for a better way to monitor the line.**
- **Enter the Extended Superframe.**
  - Consists of 24 frames so we have 24 framing bits
  - Six are allocated to the “Frame alignment signal (FAS)” {0,0,1,0,1,1}.
  - Six are allocated to a CRC.
  - Twelve are allocate to a data link channel.

DS1 Extended Superframe

Frame number	Bit number	Frame Alignment Signal bit value	Data Channel bits	CRC bits	Signaling bit value in low order data bits
1	0	-	M1		
2	193	-		C1	
3	386	-	M2		
4	579	0			
5	772	-	M3		
6	965	-		C2	A
7	1158	-	M4		
8	1351	0			
9	1544	-	M5		
10	1737	-		C3	
11	1930	-	M6		
12	2123	1			B
13	2316	-	M7		
14	2509	-		C4	
15	2702	-	M8		
16	2895	0			
17	3088	-	M9		
18	3281	-		C5	C
19	3474	-	M10		
20	3667	1			
21	3860	-	M11		
22	4053	-		C6	
23	4246	-	M12		
24	4439	1			D

DS1 Extended Superframe

- **Note we now have four signaling bits, {A,B,C,D}, giving us 16 signaling states.**
- **Frame alignment is found from the frame alignment signal and will now take an average of 96ms.**
- **Extended superframe alignment can be checked with the CRC bits. So first find alignment with the FAS, then look to see that the CRC checks. If so, you have Extended Superframe sync.**
  - The CRC is the check of the previous block and assumes the FAS bits are all 1.

DS2

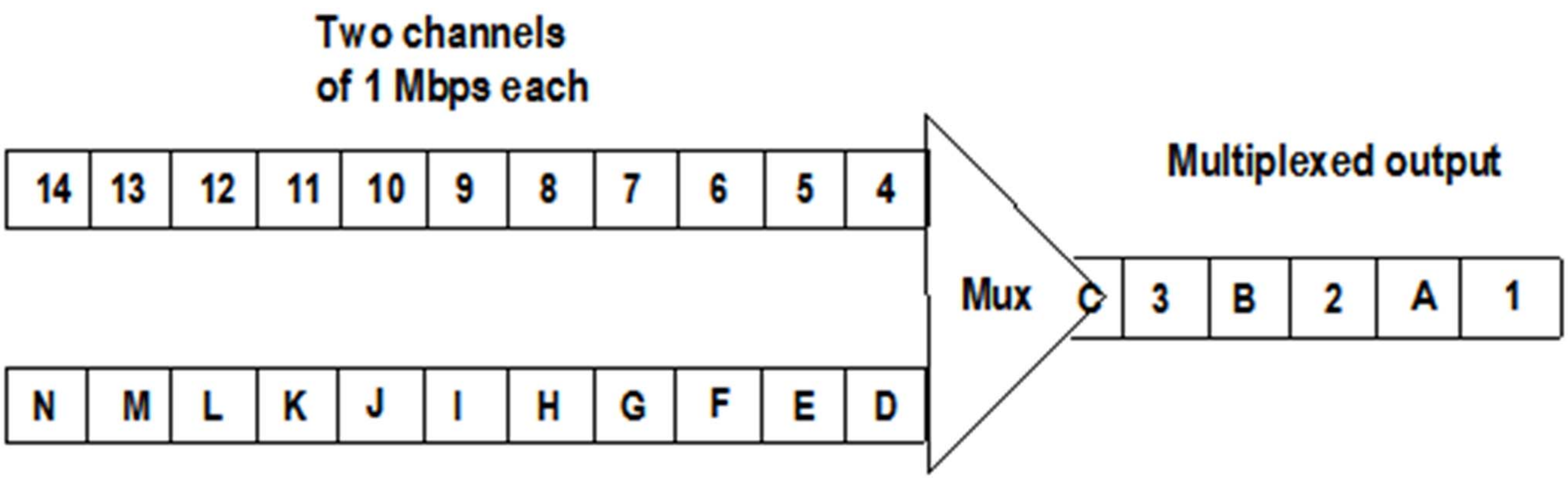
- DS2 is no longer used in the network. The only place it's used in inside an M13 multiplexer.
- The problem with multiplexing DS1 lines is that each line runs at a slightly different rate. And clock variations can occur during use.
- Let's discuss how we can deal with multiplexing data streams that have slightly different data rates.

Multiplexing

- Suppose you wanted to multiplex two nominal 1Mbps data streams into a single data stream.
- We have to multiplex them into a data stream that's slightly more than 2Mbps so that if either input is slightly more than 1Mbps, we'll be able to accommodate them.
- We'll also need some framing bits in the output so that we can extract the data streams correctly.

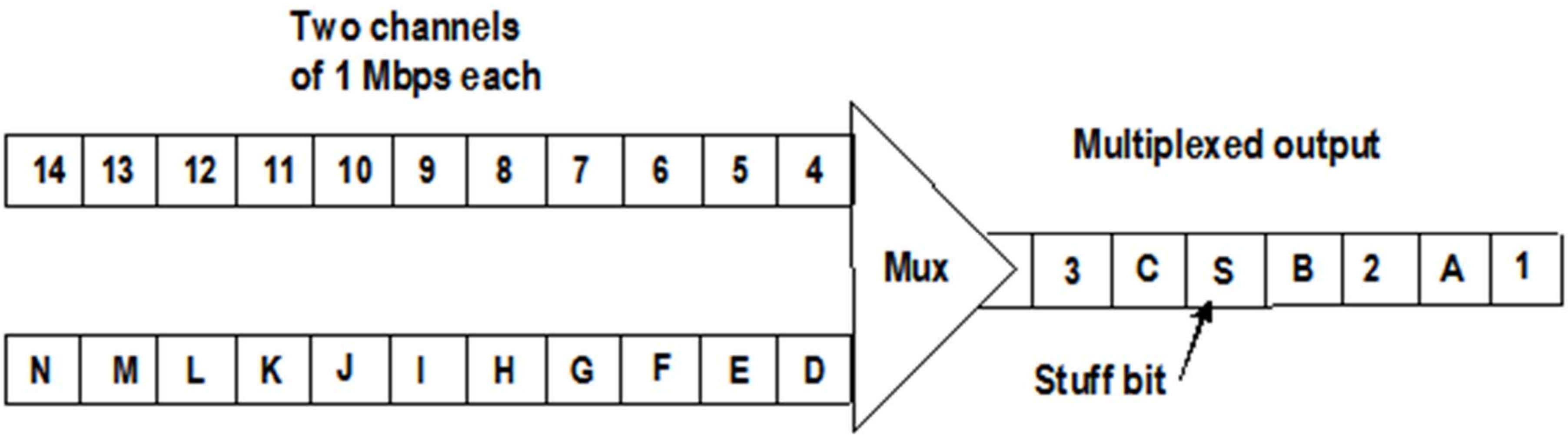
Multiplexing

- Here I'm ignoring any framing bits needed to extract the data and only showing how the bits are multiplexed bit interleaved.



Multiplexing

- Let's look at what happens if the top stream is a bit less than 1Mbps.
- In this case, we'll come to a situation were we need to output a bit but none is available from the first line.
- To deal with that, we insert a "stuff" bit in the output stream.
- The problem is how to let the other side know it's a stuff bit.

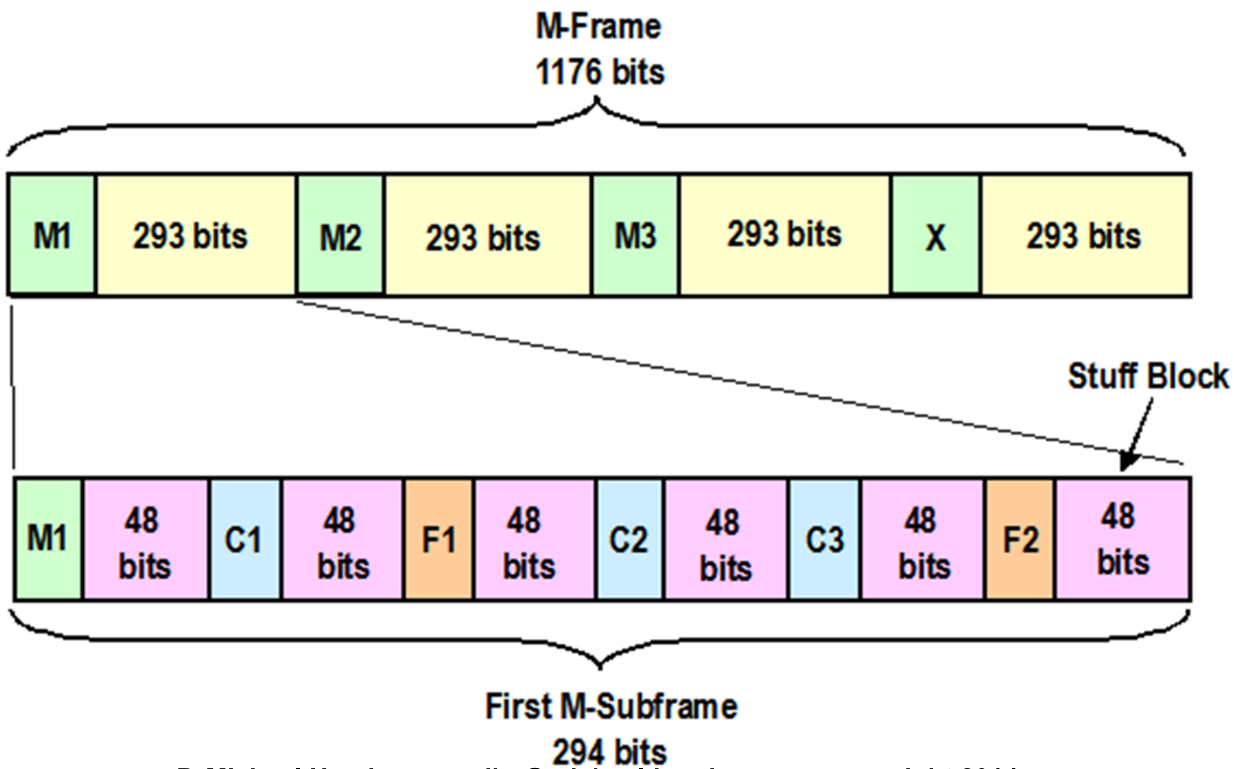


DS2 Framing and Multiplexing

- **Four DS1 lines are multiplexed into a DS2.**
- **A DS2 has a nominal bit rate of 6.312Mbps**
- **Four DS1 lines at the nominal rate of 1.544Mbps equal 6.176Mbps so the DS2 line has excess bits to handle clock variation and framing.**
- **Let's look at the DS2 framing.**

DS2 Framing

- The basic frame sync is found from the F1, F2 bits which are alternating {0,1}.
- The M-frame sync is found from the M1, M2, M3 bits which are {0,1,1}.



DS2 Multiplexing

- The bits from the DS1s are bit interleaved into the 48 bit blocks of the DS2 frame.
- Each 48 bit block carries 12 bits for each DS1, except the last 48 bit block.
- The first bit in the last 48 bit block in the first subframe is the stuff bit for the first DS1.
- Stuffing is controlled by the “C” bits. A 1 indicates the bit is stuff, while a 0 indicates the bit is data. Majority voting applies.
- Thus, the first DS2 subframe carries either 71 or 72 bits for the first DS1, and 72 bits for the other DS1s.

DS2 Multiplexing

- The stuff bit for the second DS1 is the second bit in the last 48 bit block of the second subframe.
- The second subframe carries either 71 or 72 bits for the second DS1, and 72 bits for all the other DS1s.
- The third and fourth DS1s are handled the same way, using the third and fourth subframes.

DS2 Multiplexing

- **Each DS2 frame, therefore, carries either 287 or 288 bits per DS1.**
- **This means that the tolerance for the DS1s is from 1.5404Mbps to 1.5458Mbps.**

$$\frac{6.312 * 287}{1176} = 1.5404 \text{ Mbps}$$

$$\frac{6.312 * 288}{1176} = 1.5458 \text{ Mbps}$$

The DS2 frame rate is about 5,362+ frames per second.

Notes on DS2

- The frame rate is NOT 8,000 times per second. It's about 5,362+ frames per second.
- The DS2 frame just carries bits from the DS1s. It does not attempt to figure out alignment of the DS1 or where the voice channels are.
- This means that to extract a voice channel, you have to demux back to the DS1, find sync, and then extract the channel.

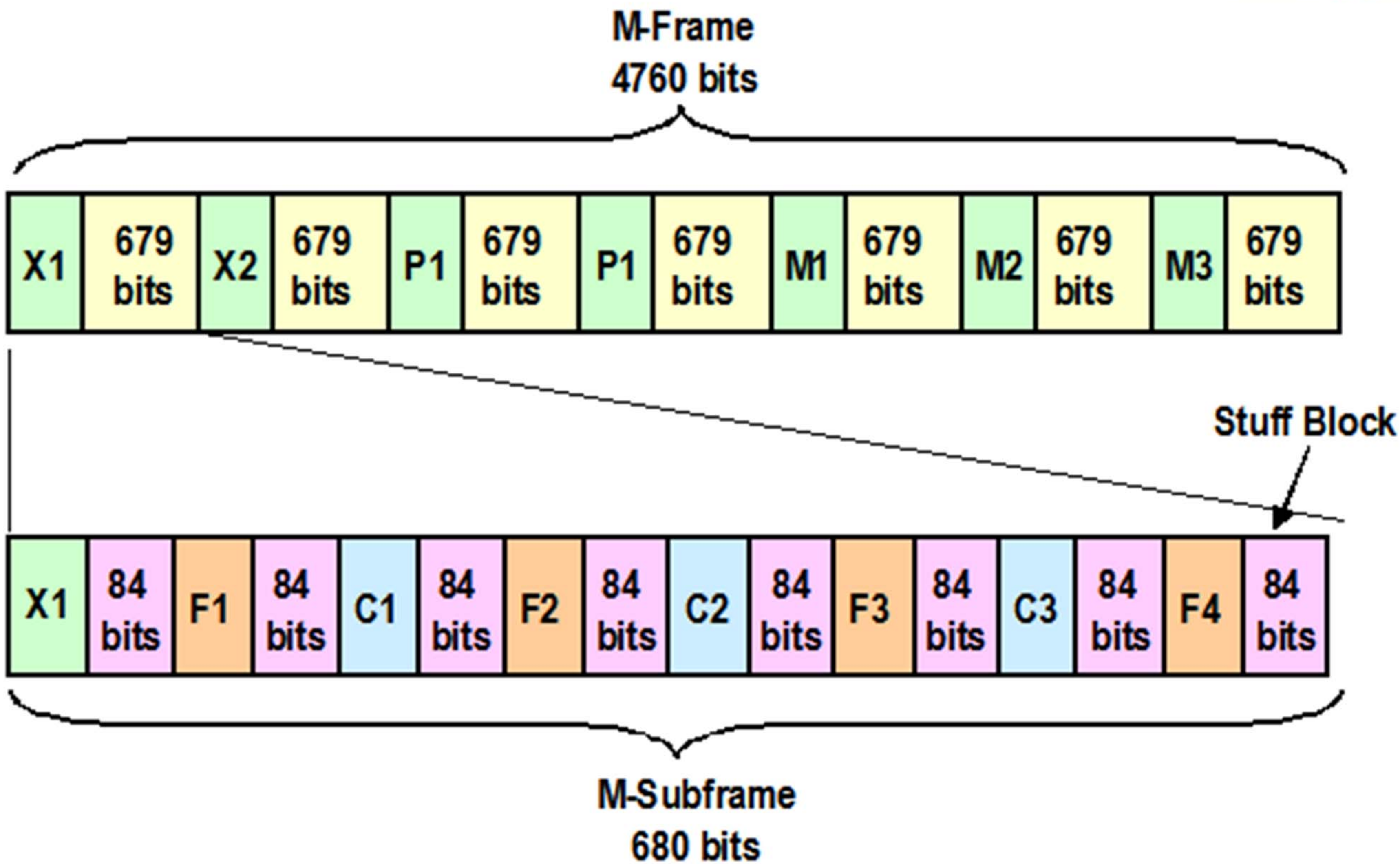
DS3 Multiplexing

- DS2's are no longer used in the network – they only exists inside an M13 multiplexer where the DS1s are multiplexed into a DS2 stream before the DS2 streams are multiplexed into the DS3 stream.
- Inside the M13 multiplexer, the DS2 rate is set to 6.306272 Mbps instead of 6.312 Mbps, which places a tighter constraint on the DS1 rates, but is still very easy to meet. The reason for 6.306272 Mbps will be seen when we look at muxing the DS2s into a DS3.
- The nominal DS3 data rate is 44.736Mbps.

DS3 multiplexing

- A DS3 signal carries seven DS2 signals, each of which carries four DS1 signals, so the DS3 carries 28 DS1s.
- This is 672 voice channels (or DS0s).
- Let’s look at how the DS2s mux into the DS3.

DS3 Frame Structure



DS3 Frame Synchronization

- M-Subframe synchronization is found from the F bits (F1, F2, F3, and F4), which have the pattern {1, 0, 1, 0}.
- Once subframe synchronization is found, M-Frame synchronization is found by looking at the M bits (M1, M2, and M3) which have the pattern {0, 1, 0}.
- The {X1,X2} and {P1,P2} bits are either {0,0} or {1,1} which means that no combination of M, X, and P bits can have the sequence {0,1,0}.
- Note: The X bits indicate received defects. The P bits are parity over the block. The data indicated in X1 or P1 is just duplicated in X2 or P2.

DS3 Stuffing

- **Bits of the DS2s are bit interleaved into the 84 bit blocks in exactly the same fashion as the DS1 are bit interleaved into the DS2 blocks.**
- **The last block of the seven M-subframes is used as the stuff block**
  - The first bit in the last block of the first M-Subframe is the stuff bit for the first DS2.
  - The second bit in the last block of the second M-Subframe is the stuff bit for the second DS2.
  - Etc.

DS3 Stuffing (continued)

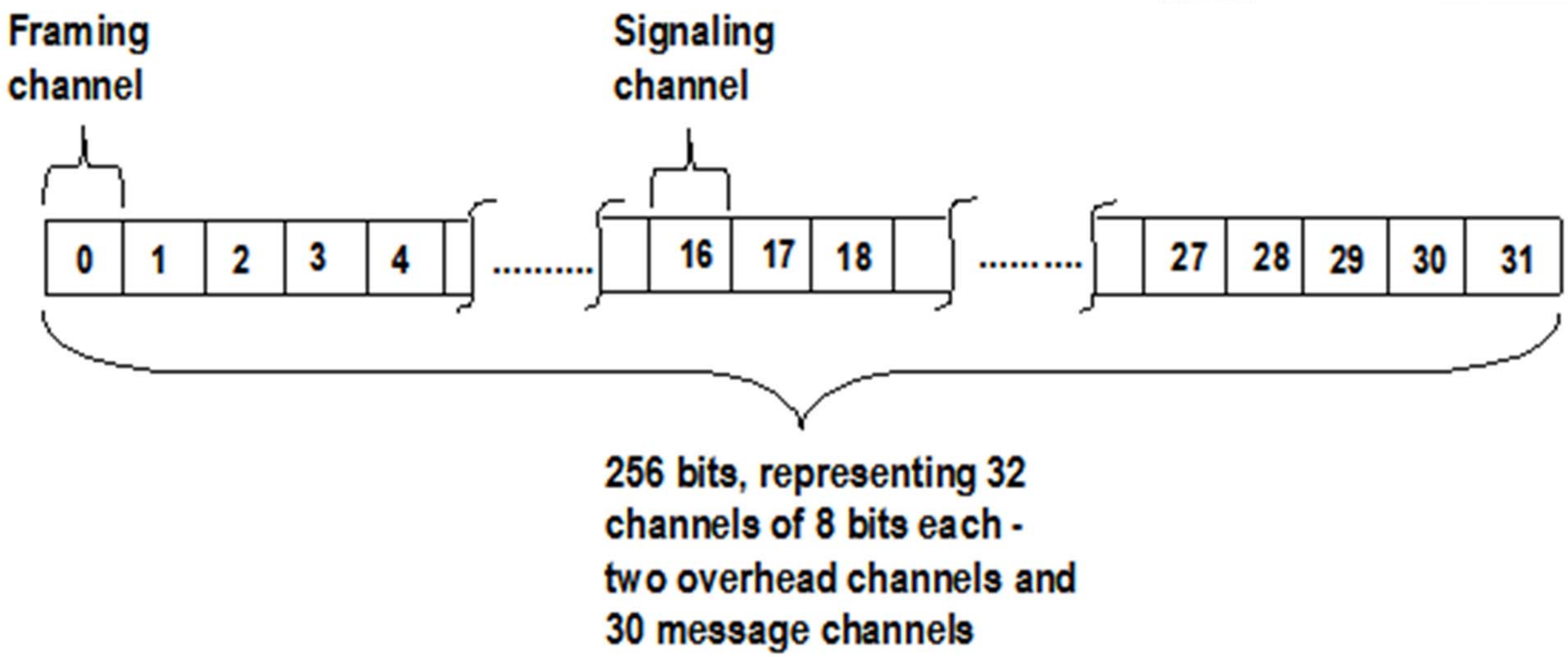
- **Stuffing is controlled by the C bits, with all ones indicating a stuff bit and all zeroes indicating a data bit.**
- **Majority voting applies.**
- **Each M-Subframe carries either 95 or 96 bits for each DS2.**
- **The M-Frame carries 671 or 672 bits for each DS2.**
- **This means that the DS3 can accept DS2 data rates from 6.306272Mbps to 6.31567Mbps. But the DS2 rate is set to 6.306272 so there should be 100% stuffing.**

$$\frac{44.736 * 671}{4760} = 6.306272 \text{ Mbps} \qquad \frac{44.736 * 672}{4760} = 6.31567 \text{ Mbps}$$

E1 Framing

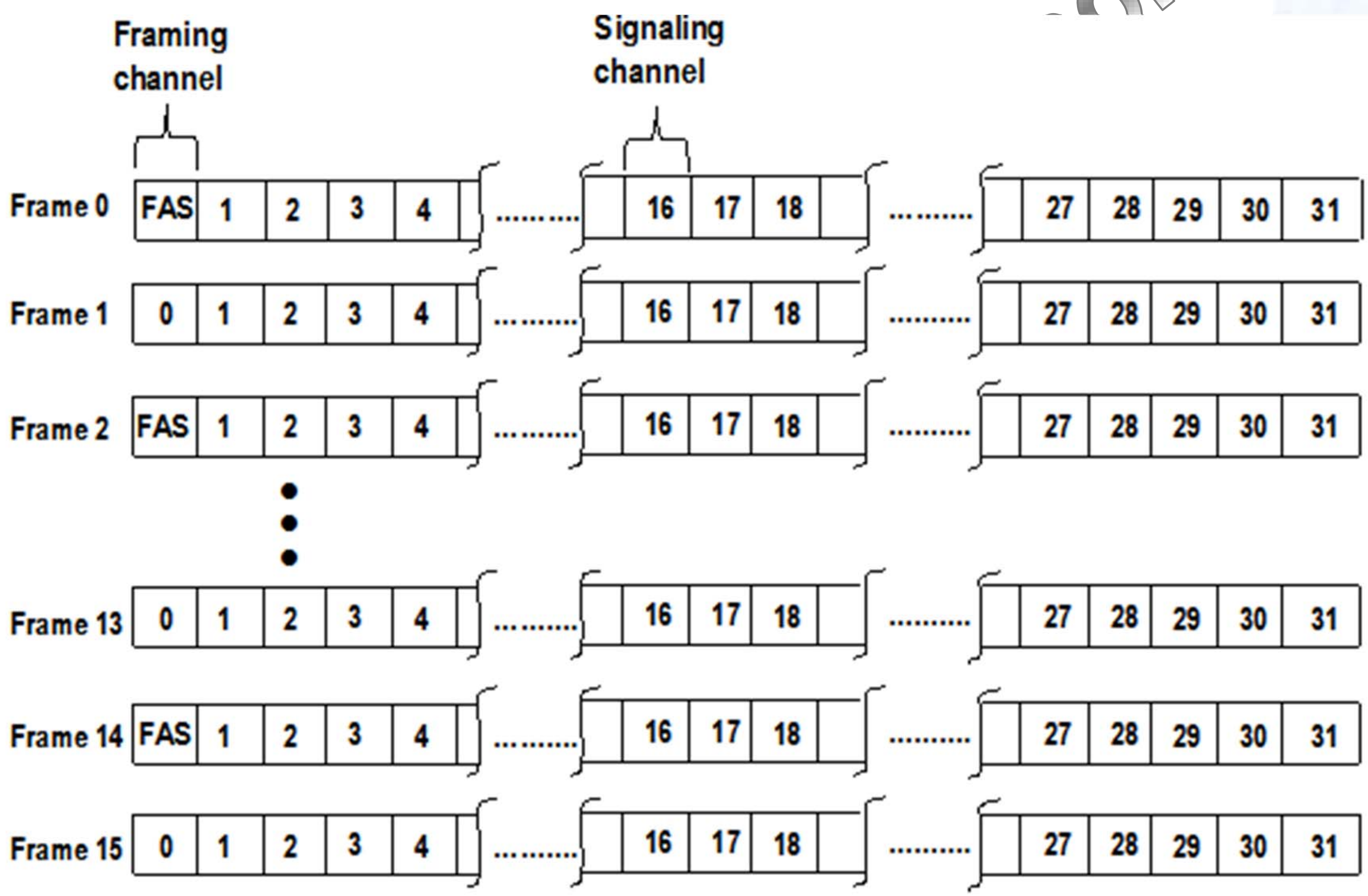
- The E series framing is a bit easier than the DS framing. Perhaps the Europeans learned from the mistakes of the Americans.
- The E1 frame is 256 bits in length (32 octets) and carries 30 voice channels.
- The first octet is the framing octet and the 17th octet (numbered 16 because numbering starts at zero) is the signaling channel.
- The frame is shown on the next slide.

E1 Framing


Framing
channel
Signaling
channel
0
1
2
3
4
....
16
17
18
....
27
28
29
30
31
256 bits, representing 32
channels of 8 bits each -
two overhead channels and
30 message channels

E1 MultiFrame

- **Sixteen of the 256 bit frames makes up a Multi-Frame**



E1 MultiFrame

- The first octet of each frame is allocated to either the Frame Alignment Signal (FAS) or for a message link for operations.
- Since the FAS only occurs every other frame, for calculation of the synchronization time, the effective block is 512 bits.
- The FAS is the sequence {0, 0, 1, 1, 0, 1, 1}, carried in the low order 7 bits of the first octet of every other frame.
- In the non-FAS frames, the second bit of the first octet is forced to 1 to guarantee that the bits in that octet can never have the pattern {0, 0, 1, 1, 0, 1, 1}.
- Finding the FAS sequence gives us sync on the frames, but we also need to find the beginning of the MultiFrame.

E1 MultiFrame Sync

- The first bit of each non-FAS frame will carry the pattern {0,0,1,0,1,1,E,E}. {E,E} will be {0,0} until sync.

	Sub-multiframe number	Frame number	Bit 1 of each frame
Multiframe	I	0	C_1
		1	0
		2	C_2
		3	0
		4	C_3
		5	1
		6	C_4
		7	0
	II	8	C_1
		9	1
		10	C_2
		11	1
		12	C_3
		13	E
		14	C_4
		15	E

E1 Synchronization Time

- In E1 we have 7 bits of framing so a different calculation needs to be used to calculate the synchronization time.

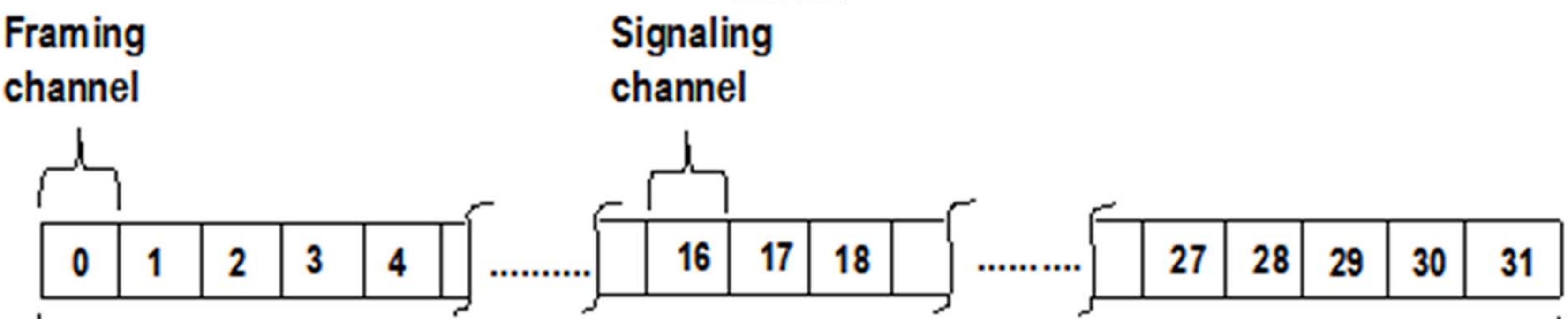
$$\text{Frame time} = \frac{N^2}{2(2^L - 1)} + \frac{N}{2} \quad \text{Bit times}$$

N = 512 (number of bits in a block, including the framing)
L = 7 (number of framing bits)

The average framing time is 1,288 bit times, about 0.63 ms, much faster than DS1 because of the larger number of framing bits and the fact that the framing bits do not alternate - they are constant for each block. After frame synchronization is found the system must find the multiframe sync but that will be fairly quick once the system knows the octet boundaries.

E1 Signaling Channel

- **Octet 16 in each frame is allocated to a signaling channel.**
- **Normally uses LAP-D but can do an A/B/C/D type signaling as is done in DS1.**

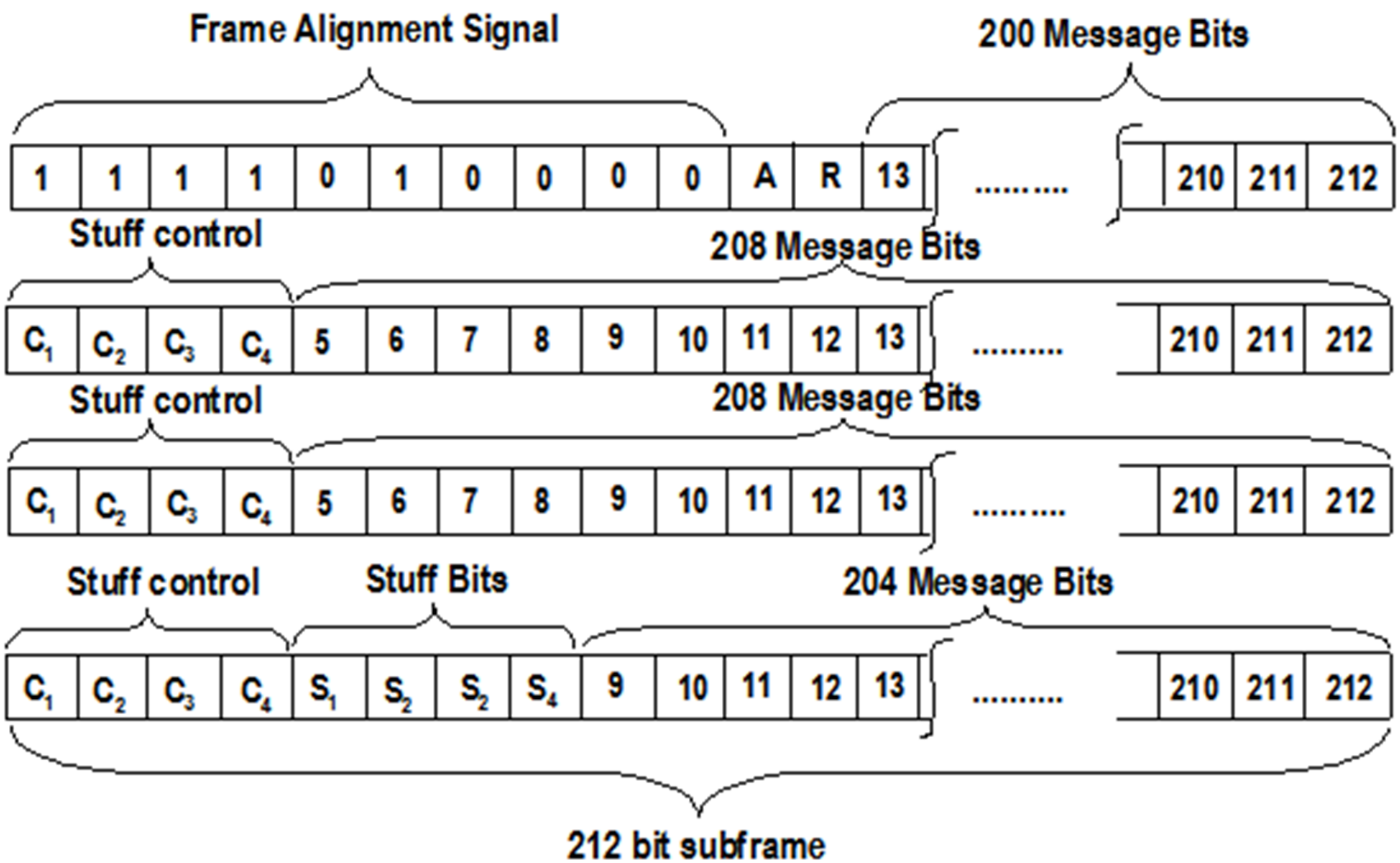


E2 Framing and Multiplexing

- Four E1 lines are multiplexed into an E2 line.
- The nominal rate for an E2 line is 8.448Mbps.
- Four E1 lines at 2.048Mbps gives a rate of 8.192Mbps so the E2 line has excess bits to handle clock variation and framing.
- Let’s look at the E2 framing.

E2 Frame

- **848 bits in size, 820 slots for E1 traffic, with four stuff slots.**



E2 Framing

- The Frame Alignment Signal is the first 10 bits {1, 1, 1, 1, 0, 1, 0, 0, 0, 0} of the frame.
- The overall frame size is 848 bits.
- The frame is divided into four subframes of 212 bits each.
- 820 bit positions are for traffic and are divided into 205 bit positions for each E1.
- Four stuff positions are provided, one for each E1.
- So the E2 frame will carry either 205 or 206 bits for each E1.

E2 Framing

- **Stuffing is controlled by the C1, C2, C3, and C4 bits (three of each).**
  - Ones in the three control positions (for example, C1, C1, C1) indicate data in the stuff position. Zeroes indicate it contains a stuff bit.
  - This provides the ability to handle E1s from 2.042264Mbps to 2.05226Mbps (2.048Mbps nominal).

$$\frac{\mathbf{8.448*205}}{\mathbf{848}} = \mathbf{2.042264}\ \text{Mbps}$$

$$\frac{\mathbf{8.448*206}}{\mathbf{848}} = \mathbf{2.052226}\ \text{Mbps}$$

E2 Framing

- Average frame sync time is about 0.1ms

Frame time = $\frac{N^2}{2(2^L-1)} + \frac{N}{2}$ Bit times

N = 848 (number of bits in a frame)
L = 10 (number of framing bits)

Notes on E2

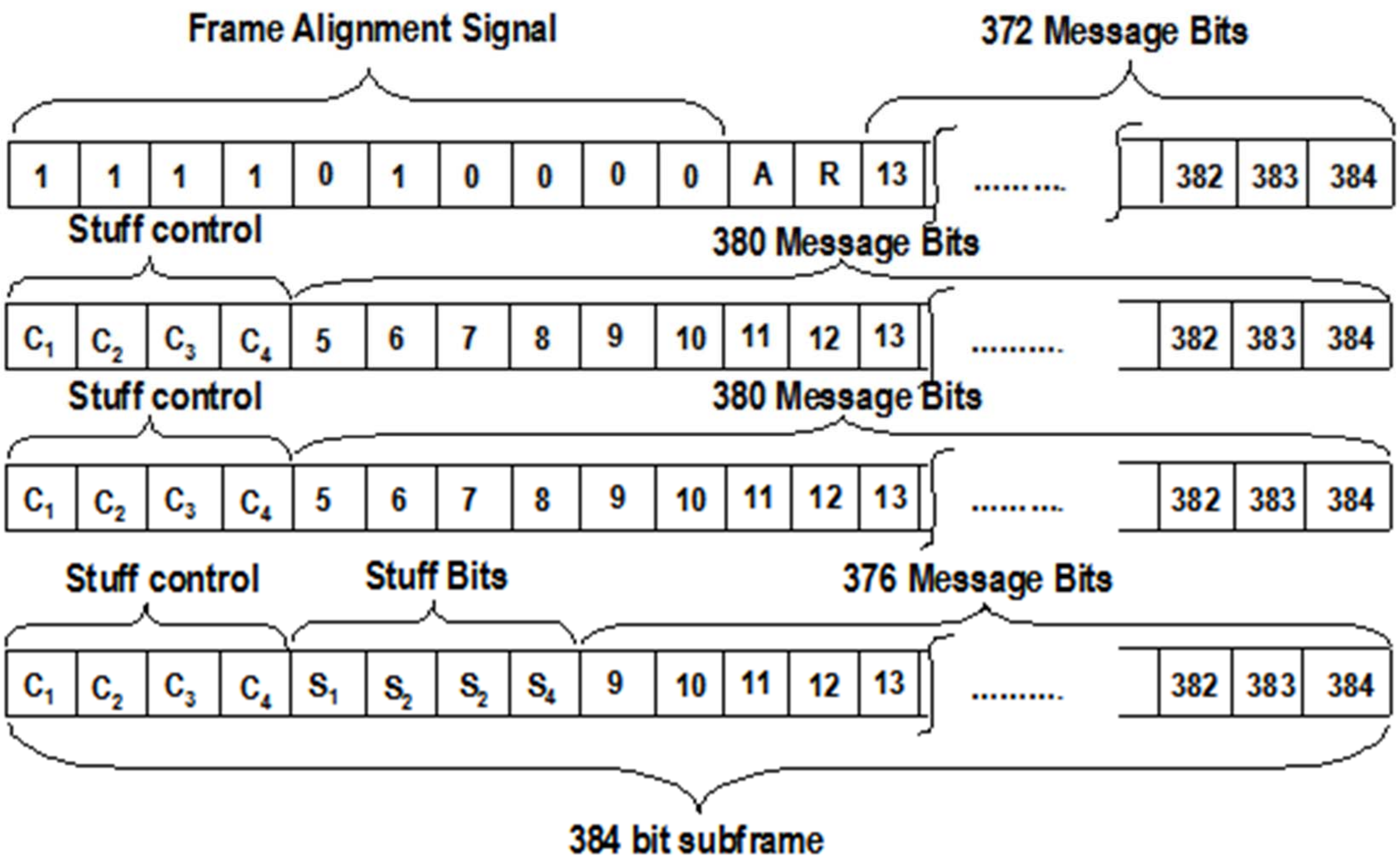
- The frame rate is NOT 8,000 times per second. It's about 9,962+ frames per second.
- The E2 frame just carries bits from the E1s. It does not attempt to figure out alignment of the E1 or where the voice channels are.
- This means that to extract a voice channel, you have to demux back to the E1, find sync, and then extract the channel.

E3 Framing and Multiplexing

- Four E2 lines are multiplexed into an E3 line.
- The nominal rate for an E3 line is 34.368Mbps.
- Four E2 lines at 8.448 Mbps gives a rate of 33.792Mbps so the E3 line has excess bits to handle clock variation and framing.
- Let's look at the E3 framing.

E3 Frame

- **Follows the basis structure of the E2 frame, but bigger.**



E3 Framing

- The Frame Alignment Signal is the first 10 bits {1, 1, 1, 1, 0, 1, 0, 0, 0, 0} of the frame.
- The overall frame size is 1536 bits.
- The frame is divided into four subframes of 384 bits each.
- 1508 bit positions are for traffic and are divided into 377 bit positions for each E2.
- Four stuff positions are provided, one for each E2.
- So the E3 frame will carry either 377 or 378 bits for each E2.
- Note how similar this is to the E2 framing of E1s.

E3 Framing

- **Stuffing is controlled by the C1, C2, C3, and C4 bits (three of each).**
  - Ones in the three control positions (for example, C1, C1, C1) indicate data in the stuff position. Zeroes indicate it contains a stuff bit.
  - This provides the ability to handle E2s from 8.435375Mbps to 8.45775Mbps (8.448Mbps nominal).

$$\frac{\mathbf{34.368*377}}{\mathbf{1536}} = \mathbf{8.435375}\ \text{Mbps}$$

$$\frac{\mathbf{34.368*378}}{\mathbf{1536}} = \mathbf{8.45775}\ \text{Mbps}$$

E3 Framing

- **Average frame sync time is about 0.05ms**

$$\text{Frame time} = \frac{N^2}{2(2^L - 1)} + \frac{N}{2} \text{ Bit times}$$

N = 1536 (number of bits in a frame)
L = 10 (number of framing bits)

Problems of the PDH

- **The two major global PDH systems are incompatible.**
- **Reliability is not part of the system. No automatic recovery from losses of lines or equipment.**
- **Management and monitoring is not part of the PDH.**
- **No drop and insert. To add or remove a DS1/E1 the entire data stream must be demuxed and then remuxed, which adds up to lots of equipment at a node.**
  - For telephone calls, for example, the data stream must be demuxed all the way down to the DS0/E0.

Problems of the PDH

- Too much clock variation in the network. Makes dealing with multiple data streams complex. Clocks are not multiples of each other.
- PDH is bit oriented so the electronics usually has to clock at the bit rate. The data rates possible across fiber outran the ability of silicon to keep up at the bit rate. SONET/SDH is octet oriented so once the octet boundary is found, the electronics can operate at the octet rate, 1/8 of the bit rate. In the early days, electronics that operated at the optical bit rate had to be GaAs.

SONET/SDH

What is SONET/SDH

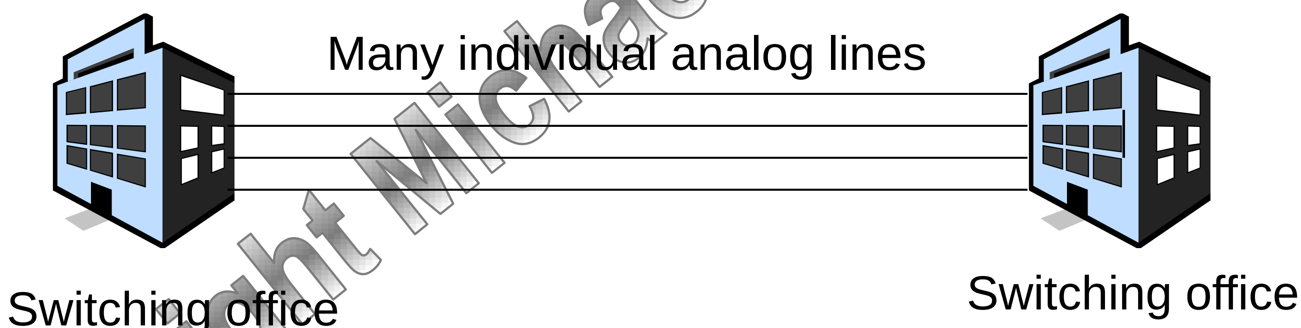
- **SONET – Synchronous Optical NETwork (ANSI).**
  - ANSI started work on SONET in1985.
- **SDH – Synchronous Digital Hierarchy (ITU).**
  - The ITU began work in 1986 to achieve the same goal.
- **SONET and SDH define a set of physical layer standards for communications over optical fiber.**

I will attempt to cover both SONET and SDH in this presentation. However, SONET and SDH use different terminology, which makes it difficult to talk about both at the same time.

I will talk mainly about SONET because you need to understand SONET to understand why some things were done in SDH.

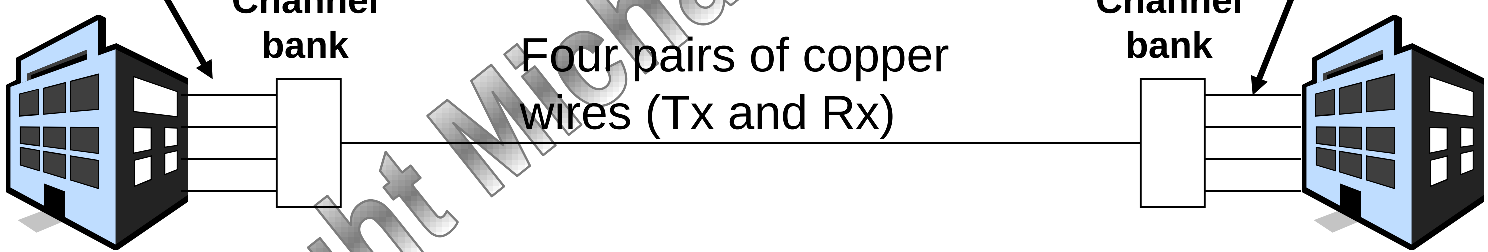
Why SONET/SDH?

- **Originally, all communications in the telephone network was analog.**
- **Analog lines or analog microwave was used to connect to switching offices.**



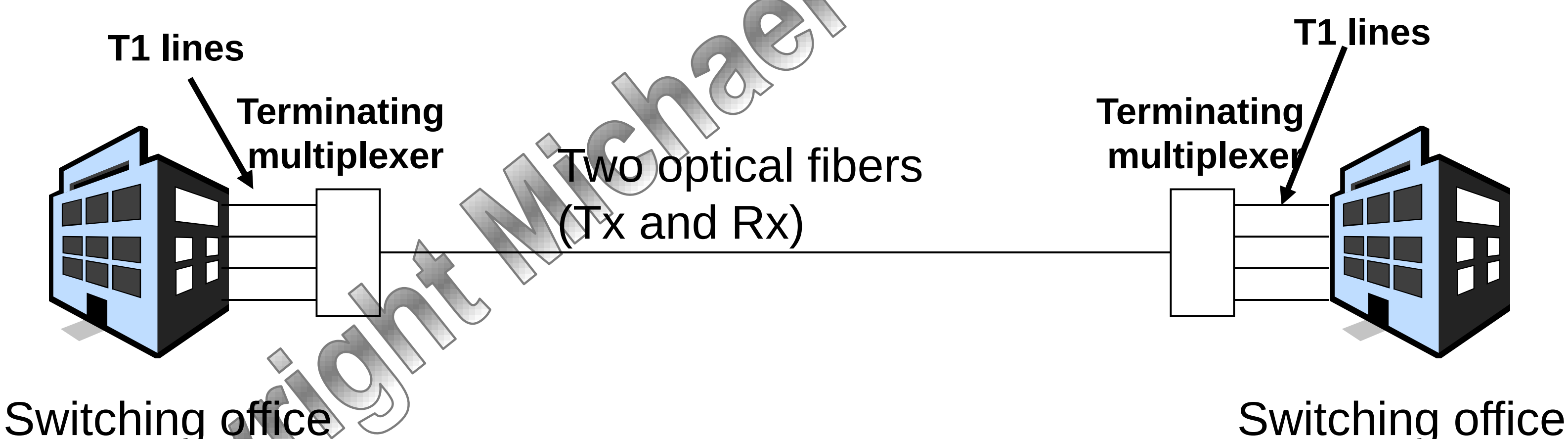
Why SONET/SDH?

- In about 1962, the network providers began using digital communications between switching centers.
- In the US, this was the D1 channel banks and T-carrier systems.



Why SONET/SDH?

- As communications needs grew, many T- or E-carrier lines were needed between switching centers.
- In the late 1970's optical communications began to be used to interconnect switching offices.



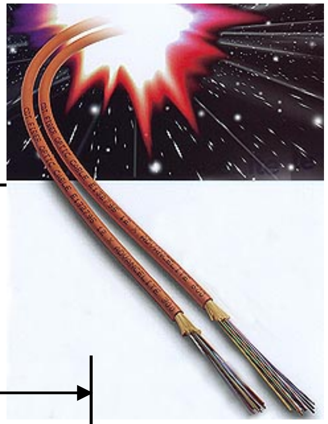
Standardization of SONET/SDH

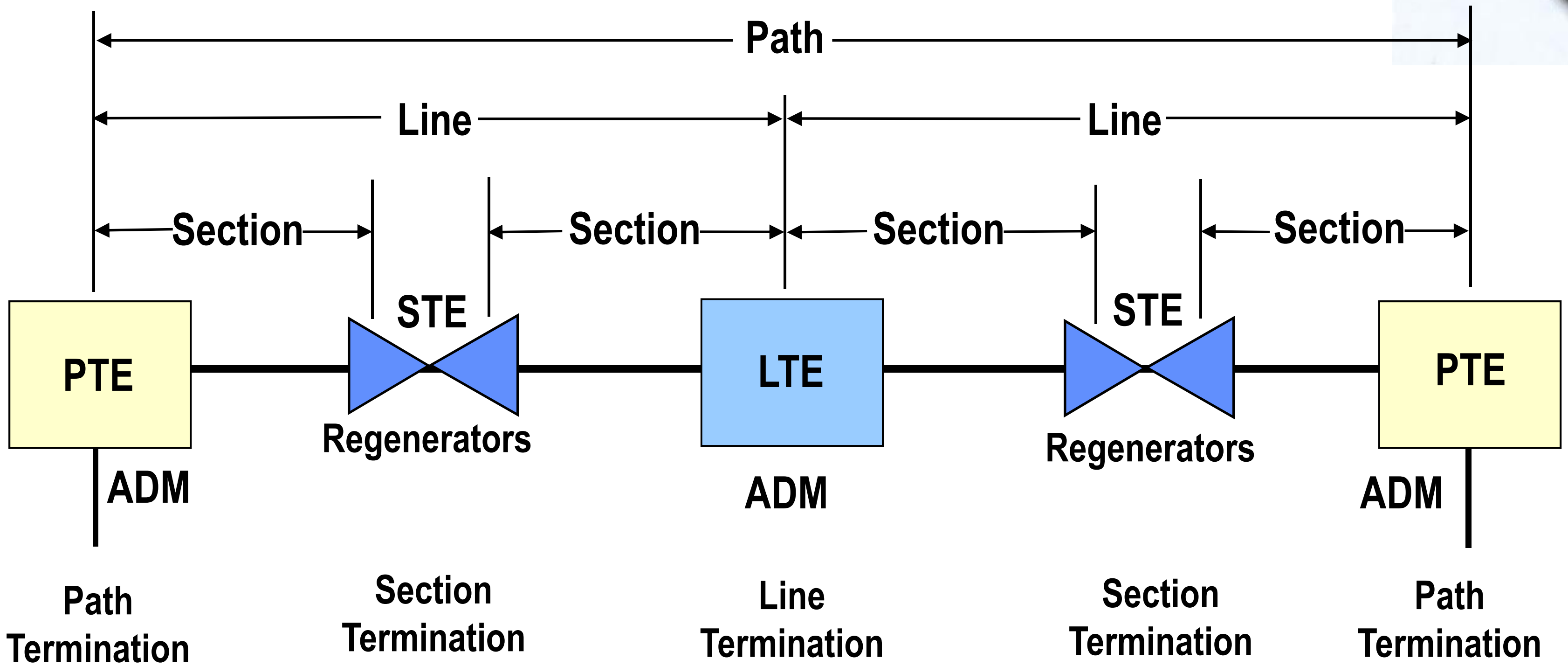
- **Prior to standardization, every manufacturer of optical communications used their own framing.**
  - Did not allow multi-vendor networks.
- **The ANSI T1X1.5 committee began work in 1985 to define standards for optical communications which would allow "a mid-span meet".**
- **The ITU began work in 1986 to achieve the same goal.**
- **Both bodies finalized their first set of standards in 1988.**
- **SONET and SDH can, and do, interoperate.**

Why do we need SONET/SDH

- **All data requires framing.**
- **Since optical networks are complex, provisions are made for management information.**
- **Many other things are provided.**
  - Multiplexing.
  - Error checking.
  - Handling variations in clocks.
  - Mapping of plesiochronous voice and data traffic.
  - Signaling for automatic switching in case of a fiber or node failure.

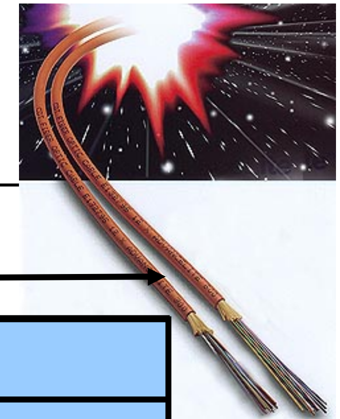
SONET/SDH Terminology

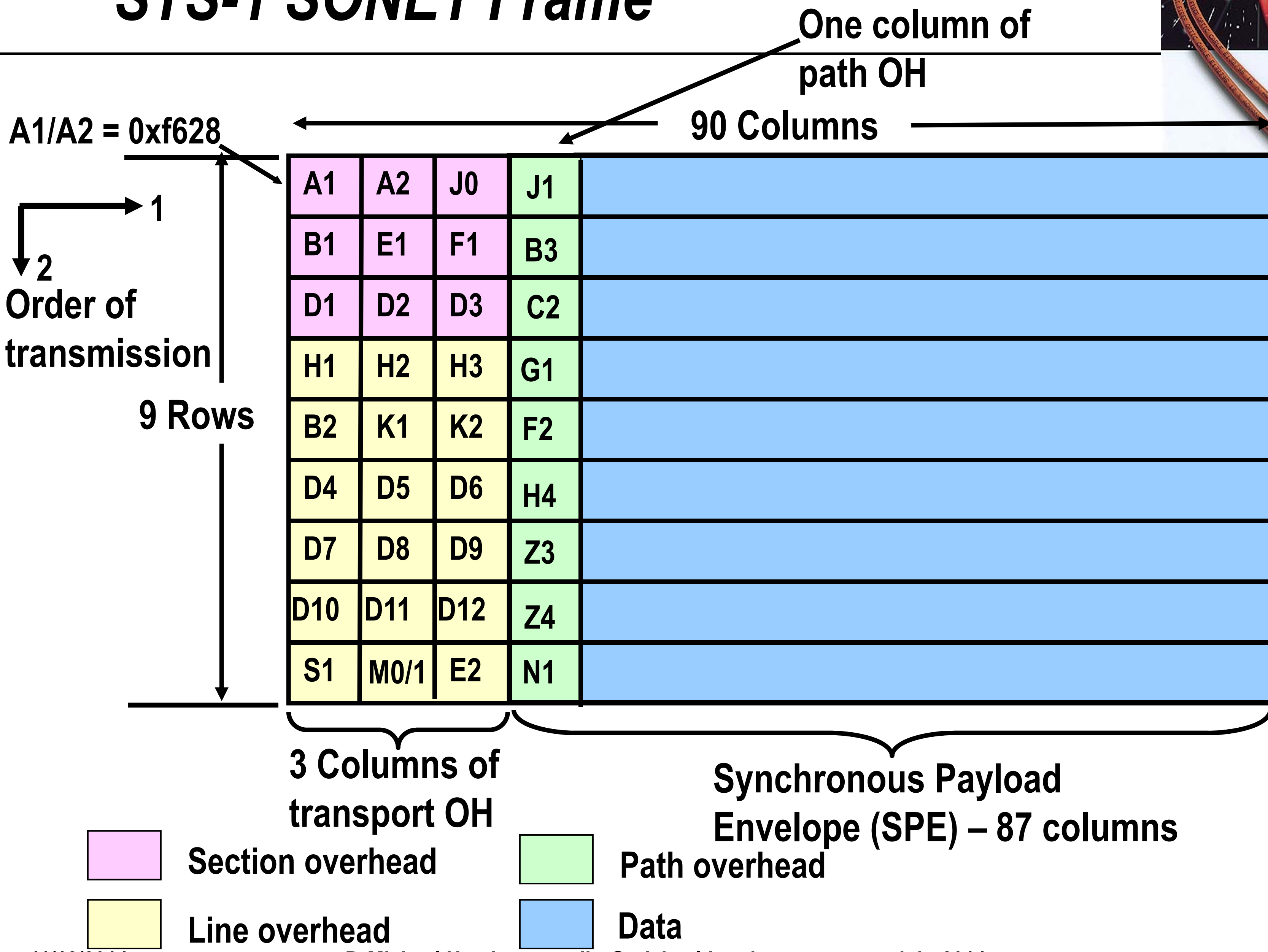




PTE = Path Terminating Equipment
LTE = Line terminating Equipment
STE = Section Terminating Equipment
ADM = Add/Drop Multiplexer

STS-1 SONET Frame

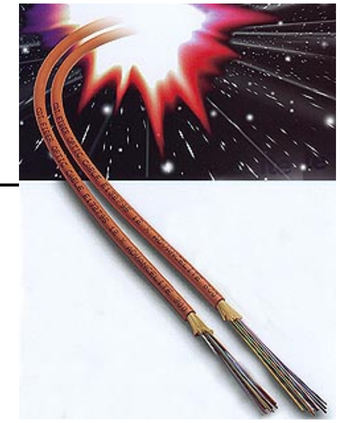




STS-1 Frame Format

- 9 rows by 90 columns – 810 octets in the frame.
- Frame is transmitted from left to right, by row.
- Frames are transmitted 8,000 times per second, every 125 μseconds.
- STS-1 bit rate is therefore 51.84 Mbps (810 octets x 8,000 times per second x 8 bits per octet).
- This lowest level SONET signal is called a Synchronous Transport Signal, level 1 (STS-1). Once the scrambler is applied, it is known as an Optical Channel, level 1 (OC-1).
- The lowest level SDH signal is known as a Synchronous Transport Module, level 1 (STM-1).

Transmission order for basic SONET frame



Order of transmission

1

2

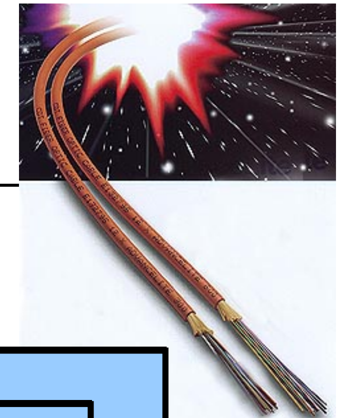
A1	A2	J0	J1	
B1	E1	F1	B3	
D1	D2	D3	C2	
H1	H2	H3	G1	
B2	K1	K2	F2	
D4	D5	D6	H4	
D7	D8	D9	Z3	
D10	D11	D12	Z4	
S1	M0/1	E2	N1	

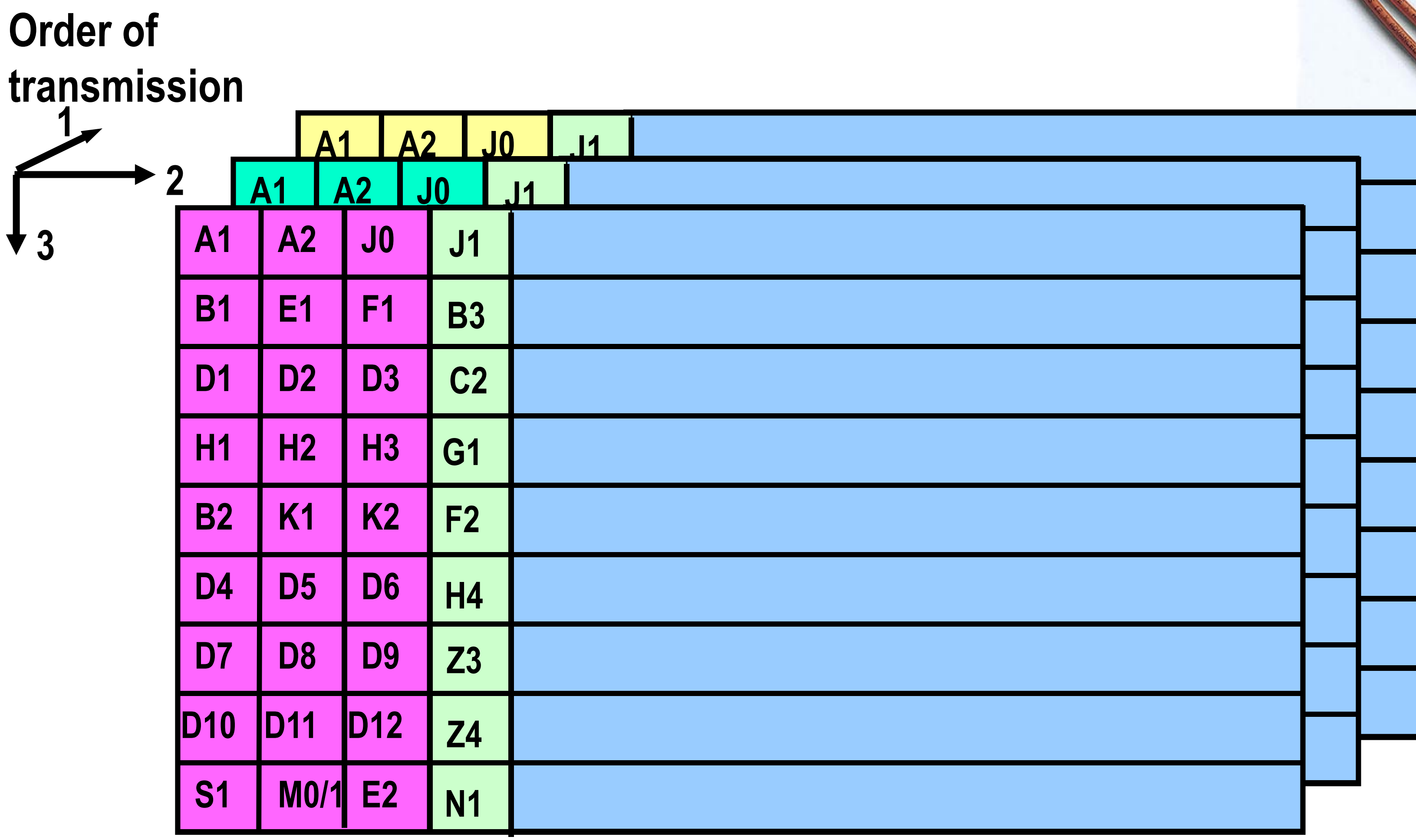
SONET/SDH data rates

SONET name	SDH name	Line rate (Mbps)	SPE rate (Mbps)	Overhead rate (Mbps)
OC-1	STM-0	51.84	50.112	1.728
OC-3	STM-1	155.52	150.336	5.184
OC-12	STM-4	622.08	601.344	20.736
OC-48	STM-16	2,488.32	2,405.376	82.944
OC-192	STM-64	9,953.28	9,621.504	331.776
OC-768	STM-256	39,813.12	38,486.016	1,327.104

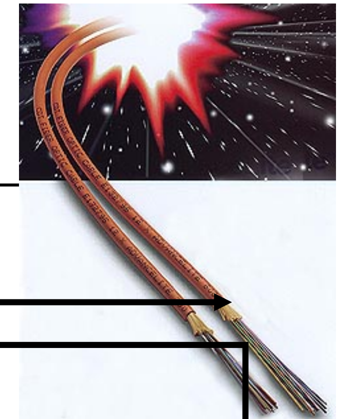
Overhead is a constant 3.33%

Interleaving of SONET/SDH signals





Interleaving of SONET/SDH signals



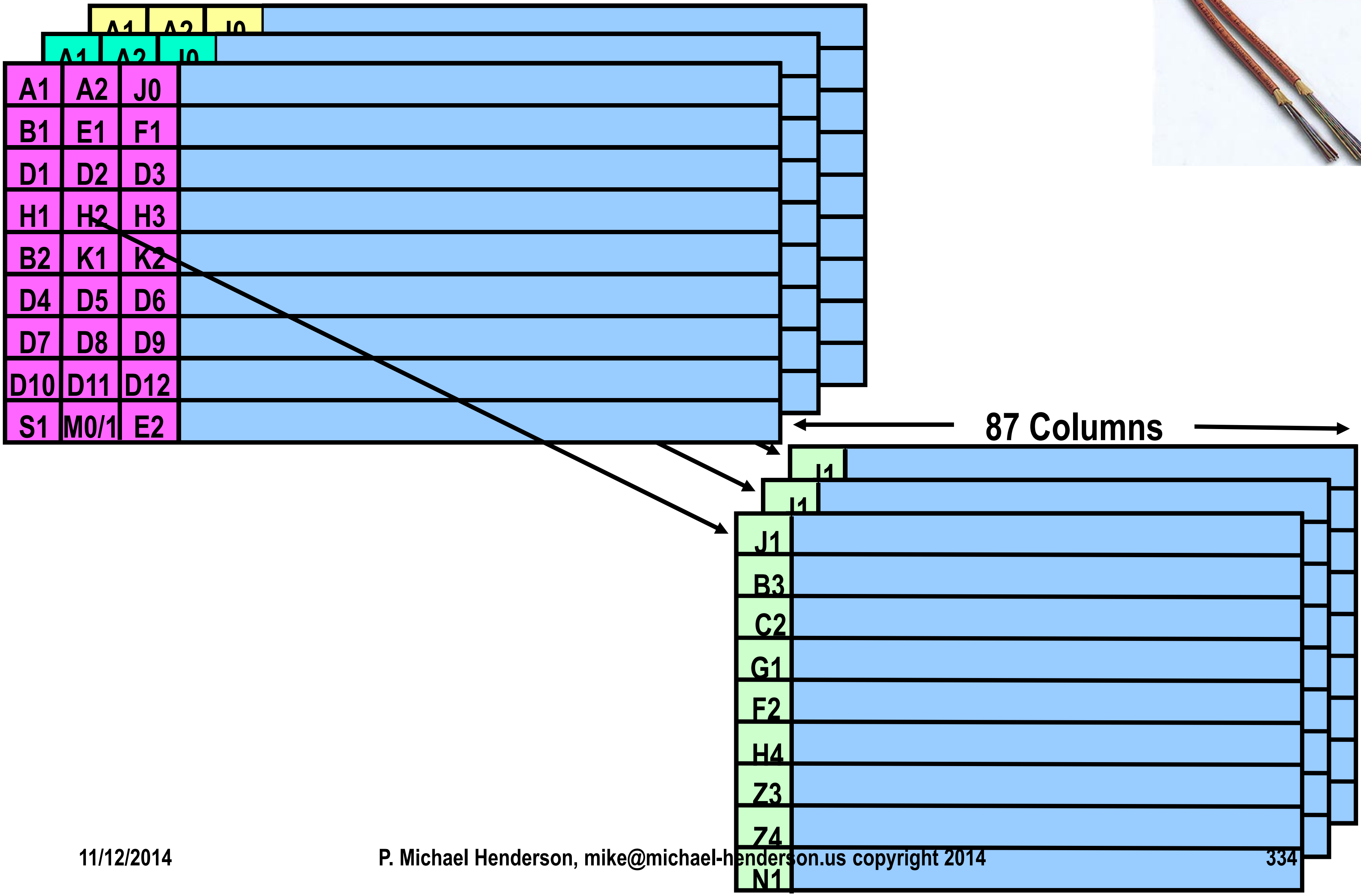
270 Columns

A1	A1	A1	A2	A2	A2	J0	Z0	Z0	
B1	X	X	E1	X	X	F1	X	X	
D1	X	X	D2	X	X	D3	X	X	
H1	H1	H1	H2	H2	H2	H3	H3	H3	
B2	B2	B2	K1	X	X	K2	X	X	
D4	X	X	D5	X	X	D6	X	X	
D7	X	X	D8	X	X	D9	X	X	
D10	X	X	D11	X	X	D12	X	X	
S1	Z1	Z1	M0/1	Z2	M2	E2	X	X	

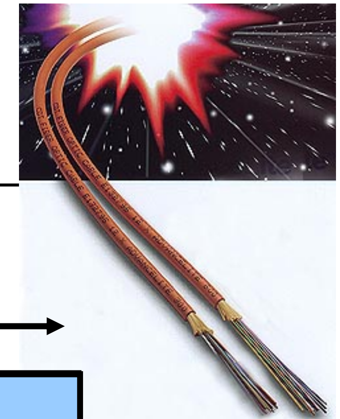
First STS-1 Second STS-1 Third STS-1

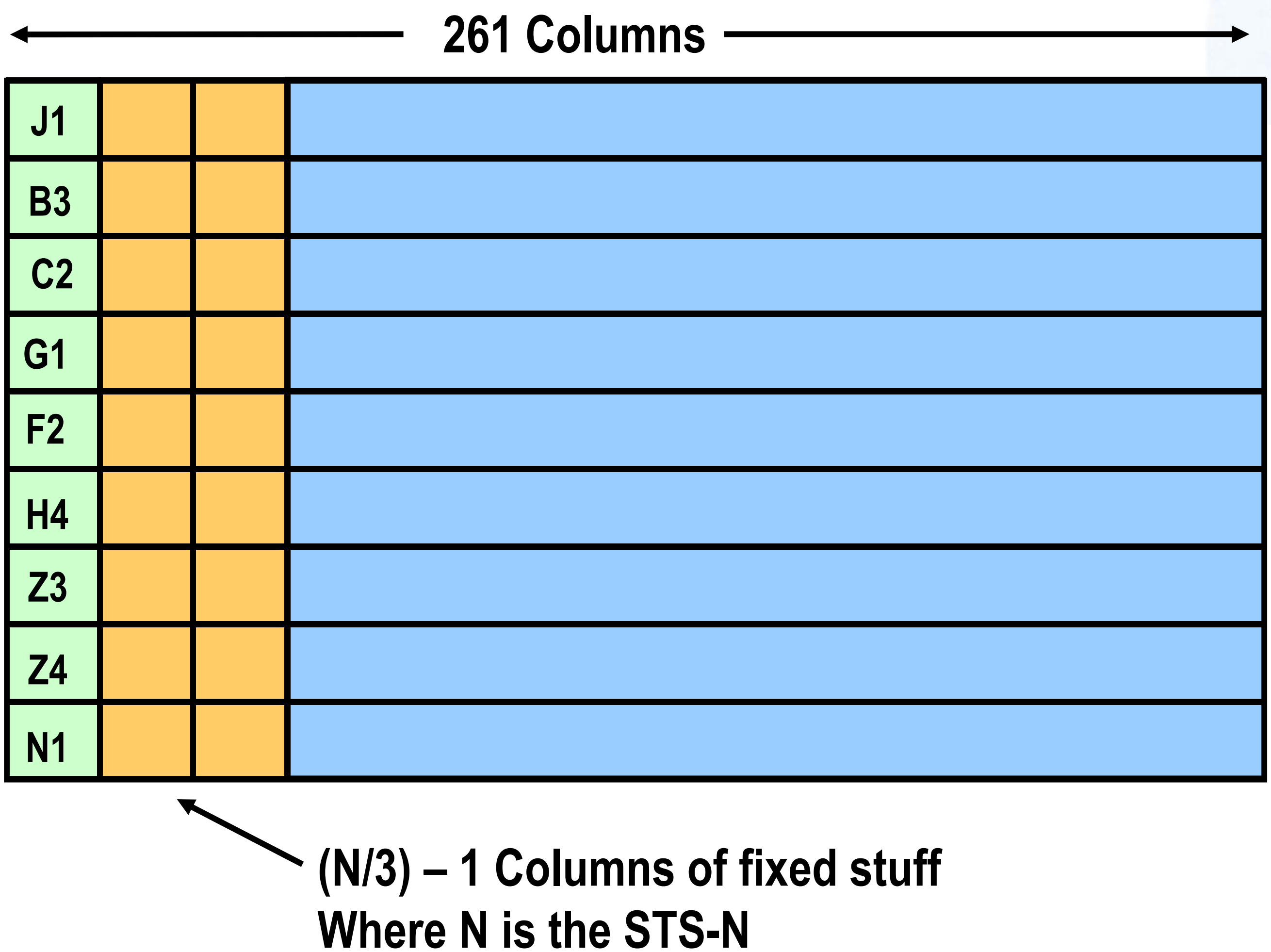
This is the format of a SDH STM-1

Payloads in Interleaved Signals

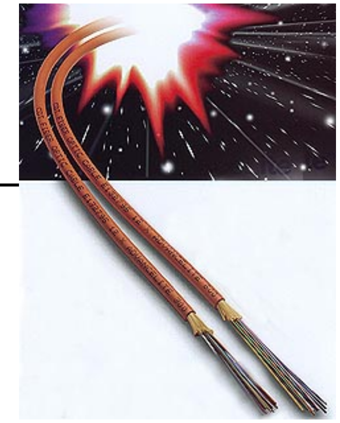


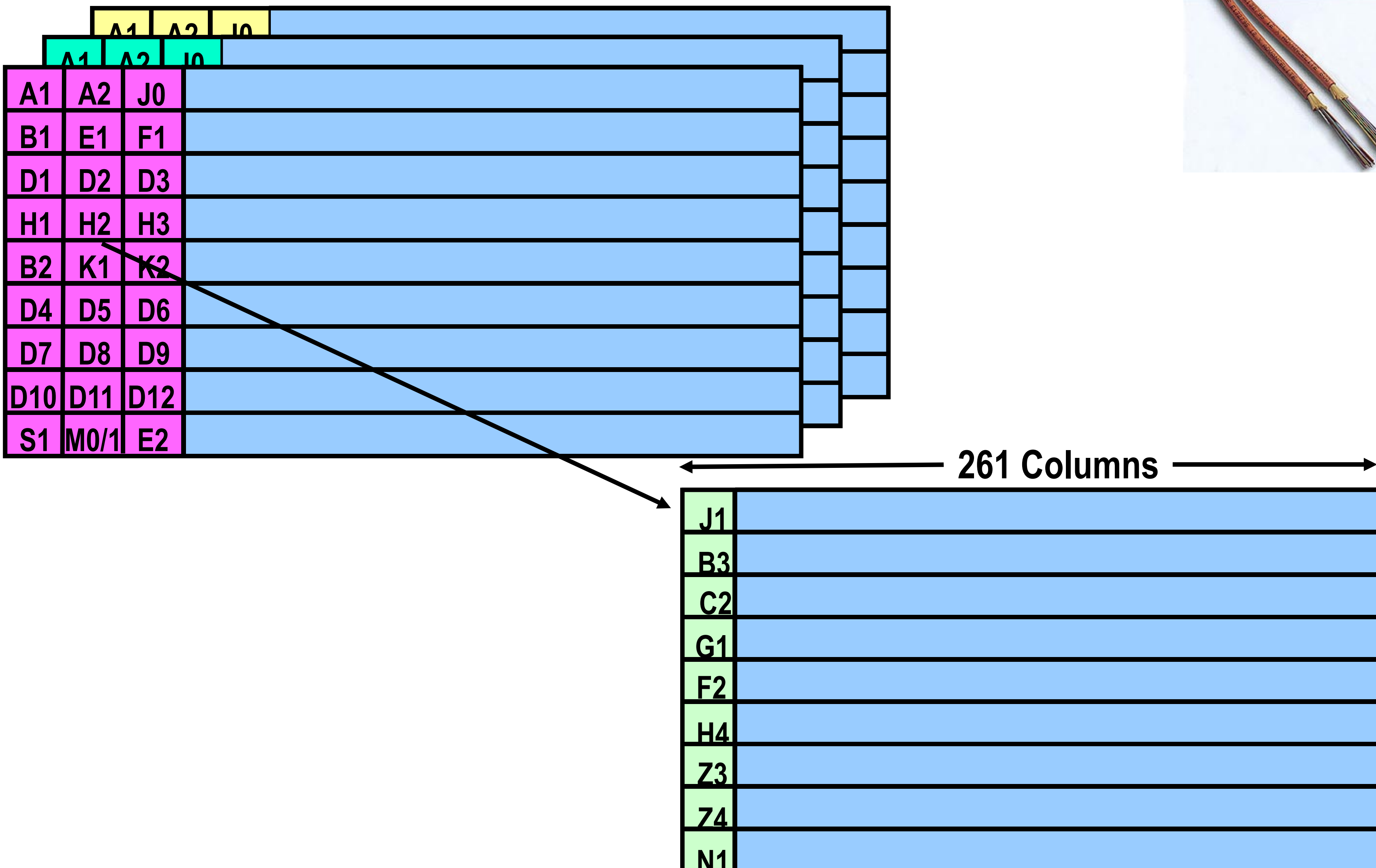
Concatenated Payloads





Concatenated Payloads





Framing and Section Trace

A1	A2	J0
B1	E1	F1
D1	D2	D3
H1	H2	H3
B2	K1	K2
D4	D5	D6
D7	D8	D9
D10	D11	D12
S1	M0/1	E2

- Two octets used for framing: A1, A2.
- Bit pattern is A1 = 1111 0110, A2 = 0010 1000
- Bit pattern is DC balanced.
- Higher levels of STS-N have N A1 octets and N A2 octets.
- Section Trace (J0) is used to verify continued connection between section entities. A 16 octet message is sent by sending one octet each frame.

Monitoring for Bit Errors

A1	A2	J0
B1	E1	F1
D1	D2	D3
H1	H2	H3
B2	K1	K2
D4	D5	D6
D7	D8	D9
D10	D11	D12
S1	M0/1	E2

- Does a bit interleaved parity (BIP-8) over certain octets in the (previous) frame.
- Even parity.
- Parity check applies to *previous* frame
- Separate check for section and line.
- Only one B1 octet, no matter what the STS-N.
- One B2 octet for each STS. So, for an STS-3 there will be 3 B2 octets. For STS-12, there will be 12 B2 octets.

Bit Interleaved Parity

- For BIP-8, parity will be checked for each octet (each 8 bits) of the block to be checked.
- The number of bits in each bit position is totaled, and the corresponding bit in the parity octet is set to give even parity.

Example of BIP-8

"Monitored portion"

8-bits word	8-bits word	8-bits word	8-bits word	8-bits word
Byte 5	Byte 4	Byte 3	Byte 2	Byte 1

Bit position	1	2	3	4	5	6	7	8
Word 1	1	0	1	0	1	0	1	0
Word 2	0	1	0	1	0	1	0	1
Word 3	1	0	1	0	1	0	1	0
Word 4	0	1	0	1	0	1	0	1
Word 5	1	1	0	0	1	1	0	0
«1's» count	3	3	2	2	3	3	2	2
Odd/Even	Odd	Odd	Even	Even	Odd	Odd	Even	Even

An odd total in the "1's" count row causes a binary "1" to be placed in the same position below.

BIP-8 code	1	1	0	0	1	1	0	0

BIP- 8 calculation process

Communication Channels

A1	A2	J0
B1	E1	F1
D1	D2	D3
H1	H2	H3
B2	K1	K2
D4	D5	D6
D7	D8	D9
D10	D11	D12
S1	M0/1	E2

- **E1, E2 (orderwire) was intended to be used by craftspersons while installing a line. Craftspeople use cellular phones so this octet is not used very much.**
- **F1 is available for the network provider to use as they wish.**

Communication Channels

A1	A2	J0
B1	E1	F1
D1	D2	D3
H1	H2	H3
B2	K1	K2
D4	D5	D6
D7	D8	D9
D10	D11	D12
S1	M0/1	E2

- These channels are used for network management.
- Technically, D1, D2, D3 is intended for section messages but this is not always adhered to. They create a 192Kbps channel between STEs.
- D4 through D12 is used to create a 576Kbps channel between the LTEs (Line Termination Equipment).

Automatic Protection Switching

A1	A2	J0
B1	E1	F1
D1	D2	D3
H1	H2	H3
B2	K1	K2
D4	D5	D6
D7	D8	D9
D10	D11	D12
S1	M0/1	E2

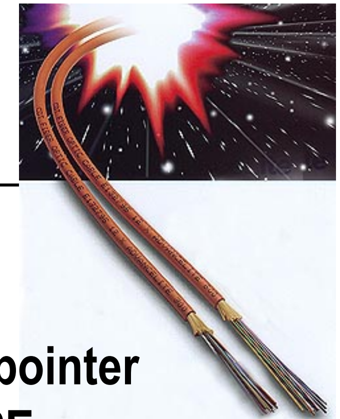
- **These octets (K1, K2) are used to send messages between two nodes when a failure is detected between them.**
- **Messages are sent both ways around the ring (if possible).**
- **Last four bits of K1 specifies the address of the addressed node.**
- **First four bits of K2 specifies the address of the sending node.**
- **Means that there can only be 16 nodes on a ring.**

Sync and Error Indication

A1	A2	J0
B1	E1	F1
D1	D2	D3
H1	H2	H3
B2	K1	K2
D4	D5	D6
D7	D8	D9
D10	D11	D12
S1	M0/1	E2

- S1 is used for synchronization status. xxxx 0001 indicates Stratum 1 clock.
- M0/M1 is used to send back to the sender, the error status of the received signal (determined by the Bx octets). Number of bit errors detected.

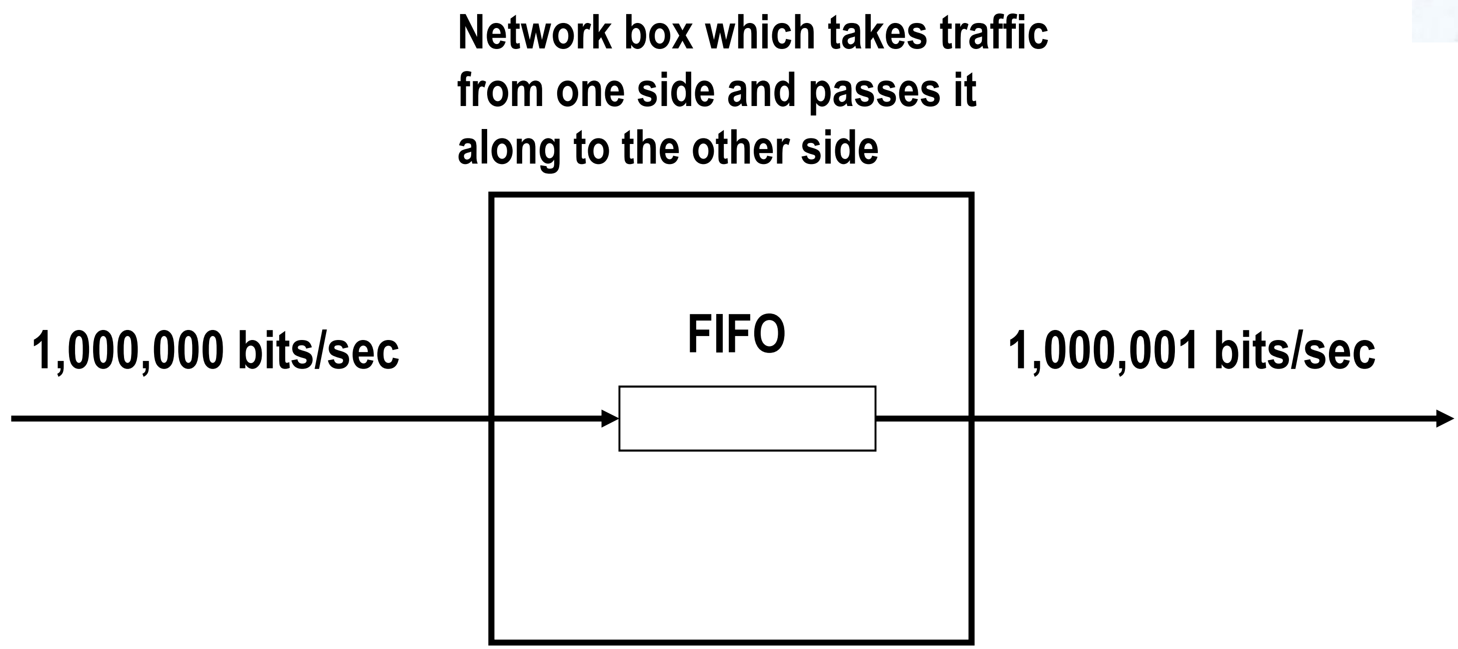
Payload Pointers



A1	A2	J0
B1	E1	F1
D1	D2	D3
H1	H2	H3
B2	K1	K2
D4	D5	D6
D7	D8	D9
D10	D11	D12
S1	M0/1	E2

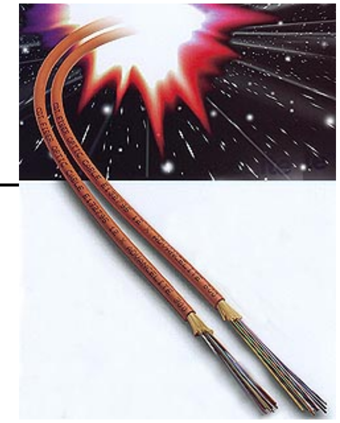
- **The H1, H2 octets are joined to form a pointer which points to the first octet of the SPE.**
- **The first 4 bits of the 16 bits indicates if the pointer is changing.**
- **The next two bits are not used.**
- **The last 10 bits are the pointer and can have a value from zero to 782.**
- **H3 is used to carry a payload octet when a negative pointer adjustment is done.**
- **These octets are covered in additional detail in the next slides.**

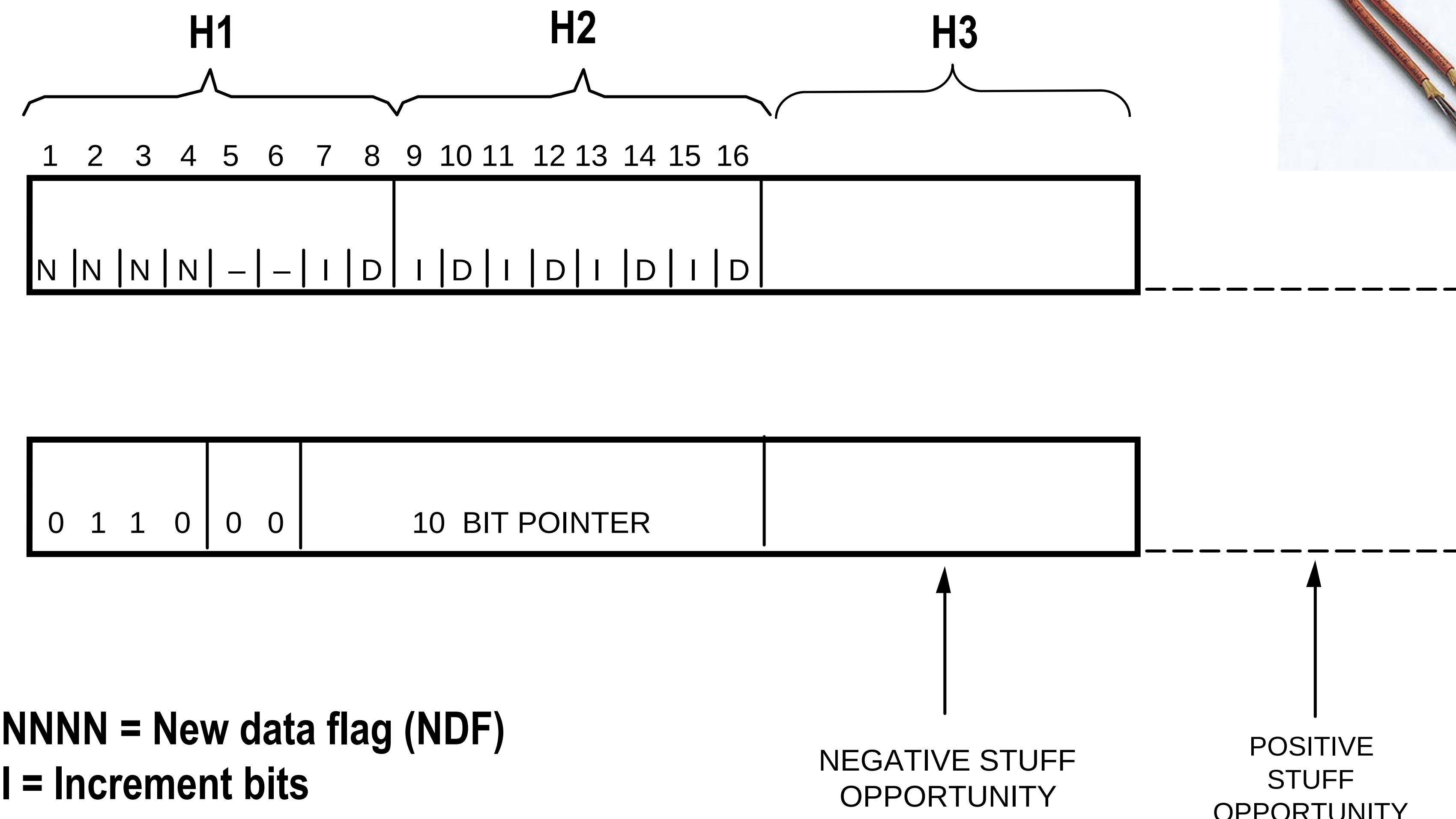
Why do we need Payload Pointers?



Every eight seconds, the FIFO will run dry. If an extra, "meaningless" octet can be sent at that time, it will give the FIFO time to fill up again. Problem: How to tell the receiver that this one octet is meaningless?

Format of the H1, H2, H3 Octets





NNNN = New data flag (NDF)
I = Increment bits
D = Decrement bits

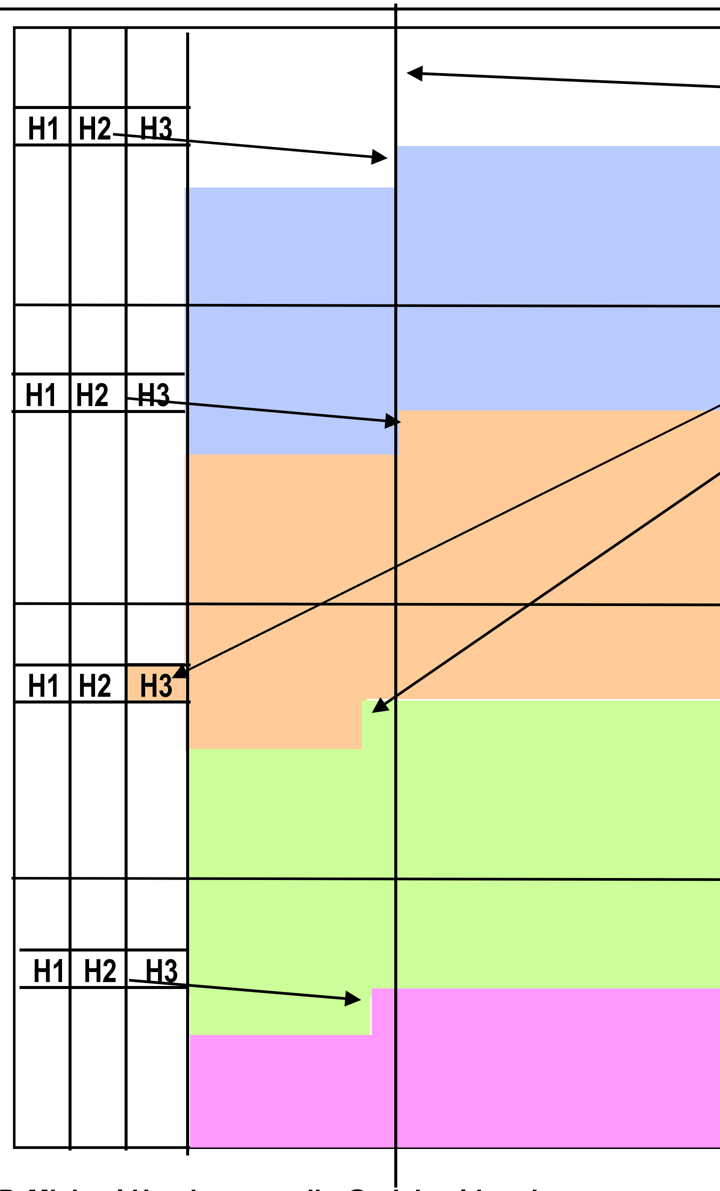
A Negative Pointer Adjustment

Normal Frame

Normal Frame

"D" bits inverted.

Pointer updated.



This line has no meaning – it's just there to help show the movement of the start of the frame (here).

Note that H3 is being used to carry a payload octet

The pointer is invalid here because of the "D" bit inversion. The value without inversion is the same as the previous frame.

Incoming payload clock is fast

Details of H1, H2, H3 octets during justification

Negative Justification using Inverted “D” Bits

Frame status	New Data Flag				Unused		I	D	I	D	I	D	I	D	I	D
Normal frame	0	1	1	0	x	x	0	0	0	1	1	1	1	1	1	0
Invert “D” Bits	0	1	1	0	x	x	0	1	0	0	1	0	1	0	1	1
New ptr value	0	1	1	0	x	x	0	0	0	1	1	1	1	1	0	1
New ptr value	0	1	1	0	x	x	0	0	0	1	1	1	1	1	0	1
Normal frame	0	1	1	0	x	x	0	0	0	1	1	1	1	1	0	1

Just FYI, the bits before negative justification are 126 decimal, and they go to 125 decimal.
Pointer adjustments can be made every fourth frame.

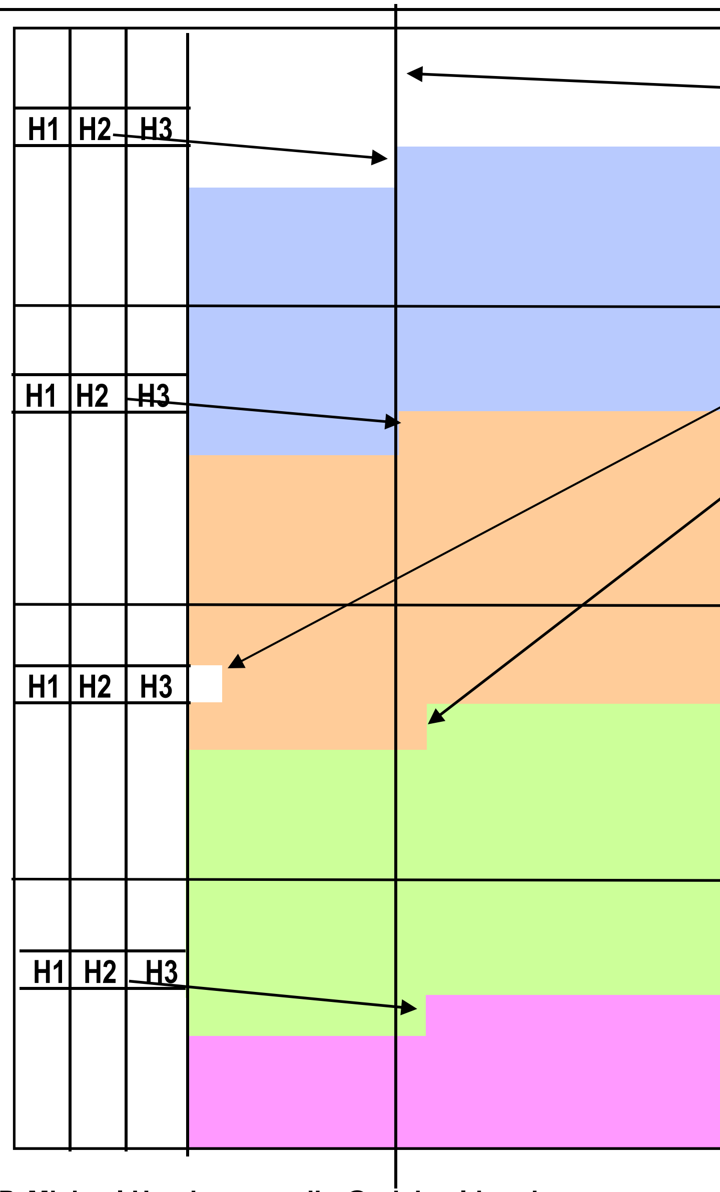
A Positive Pointer Adjustment

Normal Frame

Normal Frame

“I” bits inverted.

Pointer updated.



This line has no meaning – it’s just there to help show the movement of the start of the frame (here).

Positive stuff octet

The pointer is invalid here because of the “I” bit inversion. The value without inversion is the same as the previous frame.

Incoming payload clock is slow

Details of H1, H2, H3 octets during justification

Positive Justification using Inverted "I" Bits

Frame status	New Data Flag				Unused		I	D	I	D	I	D	I	D	I	D
Normal frame	0	1	1	0	X	X	0	0	0	1	1	1	1	1	1	0
Invert "I" Bits	0	1	1	0	X	X	1	0	1	1	0	1	0	1	0	0
New ptr value	0	1	1	0	X	X	0	0	0	1	1	1	1	1	1	1
New ptr value	0	1	1	0	X	X	0	0	0	1	1	1	1	1	1	1
Normal frame	0	1	1	0	X	X	0	0	0	1	1	1	1	1	1	1

Just FYI, the bits before positive justification are 126 decimal,
and they go to 127 decimal.
Pointer adjustments can be made every fourth frame.

Details of H1, H2, H3 octets when using NDF

One Octet Negative Adjustment using the NDF

Frame status	New Data Flag				Unused		I	D	I	D	I	D	I	D	I	D
Normal frame	0	1	1	0	X	X	0	0	0	1	1	1	1	1	1	0
NDF indicator	1	0	0	1	X	X	0	0	0	1	1	1	1	1	0	1
New ptr value	0	1	1	0	X	X	0	0	0	1	1	1	1	1	0	1
New ptr value	0	1	1	0	X	X	0	0	0	1	1	1	1	1	0	1
Normal frame	0	1	1	0	X	X	0	0	0	1	1	1	1	1	0	1

The NDF is normally used if a large change has to be made to the pointer.
Just FYI, the bits before negative justification are 126 decimal,
and they go to 125 decimal.
Pointer adjustments can be made every fourth frame.

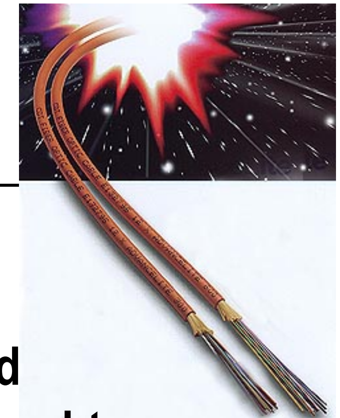
Details of H1, H2, H3 octets when using NDF

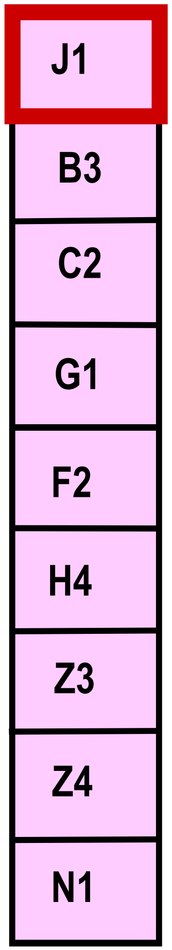
One Octet Positive Adjustment using the NDF

Frame status	New Data Flag				Unused		I	D	I	D	I	D	I	D	I	D
Normal frame	0	1	1	0	x	x	0	0	0	1	1	1	1	1	1	0
NDF indicator	1	0	0	1	x	x	0	0	0	1	1	1	1	1	1	1
New ptr value	0	1	1	0	x	x	0	0	0	1	1	1	1	1	1	1
New ptr value	0	1	1	0	x	x	0	0	0	1	1	1	1	1	1	1
Normal frame	0	1	1	0	x	x	0	0	0	1	1	1	1	1	1	1

The NDF is normally used if a large change has to be made to the pointer.
Just FYI, the bits before positive justification are 126 decimal,
and they go to 127 decimal.
Pointer adjustments can be made every fourth frame.

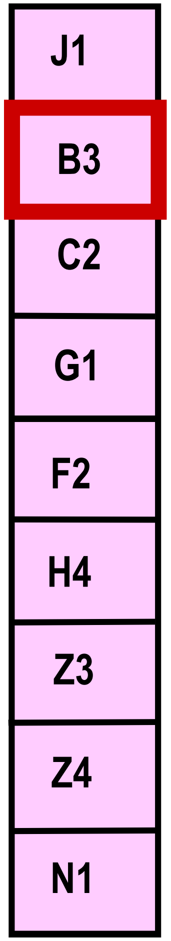
Path Overhead





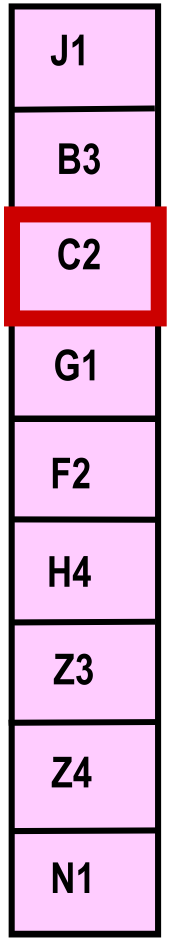
- **Path trace (J1) used to check continued connection between the path devices (end-to-end). A 16 or 64 octet message is sent, one octet per frame.**
- **64 octet for SONET and 16 octet for SDH message.**

Error Checking



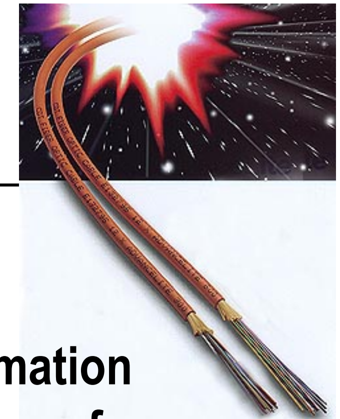
- BIP-8 parity check over the payload, only.
- Even parity.
- Parity is for *previous* payload, not this one.

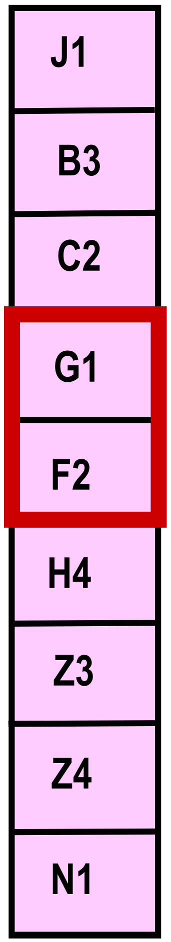
Path Signal Label



- Indicates the type of payload carried in the SPE.
- For our discussion, we'll be interested in a value of 0x02 for floating virtual tributary (VT) mode, 0x04 for asynchronous mapping of DS-3, 0x13 for mapping of ATM, 0x16 for packet over SONET (POS), and 0x1b for generic framing procedure (GFP).

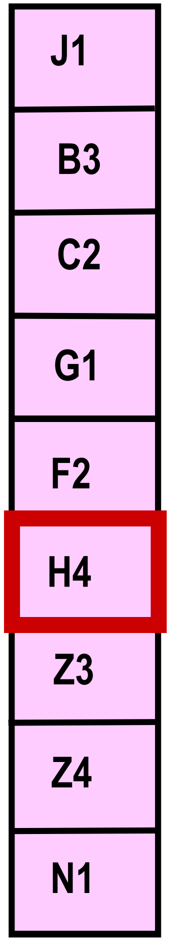
Path Status and Path User Channel





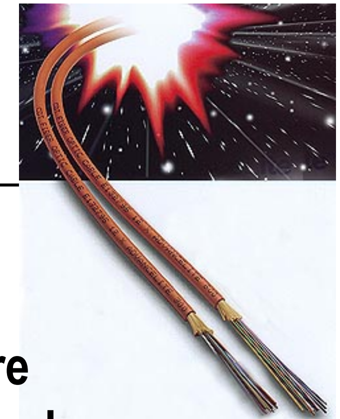
- **The G1 octet (Path Status) sends information back to the sender indicating the number of parity errors, or if a complete failure is detected.**
- **The F2 octet is a user channel for Path applications (end-to-end). It is not subject to standardization (anything can be put into it).**

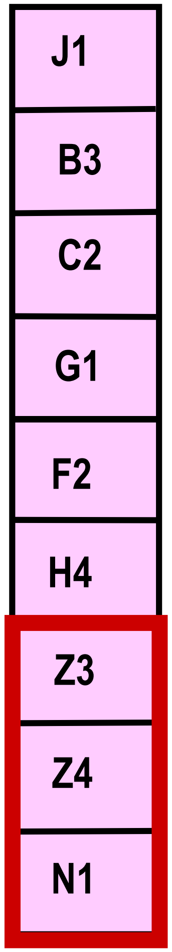
Multi-Frame Indicator



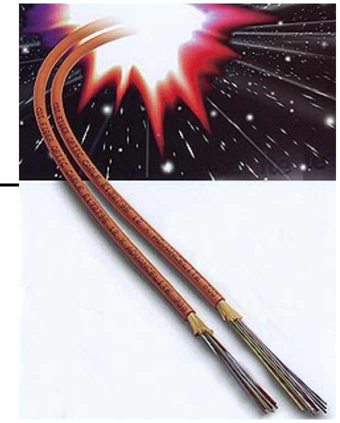
- The last two bits of this octet counts from 00 to 11 continuously to provide a multi-frame indicator for VT payloads (to be explained later).
- Some of the other bits are being defined for use with virtual concatenation but will not be described here.

TCM and Reserved octets



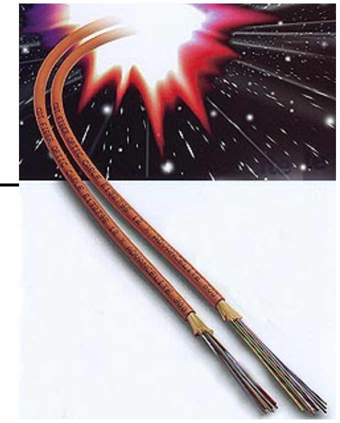


- The Z3, Z4 octets are reserved for future standardization and have no meaning today.
- The N1 octet is for tandem connection monitoring and is fairly complex. No attempt will be made here to explain it.

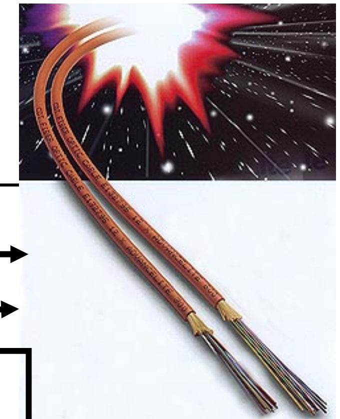


Virtual Tributaries

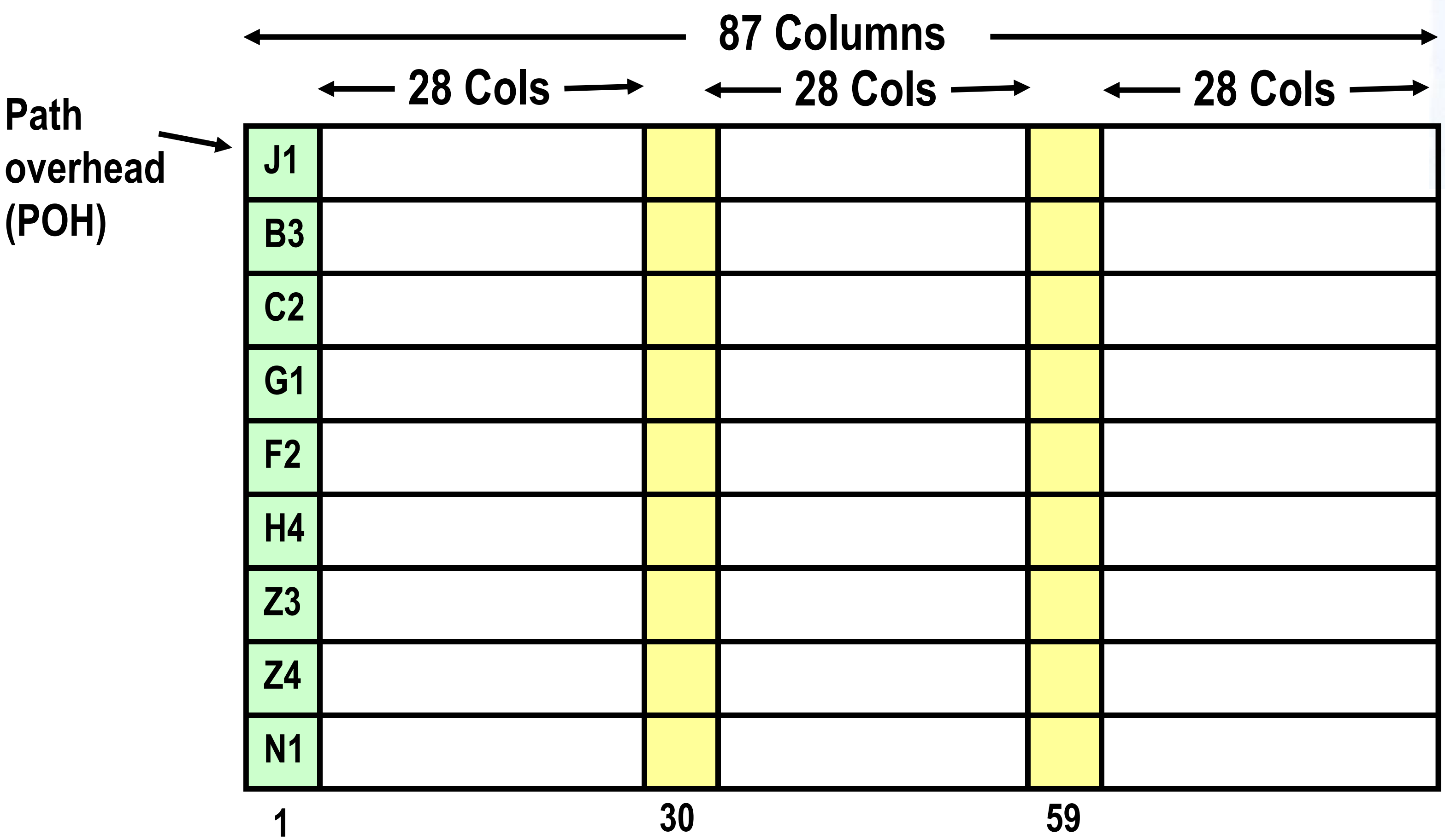
Plesiochronous Data Rates of Interest



Type of Digital Circuit	Bit Rate (Mbps)
DS-1 (T1)	1.544
E1	2.048
DS-1C	3.152
DS-2	6.312
DS-3 (T3)	44.736

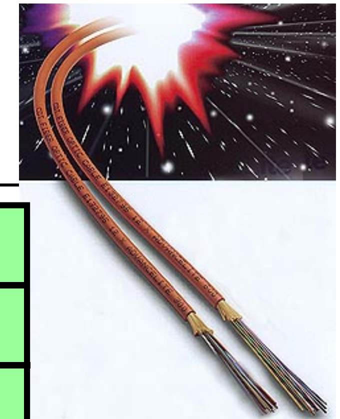


STS-1 Payload Mapping



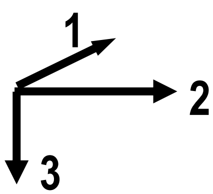
All mappings of payload into an STS-1 payload have these two columns taken as stuff columns, leaving 84 columns for data.

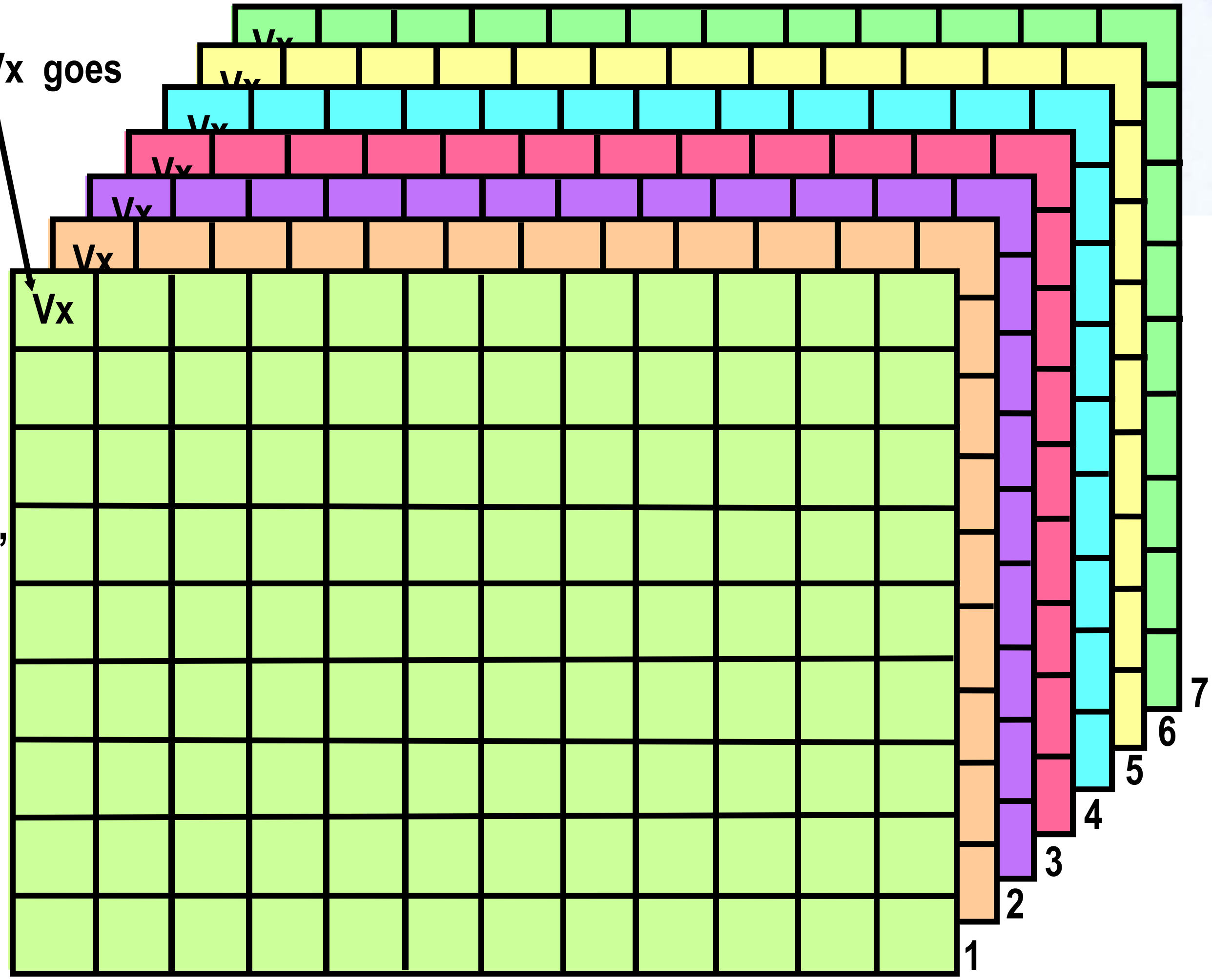
Virtual Tributary Groups



The "x" in Vx goes from 1 to 4

Order of transmission





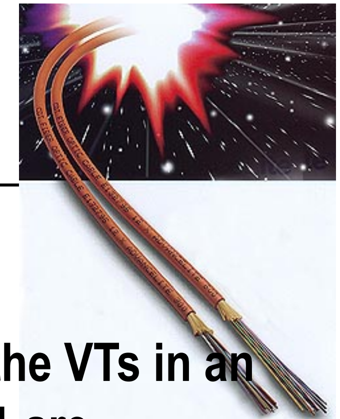
When interleaving, the first column of VTG#1 goes next to the POH. Columns 30 & 59 of the SPE are skipped.

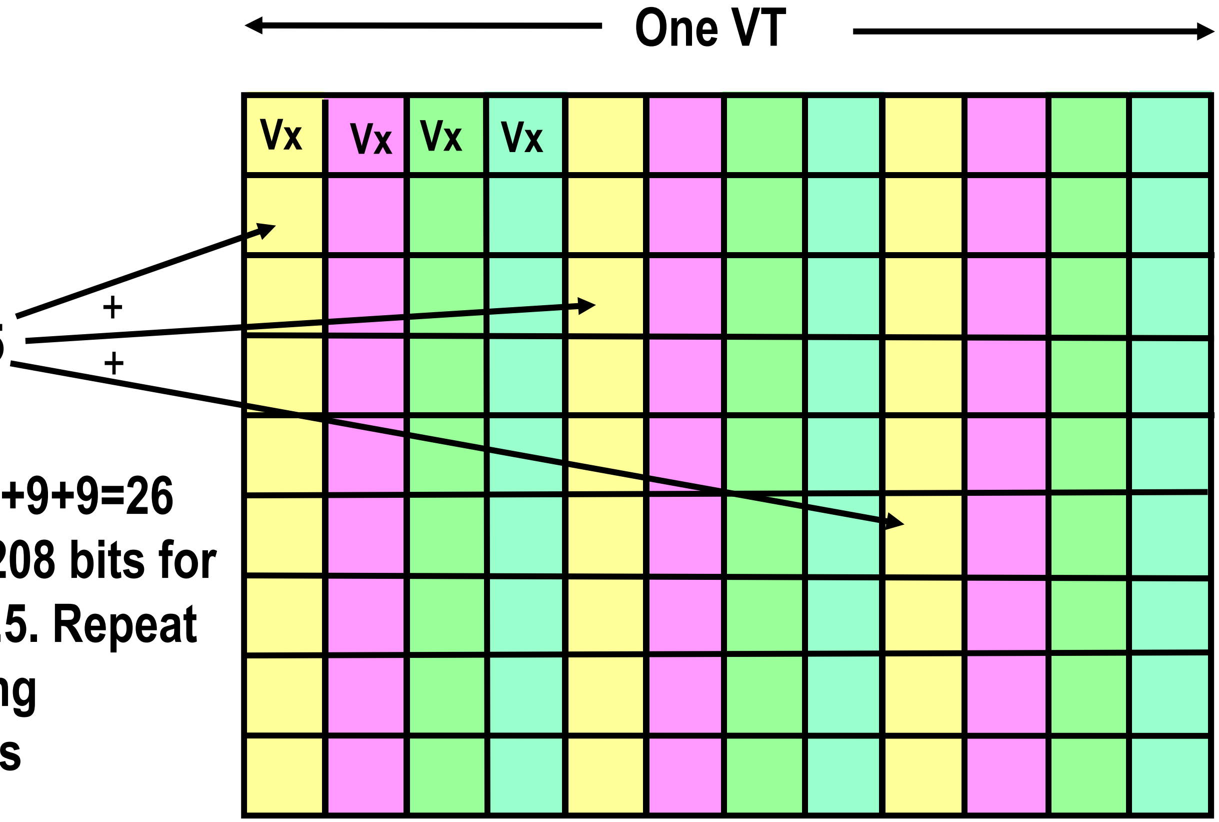
The 84 usable payload columns are divided into seven groups of twelve columns. Each set of twelve is called a "Virtual Tributary Group" (VTG)

Virtual Tributaries

- **A Virtual Tributary Group is further subdivided into Virtual Tributaries (VT).**
  - Three columns makes a VT-1.5 (1.728 Mbps gross, good for a DS-1 at 1.544 Mbps).
  - Four columns makes a VT-2 (2.304 Mbps gross, good for an E1 at 2.048 Mbps).
  - Six columns makes a VT-3 (3.456 Mbps gross, good for a DS-1C at 3.152 Mbps).
  - Twelve columns makes a VT-6 (6.912 Mbps gross, good for a DS-2 at 6.312 Mbps).
- **A VT contains four VT-1.5s, or three VT-2s, or two VT-3s, or one VT-6.**
- **A VTG can only contain one kind of VT. Different VTGs in an SPE can contain different VT types, but within a VTG, there can only be one kind of VT.**

Virtual Tributaries – The VT-1.5





One VT-1.5

We have 8+9+9=26 octets or 208 bits for each VT-1.5. Repeat 8KHz giving 1.664 Mbps

If all the VTs in an STS-1 are carrying VT-1.5s, the capacity of the STS-1 SPE is 28 DS-1s, or the same as a DS-3. (Seven VTs times four VT-1.5s per VT)

The four VT-1.5s are interleaved into the VT. Note the colors – there are three of each color, with each color indicating one VT-1.5 (for four inside the VT. Each VT is 27 octets, with one octet taken for the Vx octet.

Superframes

- **Remember that the last two bits of H4 in the POH count 00, 01, 10, 11, 00, etc. This produces a superframe of four frames.**
- **The frame after the SPE with an H4 value of 00 will have the first octet in the VT identified as V1. The one after value 01 will be identified as V2, etc. It takes four SONET frames to build the pointer for each VT so the receiver has to buffer frames.**
- **V1 and V2 form a pointer, exactly like the H1, H2 octets.**
  - Bits 5 and 6 are used to indicate the type of VT (unused in H1, H2) (VT-6=00, VT-3=01, VT-2=10, VT-1.5=11)
- **V3 is the negative stuff opportunity, exactly like H3.**
- **V4 is reserved for future standardization.**

V1	V2	V3	V4

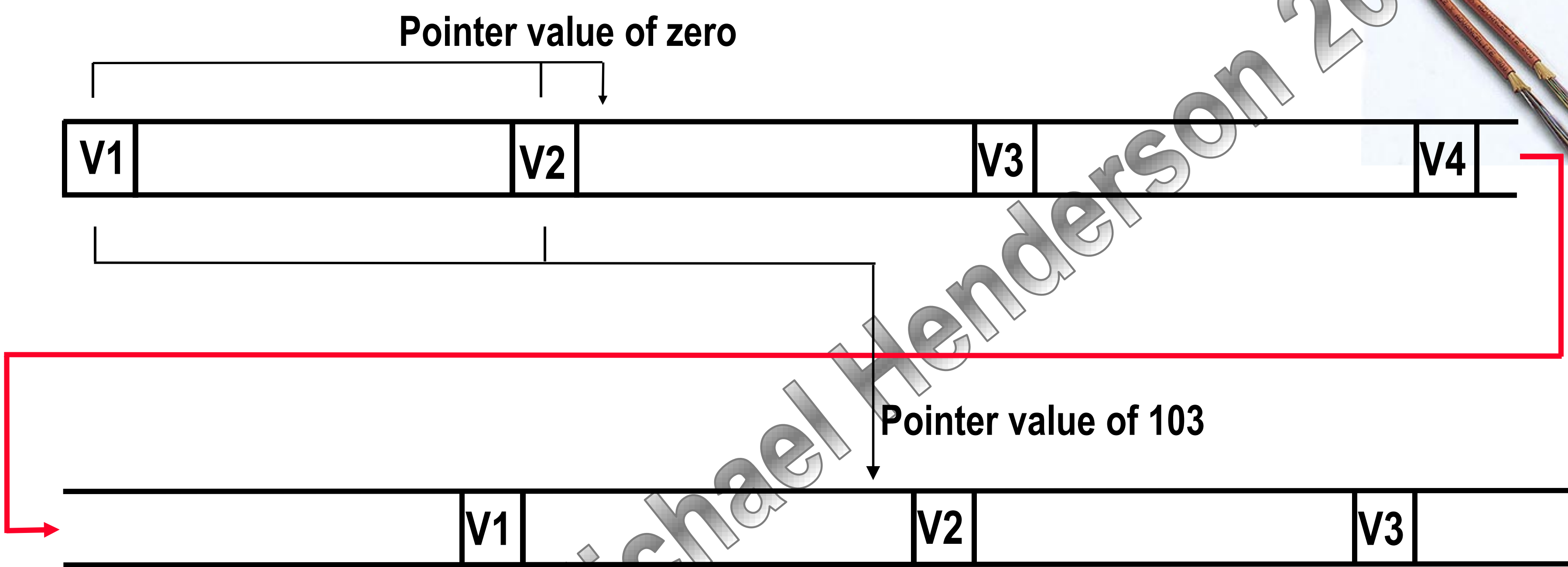
The V1, V2 Pointers

	V1								V2							
	N	N	N	N	S	S	I	D	I	D	I	D	I	D	I	D
VT6	0	1	1	0	0	0	10 BIT POINTER									
VT3	0	1	1	0	0	1	10 BIT POINTER									
VT2	0	1	1	0	1	0	10 BIT POINTER									
VT1.5	0	1	1	0	1	1	10 BIT POINTER									

NEW DATA FLAG – INVERT 4 N BITS
NEGATIVE STUFF – INVERT 5 D BITS
POSITIVE STUFF – INVERT 5 I BITS

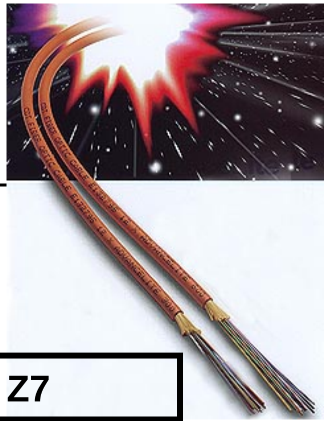
I – Increment Bit
D – Decrement Bit
N – New Data Flag BIt
S - VT Size Bit

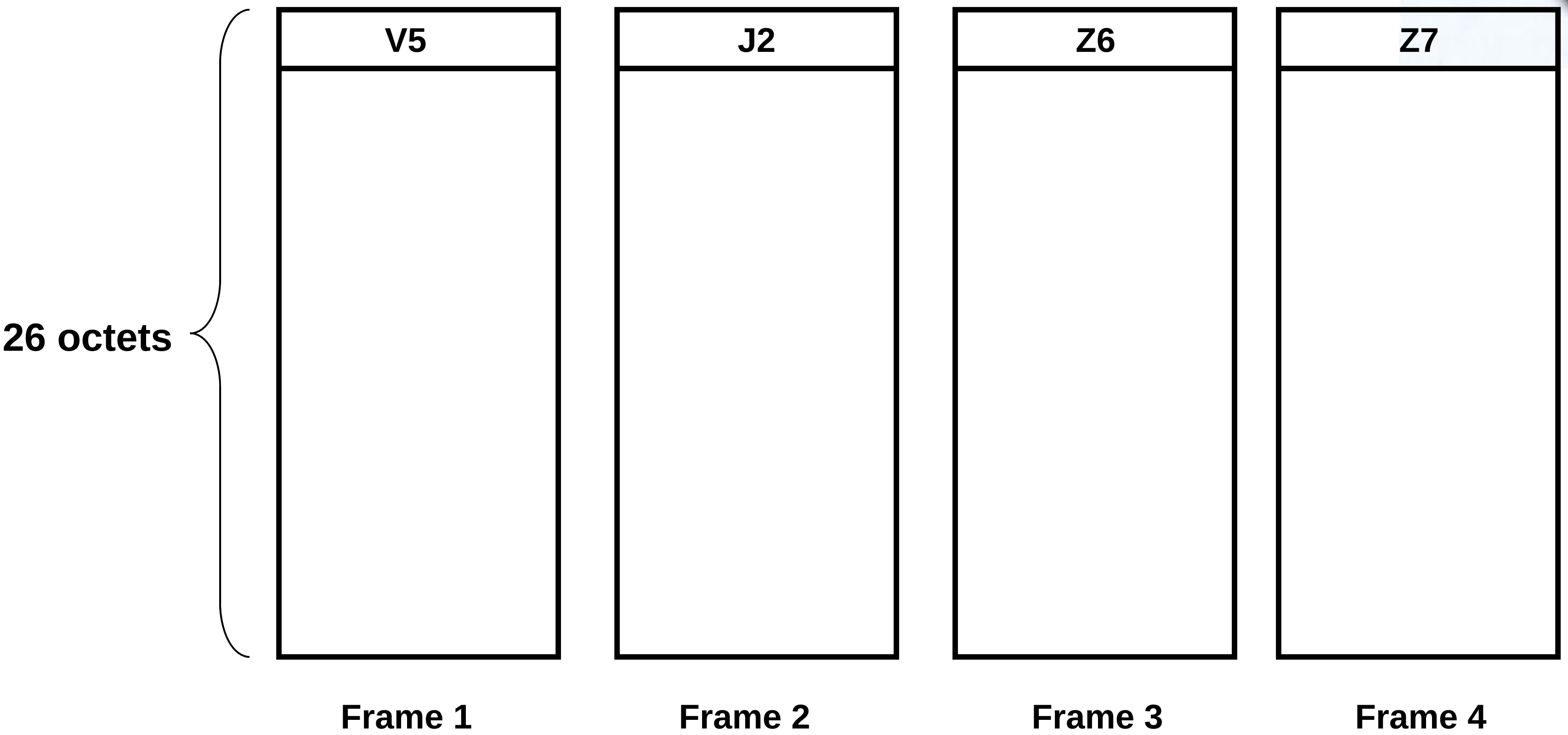
Pointing to the Start of the VT Payload



A VT-1.5 is made up of three columns of 9 octets or 27 octets. Four frames make a superframe, or 108 octets. Once the V1, V2, V3, V4 octets are removed from the count, we have 104 octets remaining. Since the pointer counts from zero, the highest value of the pointer is 103.

The VT-1.5 Payload



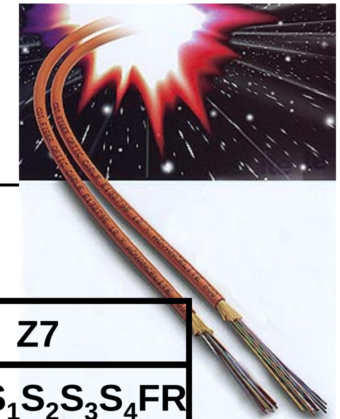


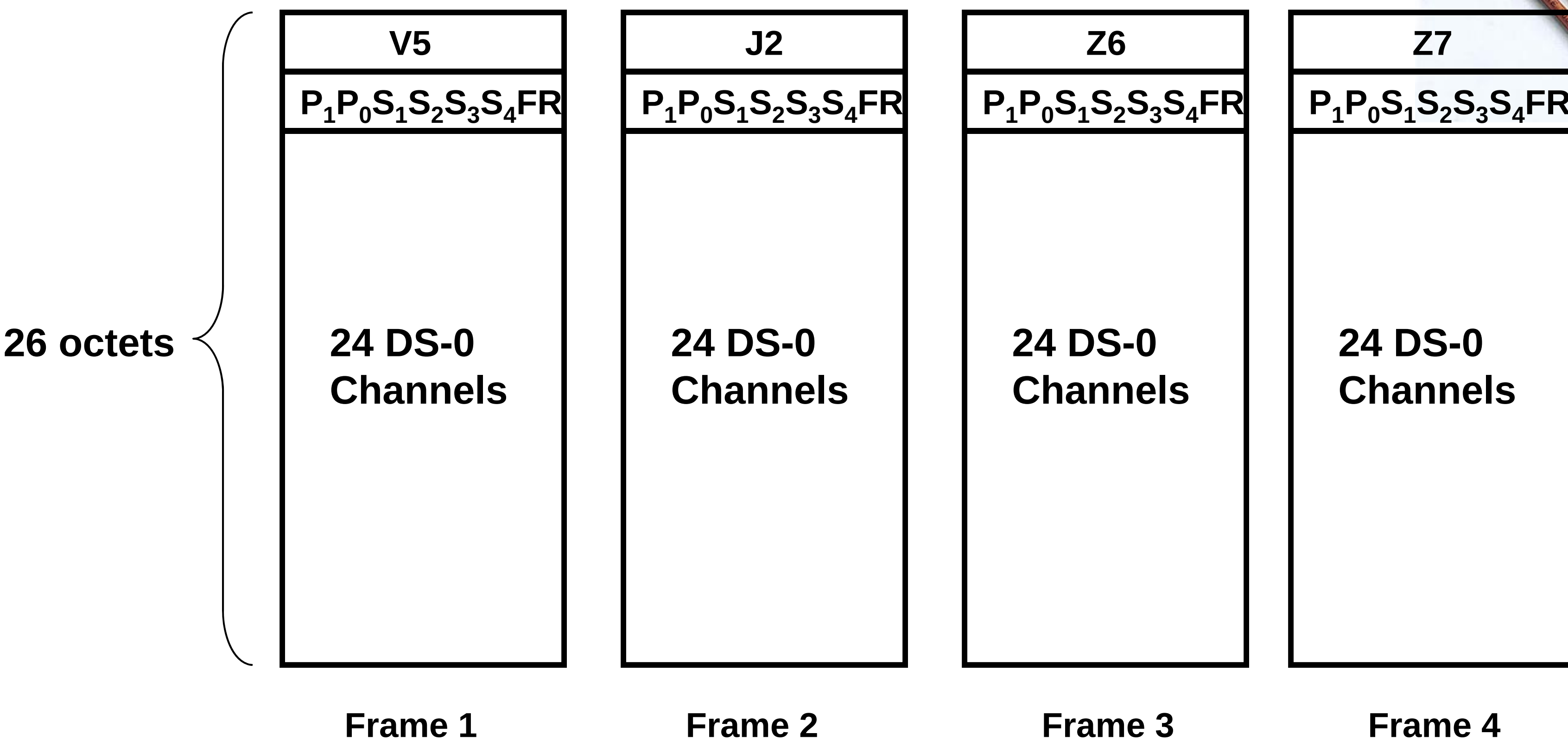
We started with 27 octets per frame but took one octet for pointers. Now we're taking another octet per frame. This leaves 25 octets per frame. 25 octets times 8 bits per octet, times 8,000 frames per second gives 1.6Mbps.

Byte Synchronous and Asynchronous VTs

- **We're going to look at two ways that DS-1 traffic can be carried in a VT payload – byte synchronous and asynchronous.**
- **Byte synchronous preserves the location of the payload octets in a T1 frame (each speech sample).**
  – Used primarily to transport channelized T1s which are carrying voice calls.
- **Asynchronous simply transports the 1.544 Mbps stream without concern for which byte is which.**
  – Used to carry T1s which are carrying data.
- **Bits 5, 6, & 7 of V5 indicate which is being carried (010 = asynchronous, 100 = byte synchronous).**

Byte Synchronous Mapping – VT-1.5





R = Fixed stuff bit

F = DS-1 framing bit

Sx = Signaling bits, A, B, C, D bits in a DS-1

Px = Phase bits, indicates which frame set for SF or ESF framing

Asynchronous Mapping – VT-1.5

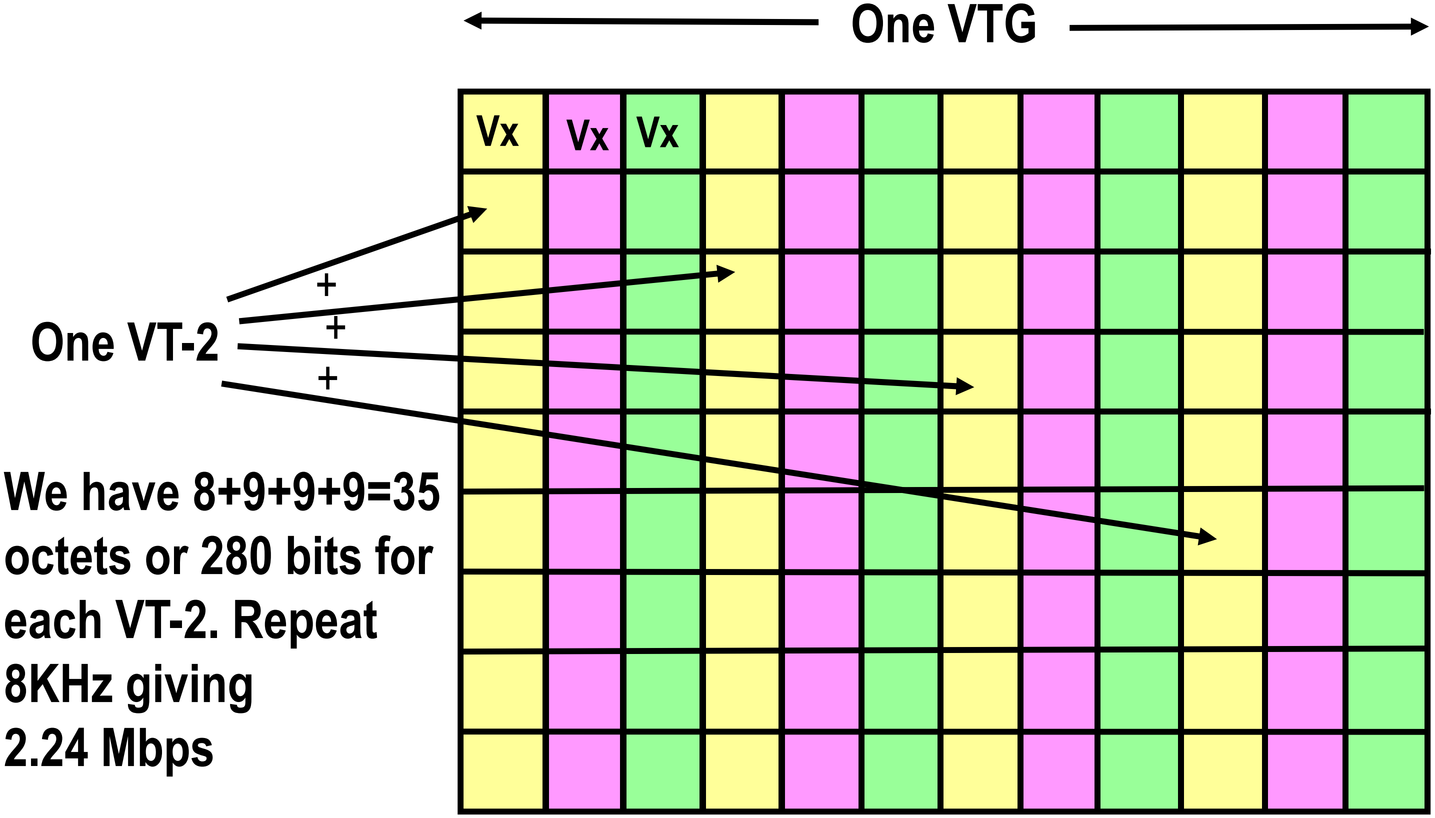
26 octets

Frame 1	Frame 2	Frame 3	Frame 4
V5	J2	Z6	Z7
R R R R R R I R	C_1C_2 O O O O I R	C_1C_2 O O O O I R	C_1C_2 R R R S_1S_2 R
24 Octets (192 bits)	24 Octets (192 bits)	24 Octets (192 bits)	24 Octets 192 bits)

R = Stuff bits – no meaning
O = Future standardization – unused
I = Information bit. Makes the 193rd bit of a DS-1 frame

Cx = Controls use of the S bits.
Sx = Stuff bits. May or may not carry information bits, depending upon values in Cx

Virtual Tributaries – The VT-2



We have 8+9+9+9=35 octets or 280 bits for each VT-2. Repeat 8KHz giving 2.24 Mbps

The three VT-2s are interleaved into the VT. Note the colors – there are four of each color, with each color indicating one VT-2 (for three inside the VT. Each VT is 36 octets, with one octet taken for the Vx octet.

Superframes

- **Remember that the last two bits of H4 in the POH count 00, 01, 10, 11, 00, etc. This produces a superframe of four frames.**
- **The frame after the SPE with an H4 value of 00 will have the first octet in the VT identified as V1. The one after value 01 will be identified as V2, etc. . It takes four SONET frames to build the pointer for each VT so the receiver has to buffer frames.**
- **V1 and V2 form a pointer, exactly like the H1, H2 octets.**
  - Bits 5 and 6 are used to indicate the type of VT (unused in H1, H2) (VT-6=00, VT-3=01, VT-2=10, VT-1.5=11)
- **V3 is the negative stuff opportunity, exactly like H3.**
- **V4 is reserved for future standardization.**

V1	V2	V3	V4

The V1, V2 Pointers

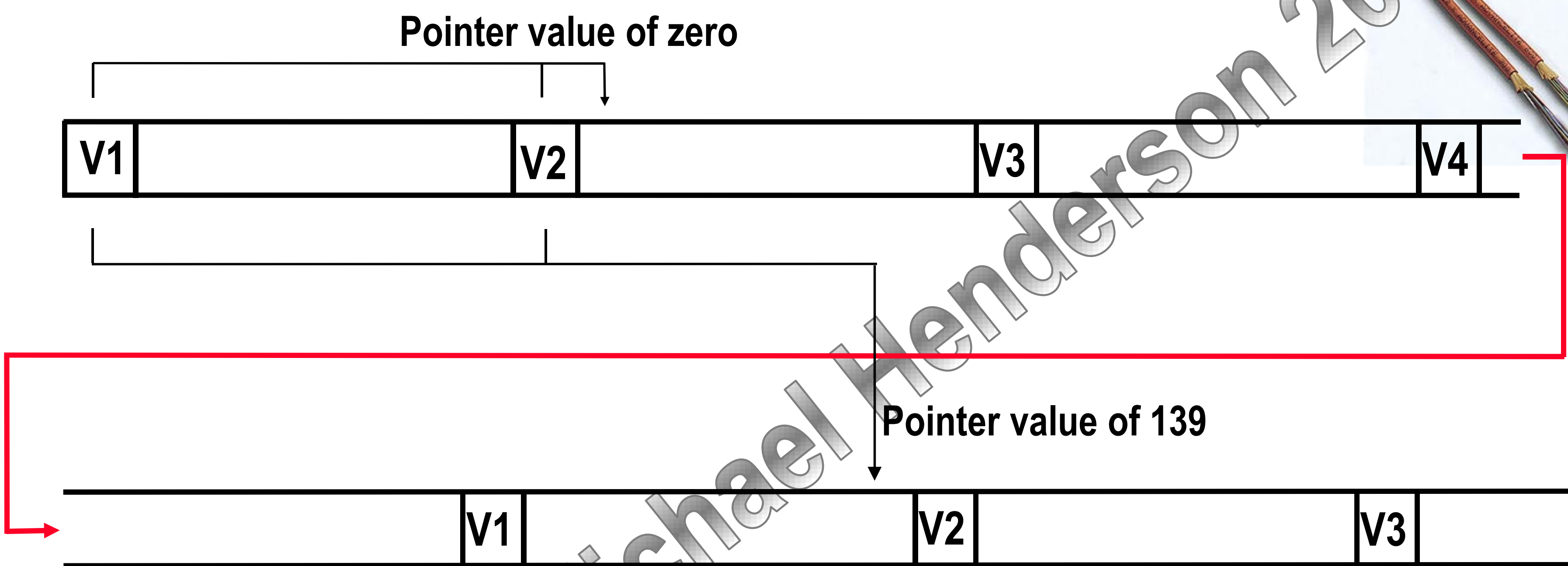
	V1							V2							
	N	N	N	N	S	S	I	D	I	D	I	D	I	D	I	D

VT6	0	1	1	0	0	0	10 BIT POINTER
VT3	0	1	1	0	0	1	10 BIT POINTER
VT2	0	1	1	0	1	0	10 BIT POINTER
VT1.5	0	1	1	0	1	1	10 BIT POINTER

NEW DATA FLAG – INVERT 4 N BITS
NEGATIVE STUFF – INVERT 5 D BITS
POSITIVE STUFF – INVERT 5 I BITS

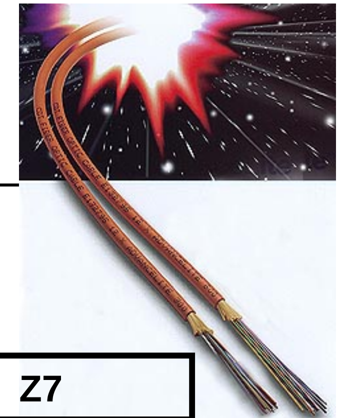
I – Increment Bit
D – Decrement Bit
N – New Data Flag BIt
S - VT Size Bit

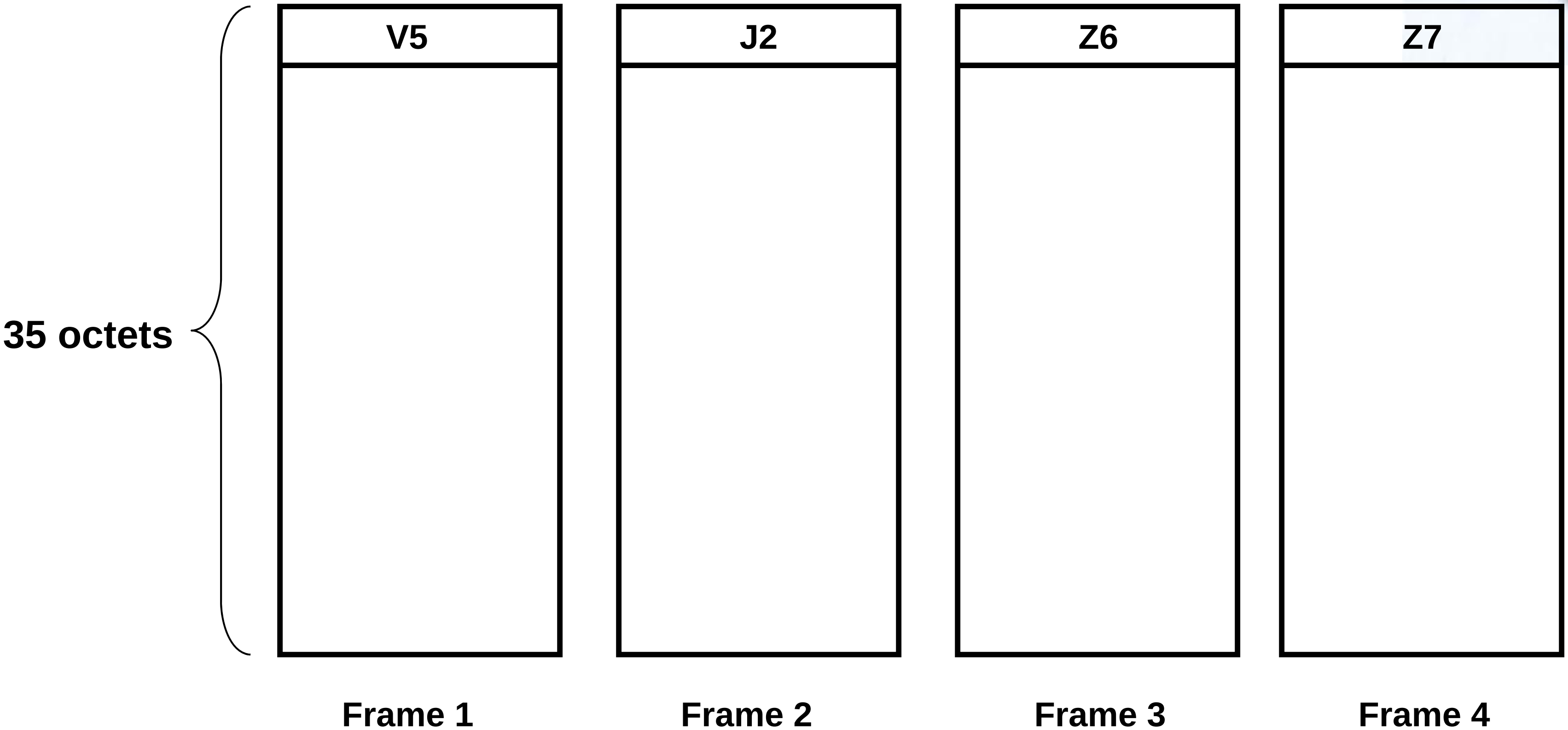
Pointing to the Start of the VT Payload



A VT-2 is made up of four columns of 9 octets or 36 octets. Four frames make a supreframe, or 144 octets. Once the V1, V2, V3, V4 octets are removed from the count, we have 140 octets remaining. Since the pointer counts from zero, the highest value of the pointer is 139.

The VT-2 Payload



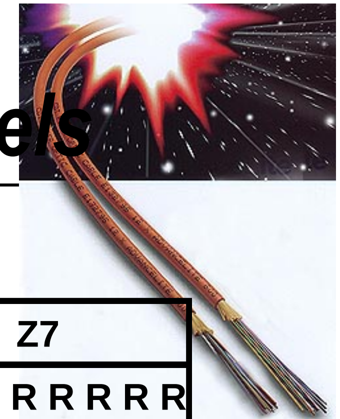


We started with 36 octets per frame but took one octet for pointers. Now we're taking another octet per frame. This leaves 34 octets per frame. 34 octets times 8 bits per octet, times 8,000 frames per second gives 2.176Mbps.

Byte Synchronous and Asynchronous VTs

- **We're going to look at two ways that E-1 traffic can be carried in a VT payload – byte synchronous and asynchronous.**
- **Byte synchronous preserves the location of the payload octets in an E1 frame (each speech sample).**
  - Used primarily to transport channelized E1s which are carrying voice calls.
- **Asynchronous simply transports the 2.048 Mbps stream without concern for which byte is which.**
  - Used to carry E1s which are carrying data.
- **Bits 5, 6, & 7 of V5 indicate which is being carried (010 = asynchronous, 100 = byte synchronous).**

Byte Synchronous Mapping – 30 Channels



35 octets

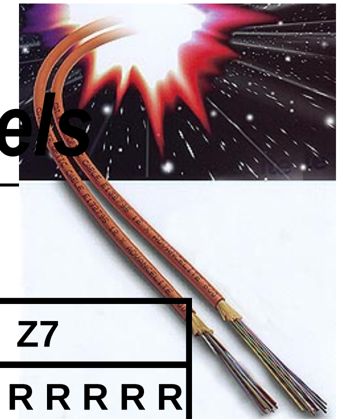
V5	J2	Z6	Z7
P_1P_0R R R R R R	P_1P_0R R R R R R	P_1P_0R R R R R R	P_1P_0R R R R R R
R R R R R R R R	R R R R R R R R	R R R R R R R R	R R R R R R R R
Channels 1 - 15	Channels 1 - 15	Channels 1 - 15	Channels 1 - 15
CAS	CAS	CAS	CAS
Channels 16 - 30	Channels 16 - 30	Channels 16 - 30	Channels 16 - 30
R R R R R R R R	R R R R R R R R	R R R R R R R R	R R R R R R R R
Frame 1	Frame 2	Frame 3	Frame 4

R = Fixed stuff bit

R = may be used for channel 0, if required

Px = Phase bits, indicates which channel the CAS bits apply to

Byte Synchronous Mapping – 31 Channels



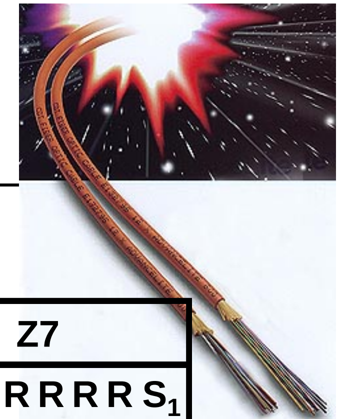
35 octets

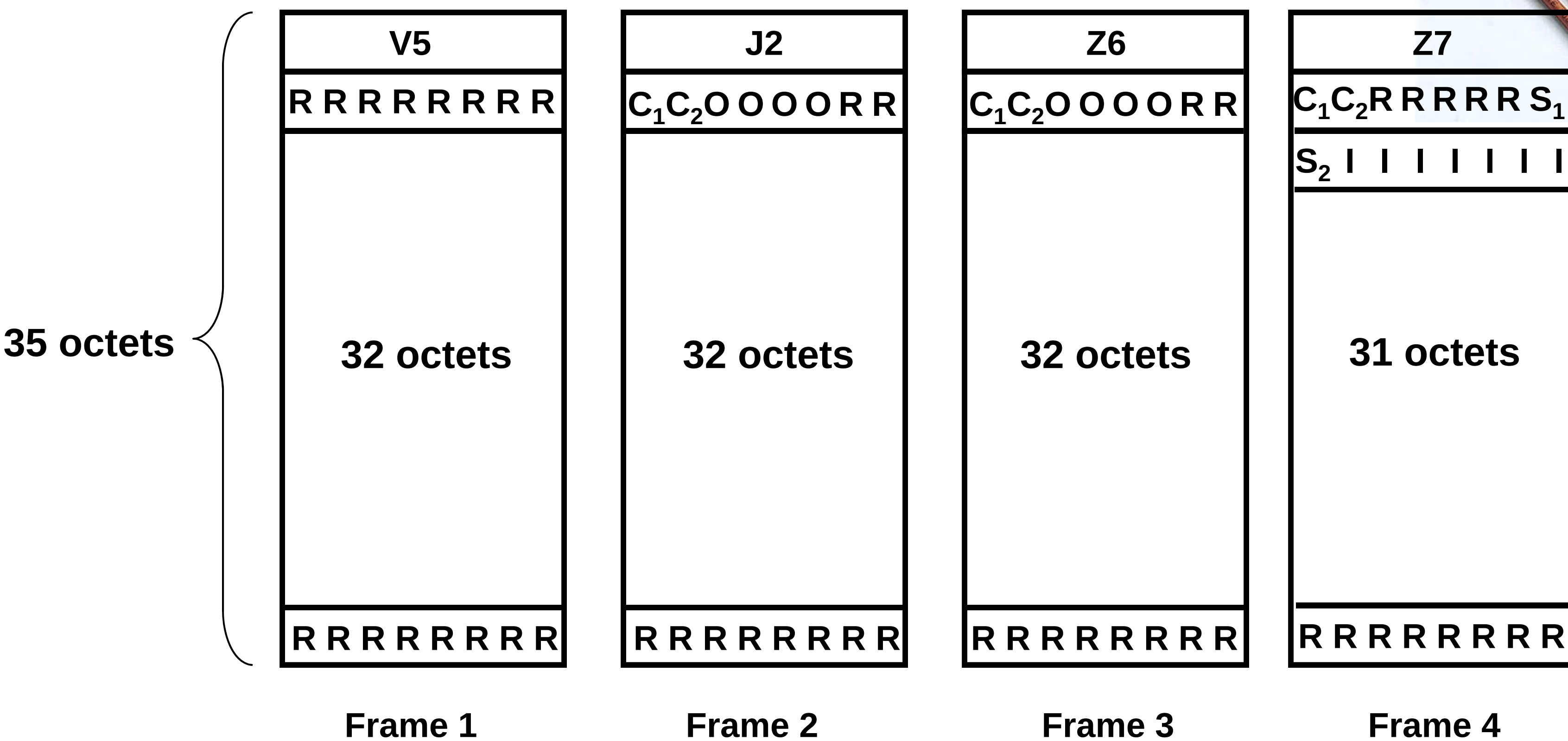
Frame 1	Frame 2	Frame 3	Frame 4
V5	J2	Z6	Z7
R R R R R R R R	R R R R R R R R	R R R R R R R R	R R R R R R R R
R R R R R R R R	R R R R R R R R	R R R R R R R R	R R R R R R R R
Channels 1 - 15	Channels 1 - 15	Channels 1 - 15	Channels 1 - 15
Channel 16	Channel 16	Channel 16	Channel 16
Channels 16 - 30	Channels 16 - 30	Channels 16 - 30	Channels 16 - 30
R R R R R R R R	R R R R R R R R	R R R R R R R R	R R R R R R R R

R = Fixed stuff bit

R = Indicates may be used for channel 0

Asynchronous Mapping – VT-2

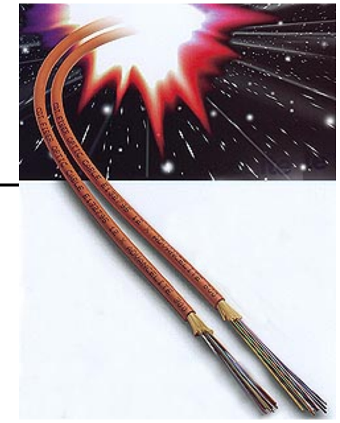




I = Information bit (Data)
R = Fixed stuff bit
O = Overhead – same as R

Cx = Stuff control
Sx = Stuff opportunity

32*3+31=127 octets.
127 octets * 8 bits=1,016 bits
1016+7=1023 1,023*2,000 times/second = 2.046Mbps
1016+9=1025 -> 2.050Mbps



Finished with Virtual Tributaries!!!

Support for DS-3 signals

- **A DS-3 signal at 44.736 Mbps takes the whole STS-1 SPE.**
  - No virtual tributaries are used.
- **A DS-3 can only be carried with asynchronous mapping.**
  - There is no byte synchronous mapping for a DS-3.

Support for DS-3 signals

	28 Octets					28 Octets					28 Octets			
STS Path Overhead	R	R	C1	25 I	Fixed Stuff	R	C2	I	25 I	Fixed Stuff	R	C3	I	25 I
	R	R	C1	25 I		R	C2	I	25 I		R	C3	I	25 I
	R	R	C1	25 I		R	C2	I	25 I		R	C3	I	25 I
	R	R	C1	25 I		R	C2	I	25 I		R	C3	I	25 I
	R	R	C1	25 I		R	C2	I	25 I		R	C3	I	25 I
	R	R	C1	25 I		R	C2	I	25 I		R	C3	I	25 I
	R	R	C1	25 I		R	C2	I	25 I		R	C3	I	25 I
	R	R	C1	25 I		R	C2	I	25 I		R	C3	I	25 I
	R	R	C1	25 I		R	C2	I	25 I		R	C3	I	25 I

Octets:

		8 bits
I	=	i i i i i i i i
R	=	r r r r r r r r
C1	=	r r c i i i i i
C2	=	c c r r r r r r
C3	=	c c r r o o r s

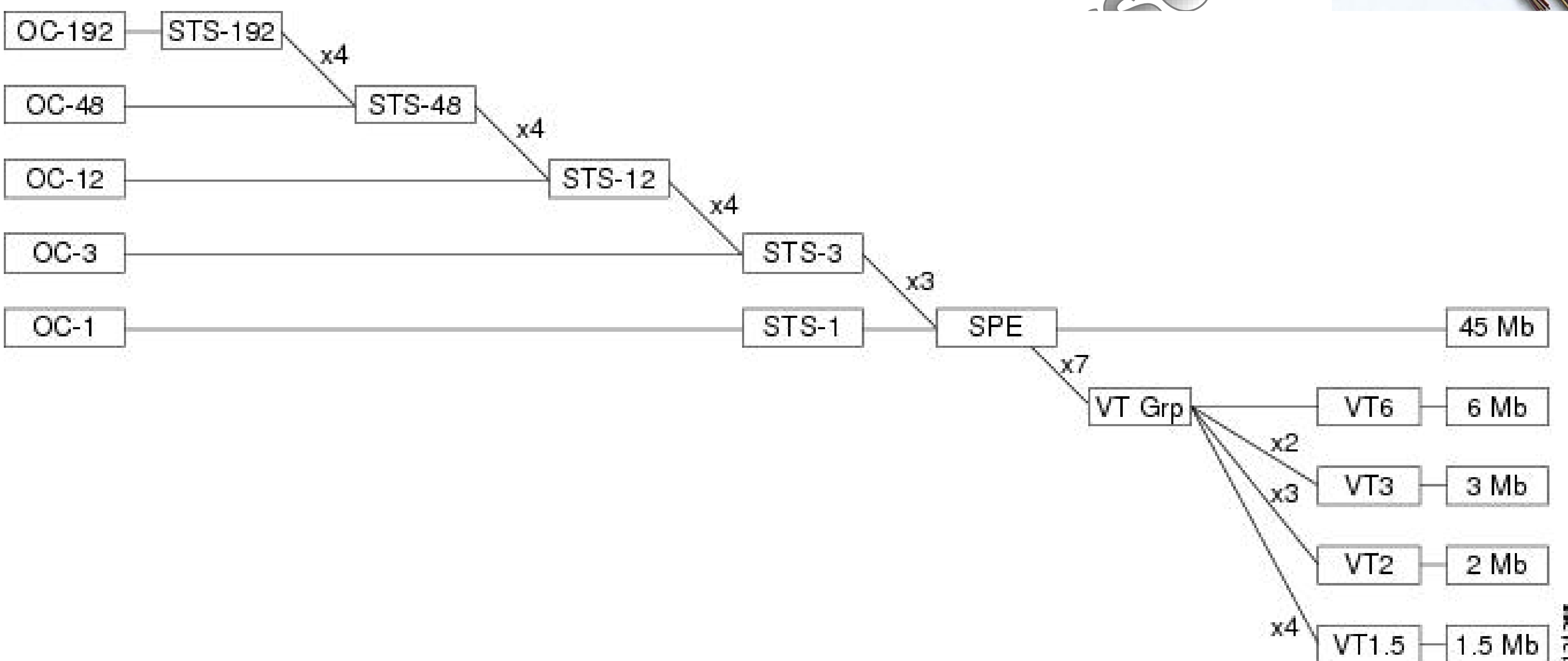
Bits:

i = information bit
r = fixed stuff bit
c = justification control bit
s = justification opportunity bit
o = overhead bit

Support for DS-3 signals

- **There are 77 full information payload octets, or 616 information bits.**
- **The C1 octet has 5 information bits, making the total number of information bits per row 621.**
- **Nine of these per frame, times 8,000 frames per second gives 44.712 Mbps.**
  – So how do we carry a 44.736 Mbps signal in 44.712 Mbps?
- **The answer is in the “s” bit in C3. Sometimes it carries an information bit.**
  – If it carried an information bit every frame, the data rate would be 44.784 Mbps.

SONET Multiplexing Hierarchy



SDH Multiplexing Hierarchy

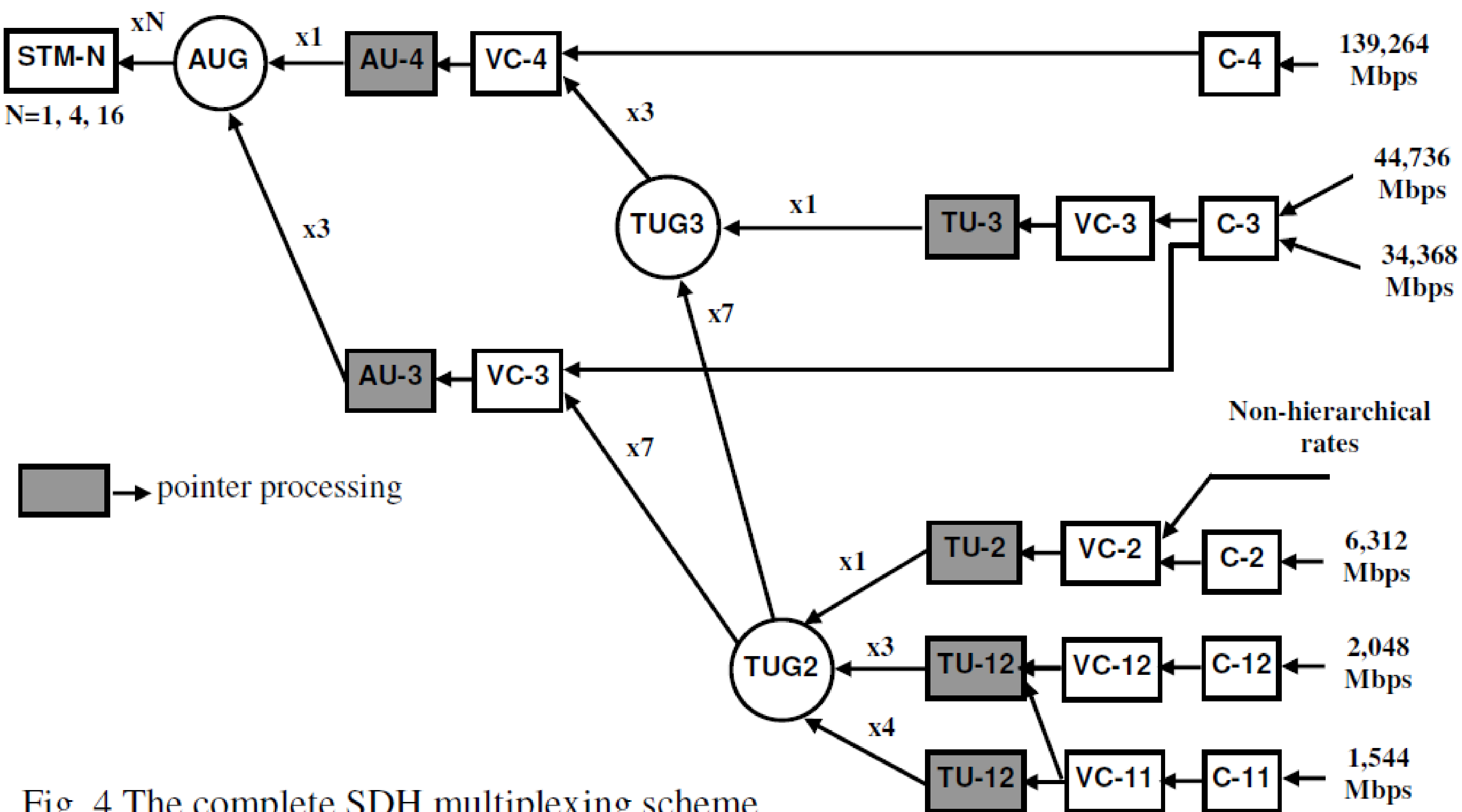


Fig. 4 The complete SDH multiplexing scheme

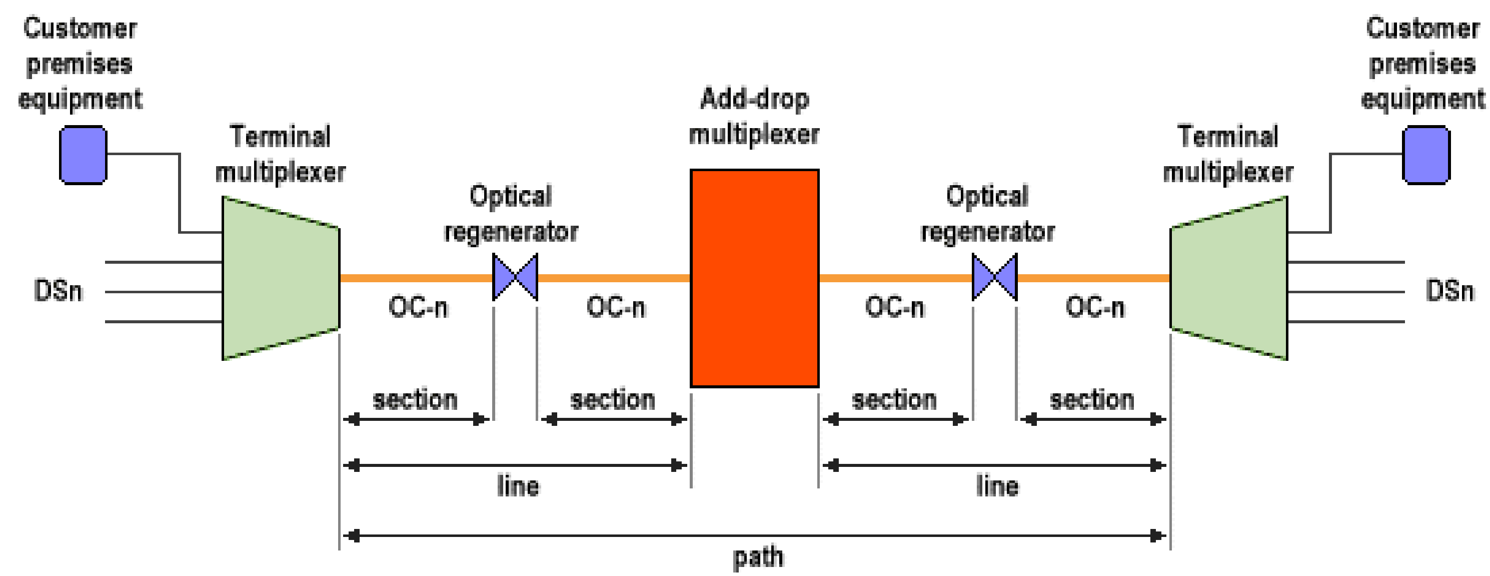
SONET/SDH Networks

SONET/SDH Networks

- SONET/SDH was designed to operate as either a linear connection, or a ring.
- Each node on a SONET/SDH network is known as an Add/Drop Multiplexer (ADM).
- At each ADM, circuits within the SONET/SDH frames can be dropped off (perhaps terminated on PDH equipment) and circuits can be added to the SONET/SDH stream, providing there's bandwidth for them.

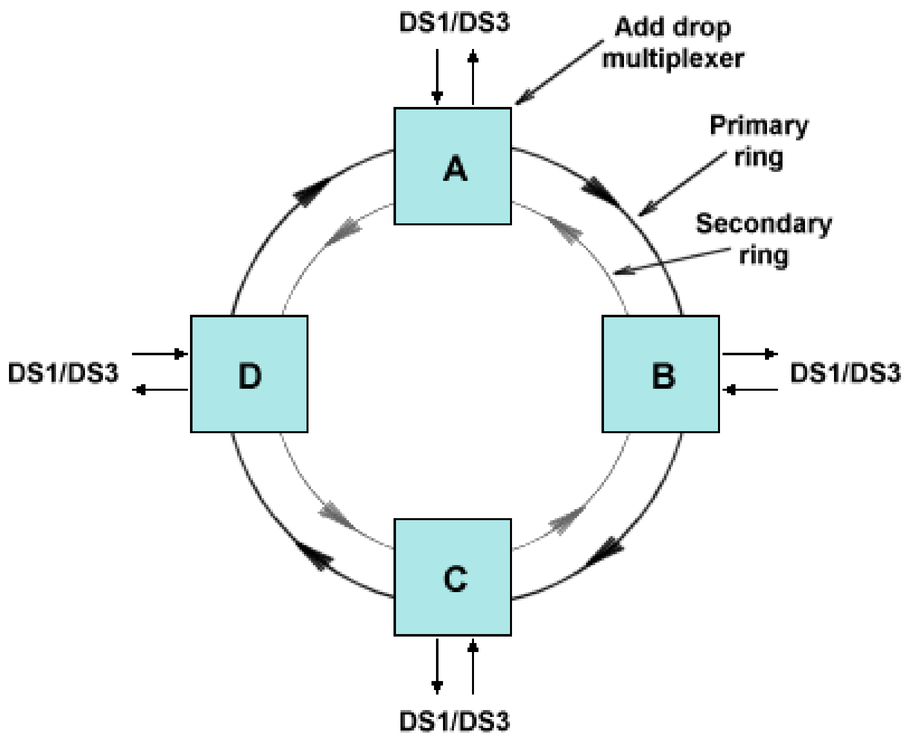
Linear Network

- In a linear network, traffic flows between the ADMs over two fibers, one carrying traffic in one direction, and the other carrying traffic in the other direction. Side note: the ADMs at the end of the fiber runs are called “terminal multiplexers.”
- A linear network can also have four fibers, with two held for backup in case of a cut to the working fiber(s).
- The backup fibers should, ideally, be run on a different route than the working fibers.



SONET/SDH Rings

- Rings are more common because they provide better recovery capability in the case of a fiber or node failure.
- We're going to examine how recovery occurs in rings in the next section.



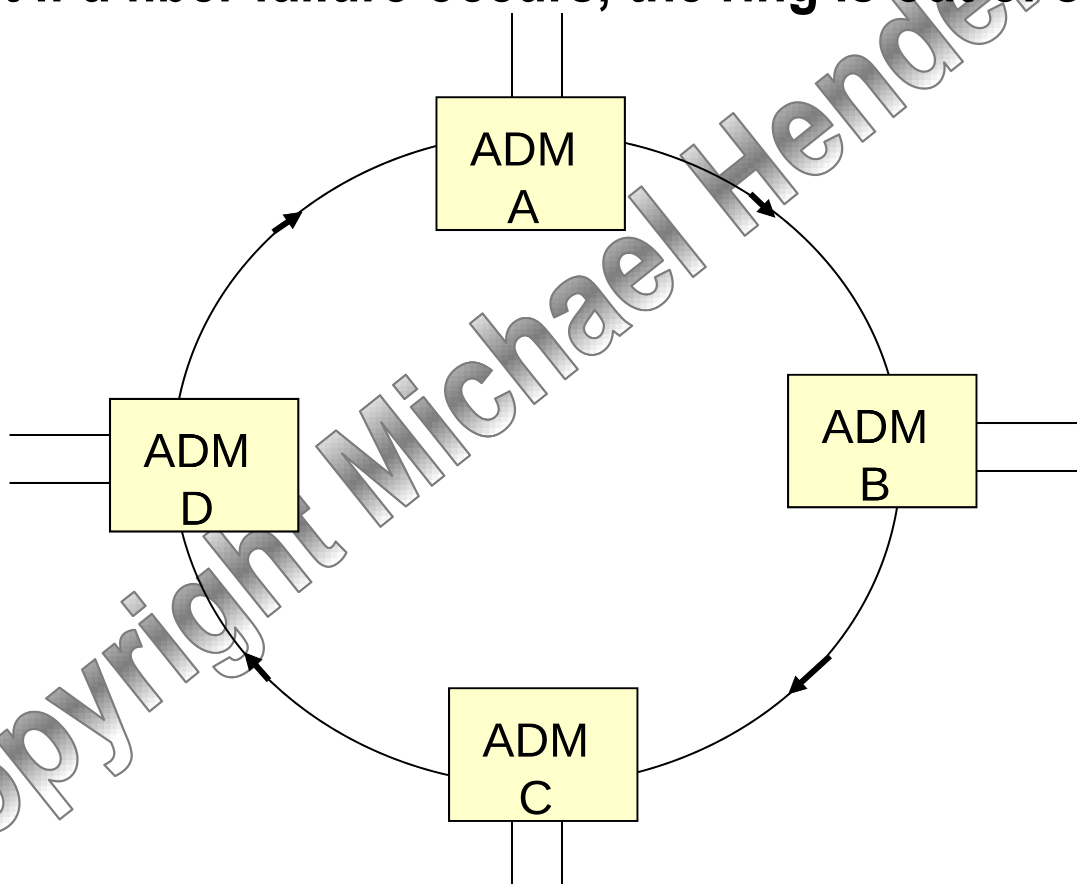
Automatic Protection Switching

SONET/SDH Rings

- There are two kinds of rings in SONET/SDH, leading to two kinds of recovery.
- The first is the Unidirectional Path Switch Ring (UPSR).
- The second is the Bidirectional Line Switched Ring (BLSR).
- We'll cover these in turn.

Unidirectional Path Switched Ring

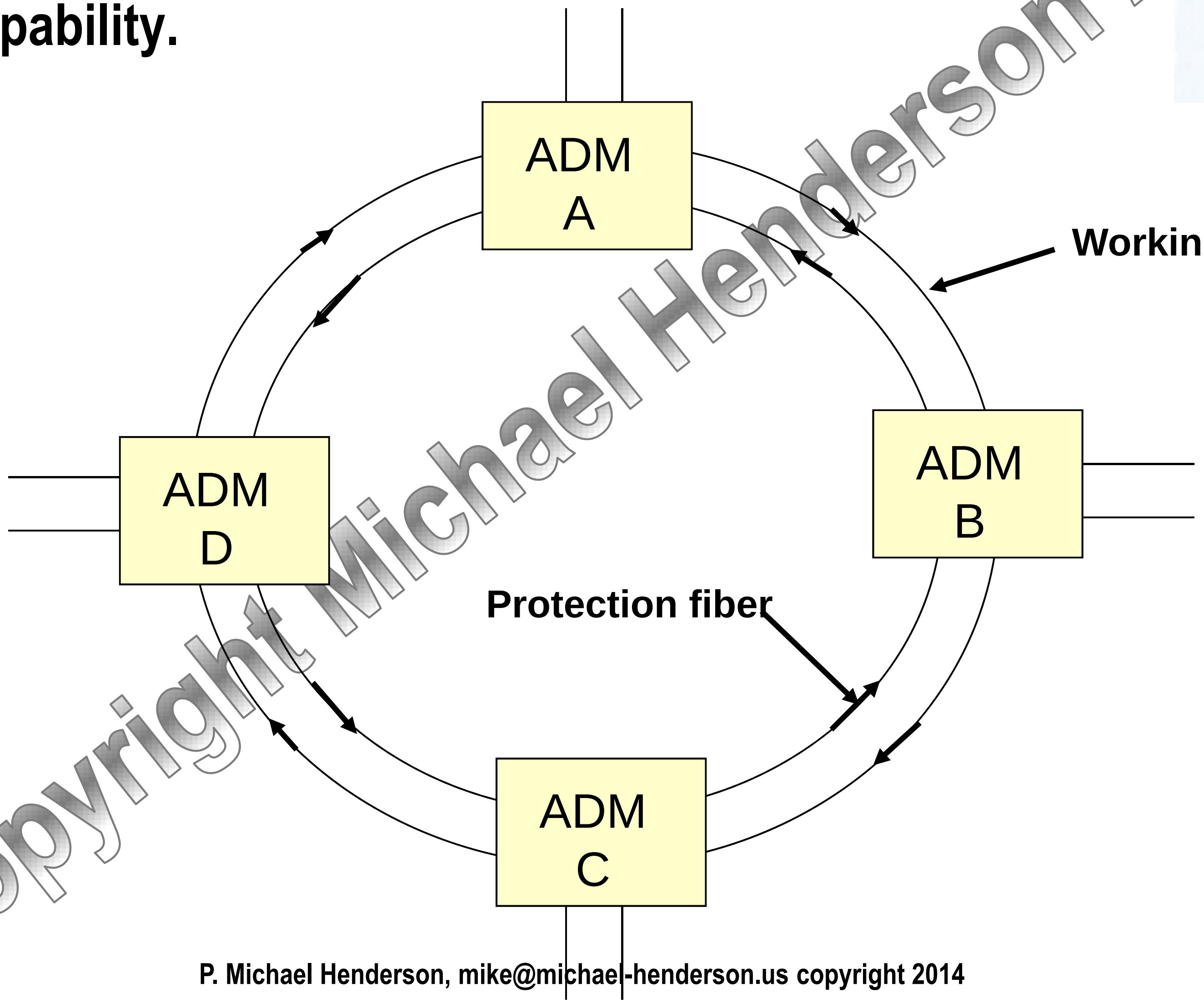
- **A ring could consist of just a single fiber connecting ADMs into a ring.**
- **But if a fiber failure occurs, the ring is out of service**



ADM A sends traffic directly to ADM B. But ADM A has to send traffic for ADM D all the way around the ring.

Unidirectional Path Switched Ring

- **By adding an additional fiber, we can provide a recovery capability.**

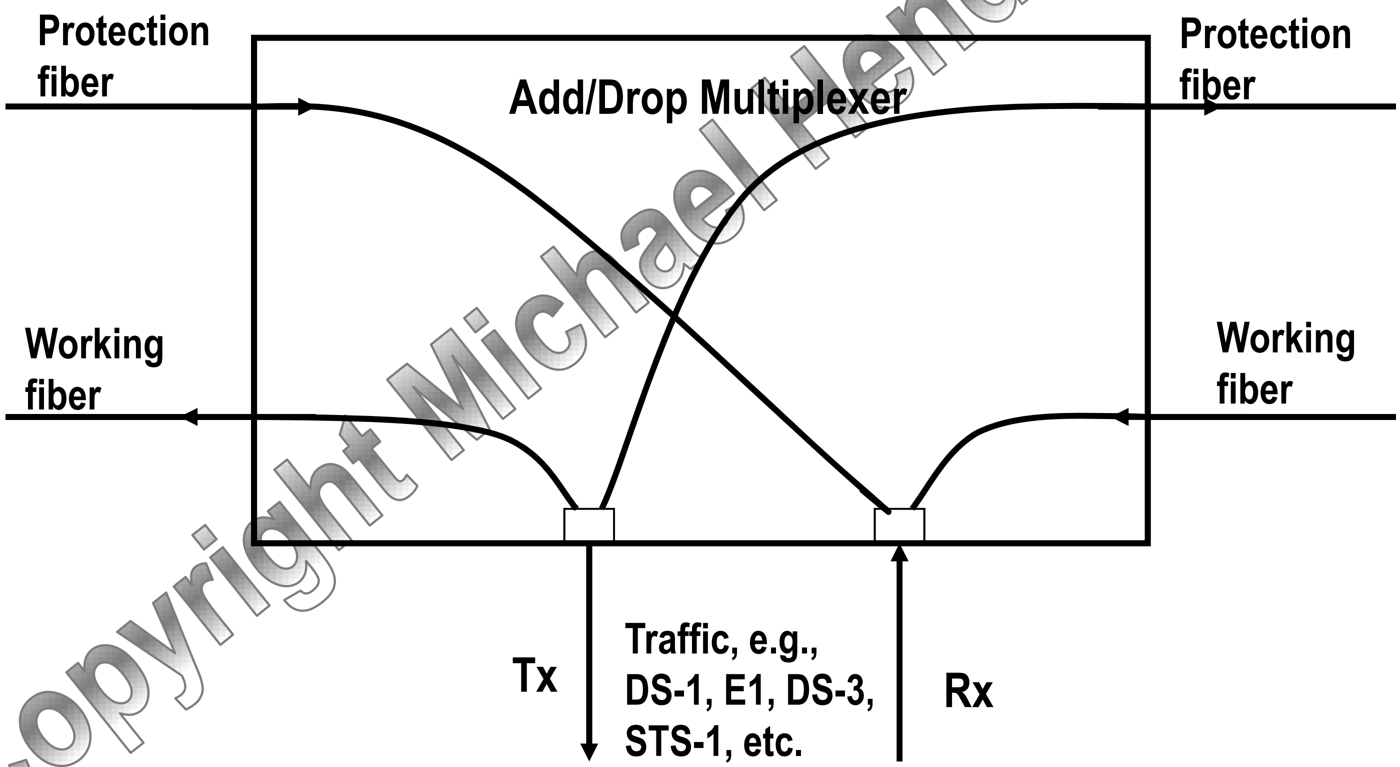


Path Switching

- **Today, path switching is only used on unidirectional rings – hence the name Unidirectional Path Switched Rings (UPSR).**
- **What does Path Switching mean?**
  - At the exit node, both fibers are monitored and the path traffic extracted.
  - If the signal is lost on the working fiber, a switch is done to use the protection fiber.
- **To a large degree, path switching can be done completely by the receiver, without coordination with the sender.**
- **But, path switching requires two sets of electronics to extract the path information.**

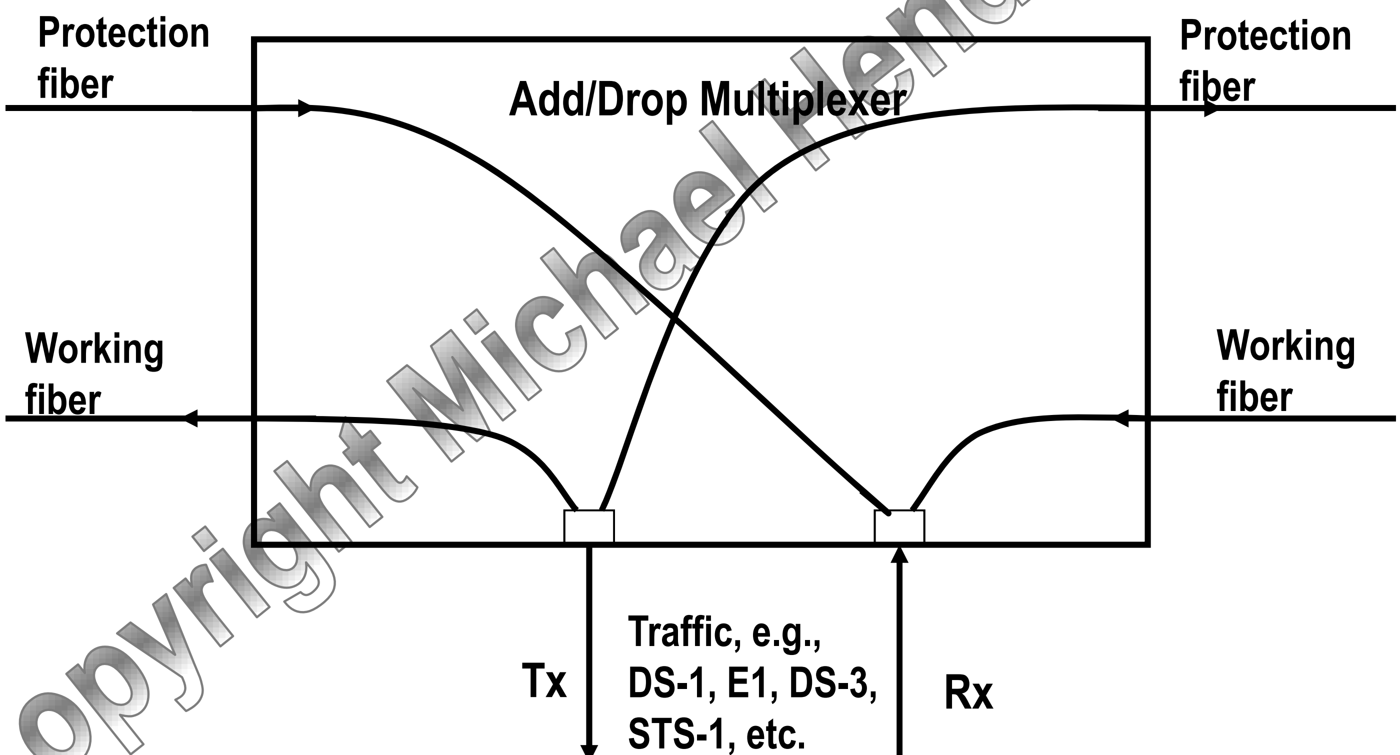
Recovery in UPSR

- As traffic flows through the ADM, each ADM transmits its signal to both the working fiber and the protection fiber.
- So the exact same traffic is flowing on both fibers in different directions.



Recovery in UPSR

- On the receive side, each ADM receives a signal from both the working fiber and the protection fiber.
- If the working fiber's signal is lost of degraded, it takes the traffic from the protection fiber. Switch over is very fast.



Disadvantage of UPSR

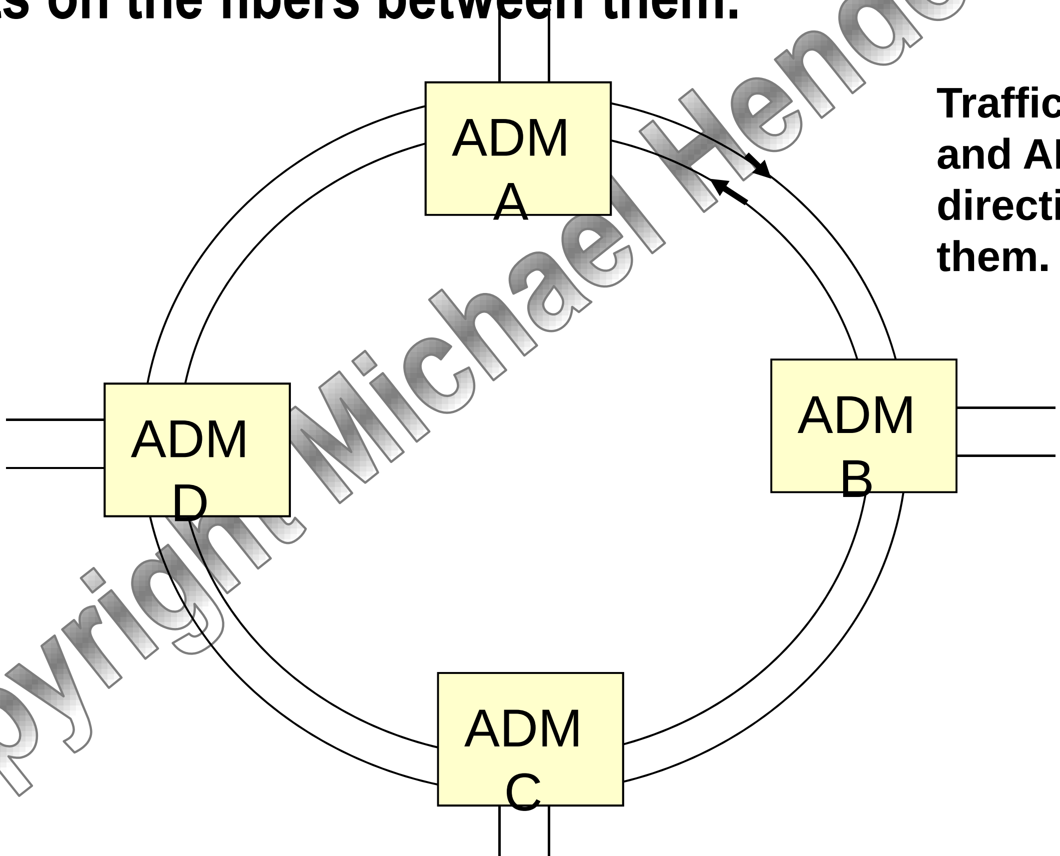
- **In a UPSR ring, all traffic flows on both fibers. So if we were to make an assumption that all of the traffic on the ring was between ADMs A and B, all of that traffic still flows through ADMs C and D.**
- **If we had a configuration where traffic between ADMs A and B only flowed between those two nodes, we could use the bandwidth of the fiber between the other nodes for additional traffic.**
- **That's why we have Bidirectional Line Switched Rings.**

Line Switching

- **Line Switching is done on bi-directional rings, either two fiber or four fiber. Thus the name Bi-directional Line Switched Ring (BLSR).**
- **The total capacity of the fibers must be twice the carried traffic. For four fiber systems, this means two fibers are dedicated for protection.**
- **What does Line Switching mean?**
  - Two adjacent nodes monitor the traffic between them.
  - If one detects “failure” on a fiber, it signals the other.
  - The two nodes coordinate switching to the protection fiber(s) or STS.
- **The K1, K2 octets in the transport overhead are used for this signaling.**

Bidirectional Line Switched Rings

- A 2-fiber BLSR ring uses the two fibers to send data in both directions around the ring.
- This means that traffic between two adjacent nodes only exists on the fibers between them.



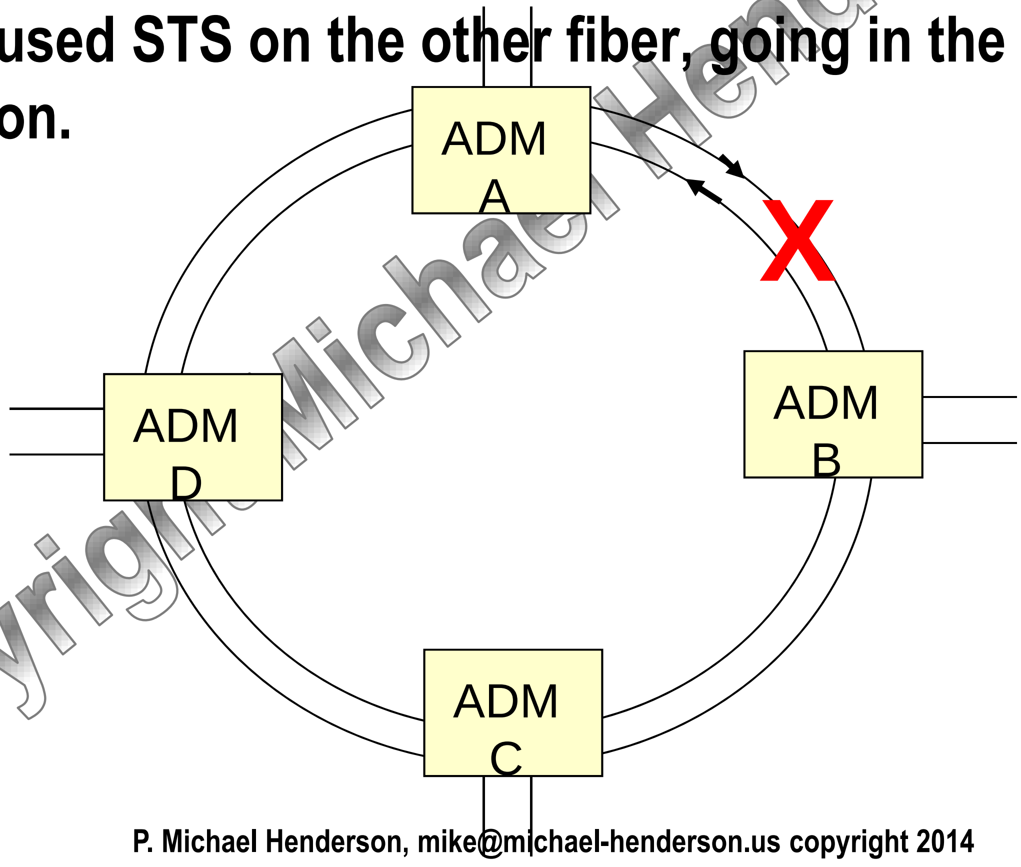
Traffic between ADM A and ADM B flows bi-directionally between them.

Bidirectional Line Switched Rings

- Each fiber only uses half the STS “frames” available on the signal.
- So one ring will use STS 1-24 for working traffic, while reserving STS 25-48 for protection, and the other ring will use STS 25-48 for working, while reserving STS 1-24 for protection (this is for an OC-48 signal).

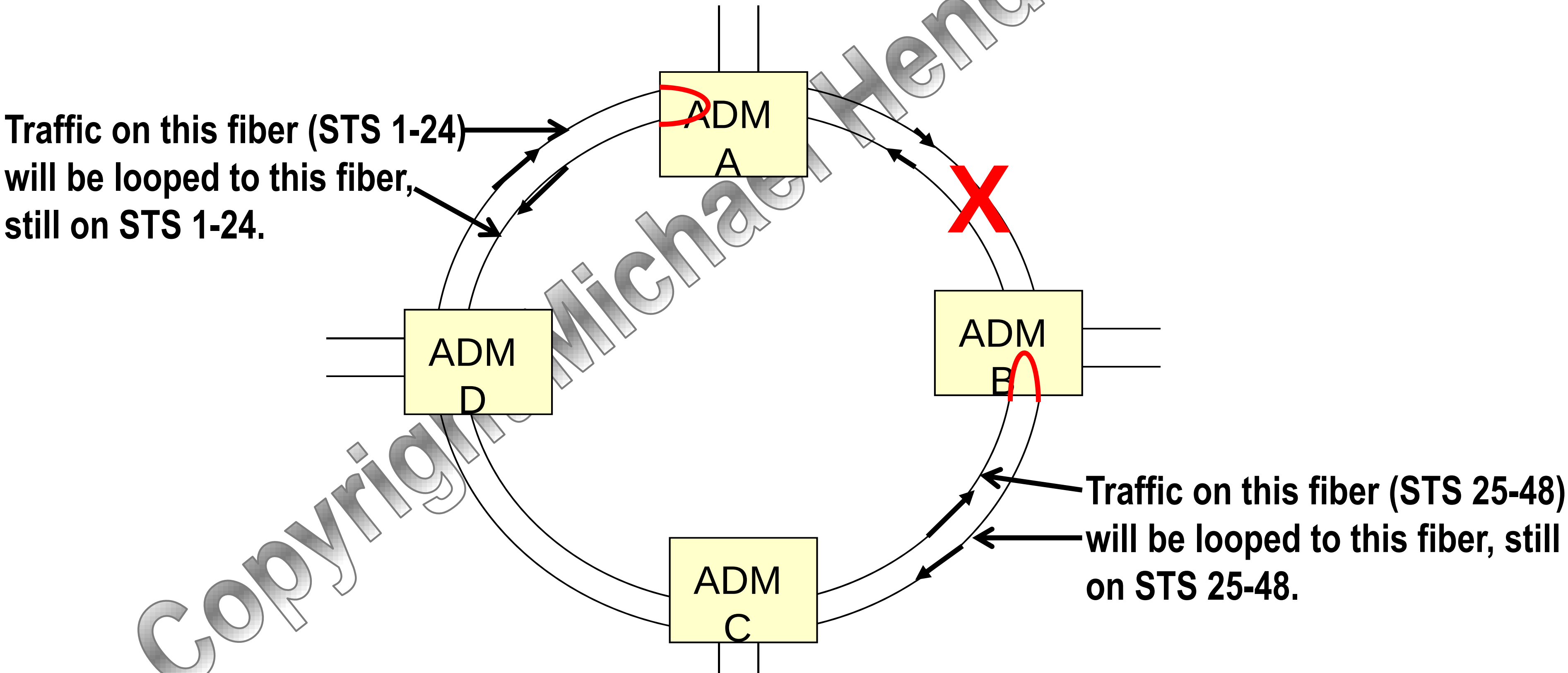
Recovery in 2-fiber BLSR

- **If the fiber breaks between ADM A and B, ADM B will detect loss of signal.**
- **Using the K1/K2 octets (to be explained), ADM B signals ADM A that the break occurred and the traffic is placed in the unused STS on the other fiber, going in the opposite direction.**

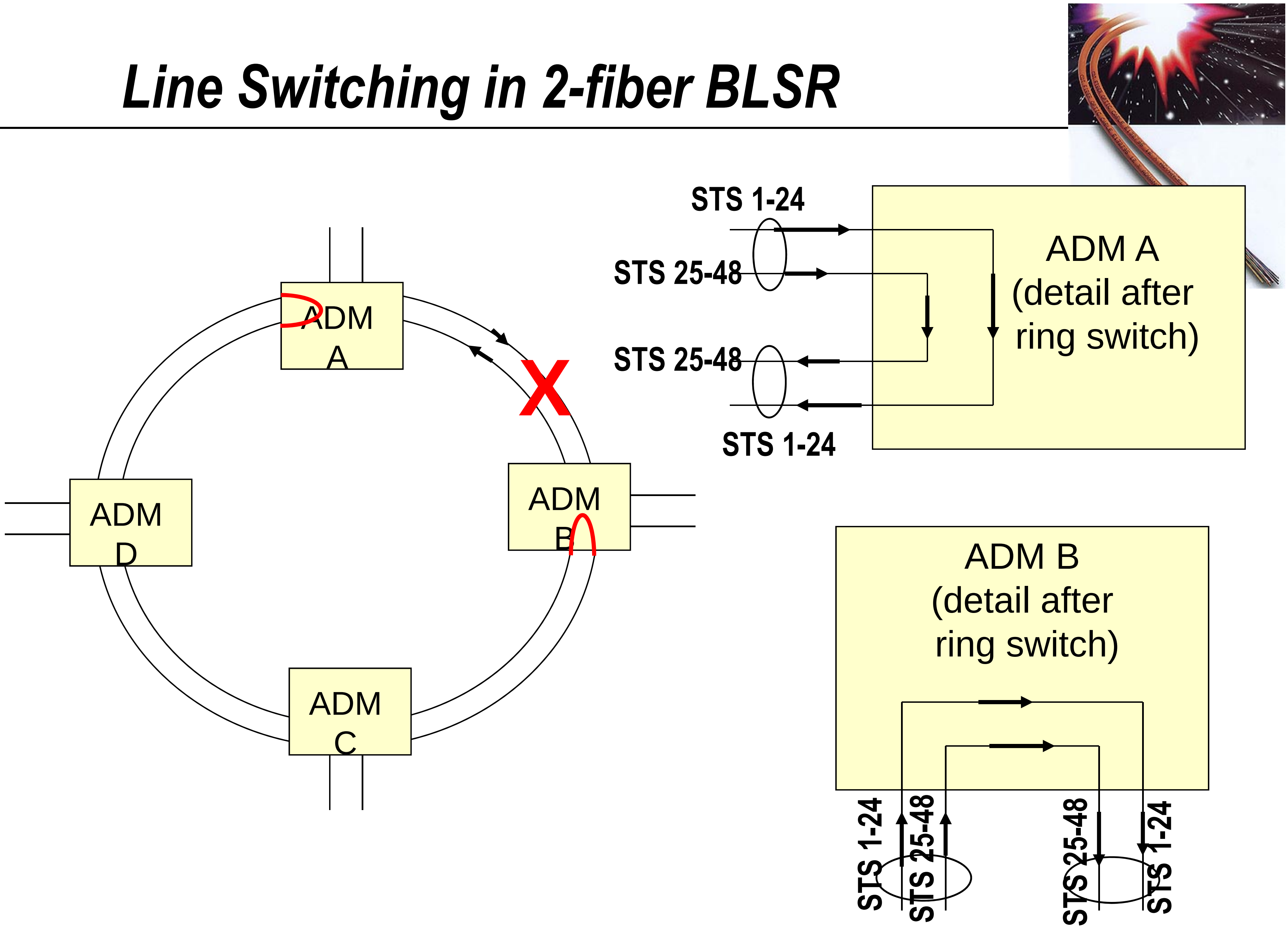


Recovery in 2-fiber BLSR

- **When a fiber failure occurs, say between A and B in the diagram, A and B will loop the traffic on the receive fiber to the transmit fiber.**



Line Switching in 2-fiber BLSR

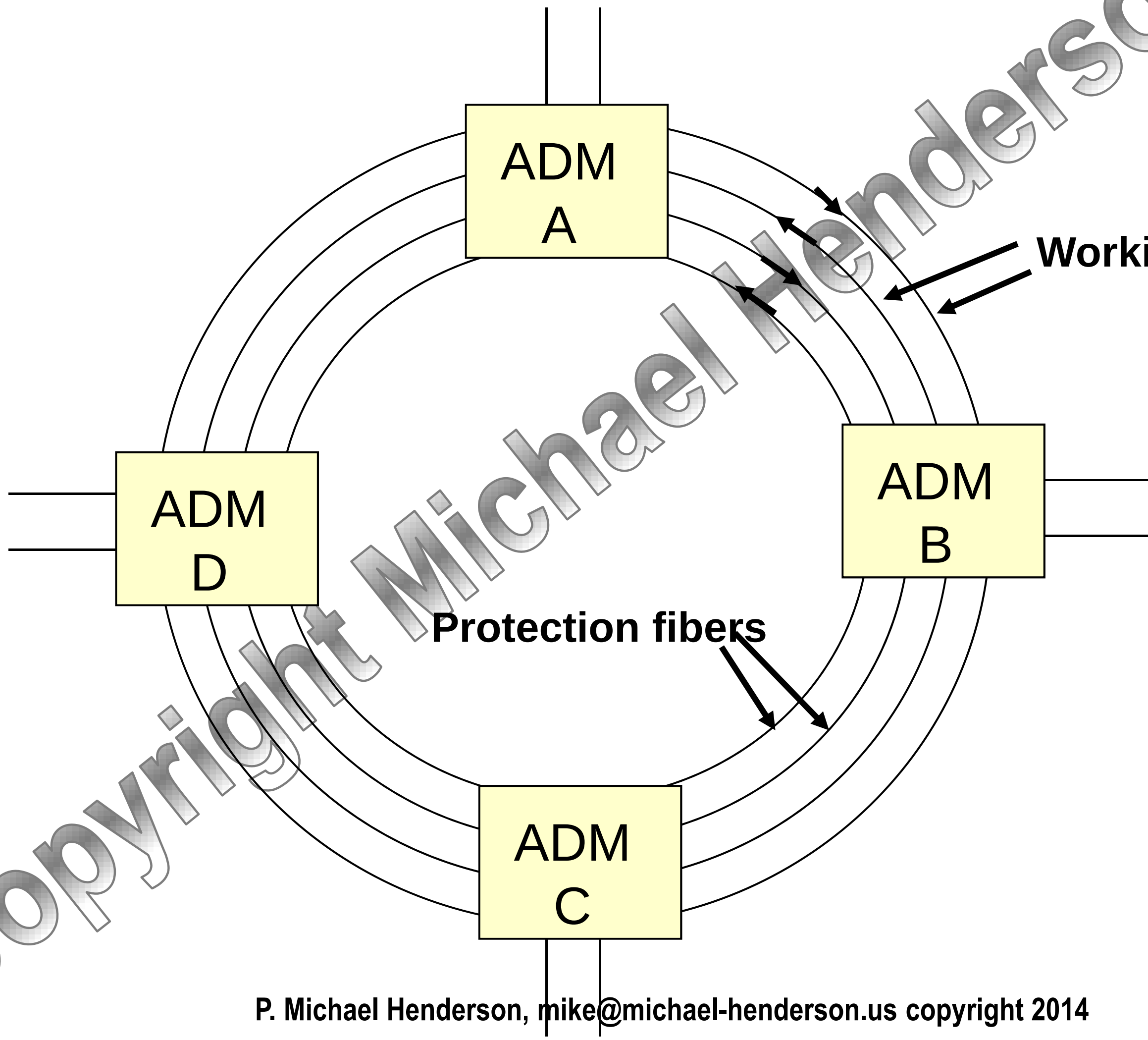


Discussion of 2-fiber BLSR

- Since traffic between two adjacent nodes only flows between those two nodes, that bandwidth could be used for traffic between the other nodes. But for 2-fiber BLSR this may require that some traffic be dropped when a line switch occurs.
- When a line switch occurs, nodes not involved in the switch continue to function normally. Nodes involved in the switch make the changes necessary to continue taking data from the proper STS.

4-fiber BLSR

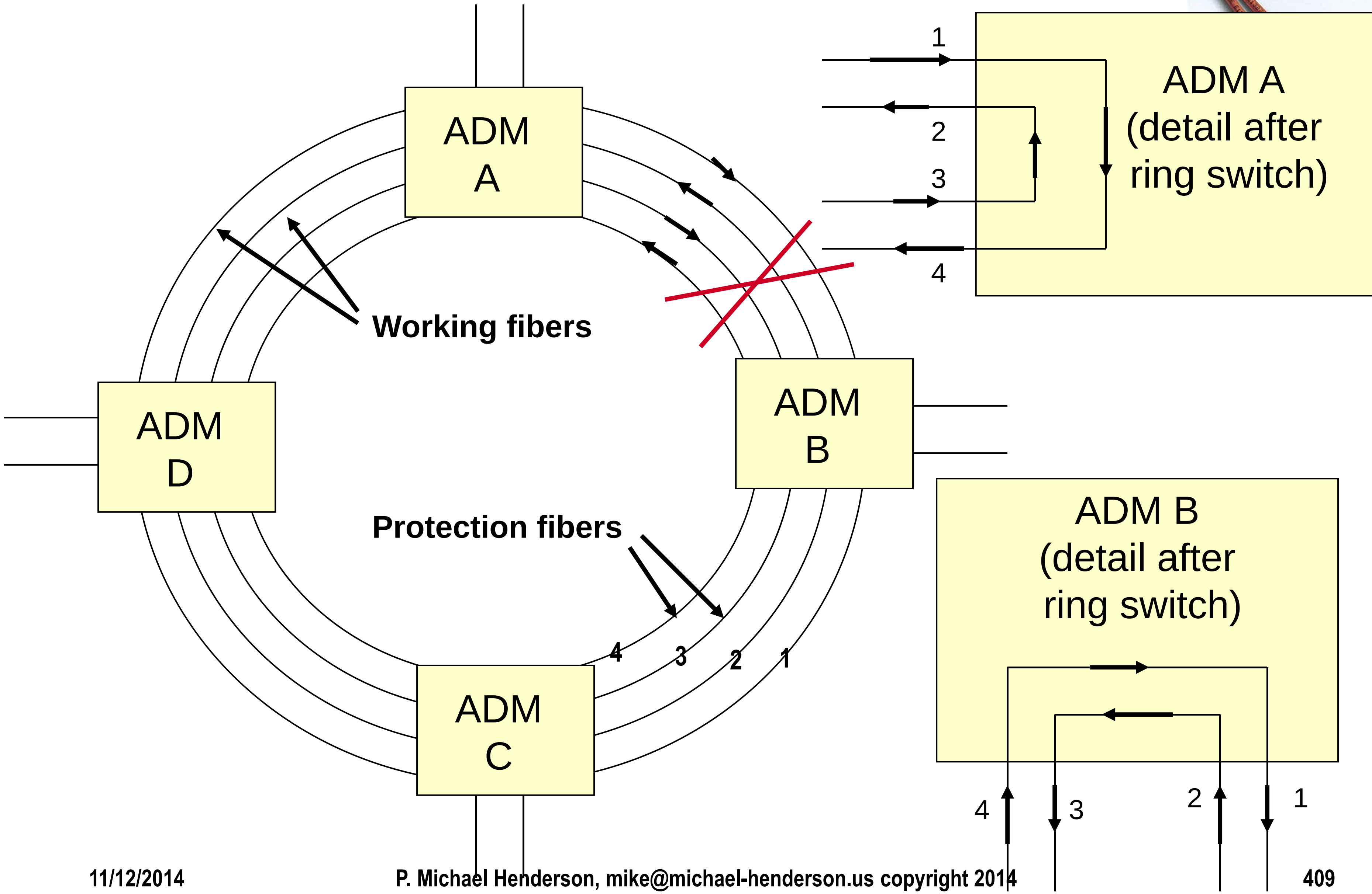
- **Switching is easier in 4-fiber than 2-fiber because the data stays on the same STS – it just switches fibers.**



Advantages of 4-fiber BLSR

- The capacity of the ring is twice the capacity of a 2-fiber BLSR.
- Traffic between two adjacent nodes, such as ADM A and B, only travels between those two nodes – it does not travel through the other nodes.
- This means bandwidth used between the two adjacent nodes – A & B for example – can now be used for traffic between the other nodes.
- This can provide more capacity on the ring than 2-fiber BLSR or UPSR. But note that some traffic may have to be dropped when a line switch occurs.

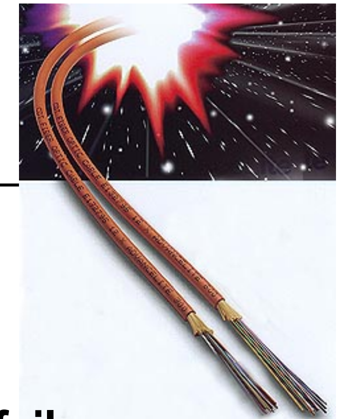
Line Switching in 4-fiber BLSR



K1/K2 Octets

- When a failure occurs, as shown in the previous diagram, Nodes A and B will signal the failure through the K1, K2 octets.
- They will place the other node's address in K1 and their address in K2. Each address is 4 bits so the other 4 bits are used to convey additional information.
- Nodes C and D will pass the data through since it is not addressed to them.
- This signaling will cause nodes A and B to execute a ring switch, as shown on the previous slide.
- The 2-fiber BLSR switch occurs exactly the same way.

K1/K2 Octets



A1	A2	J0
B1	E1	F1
D1	D2	D3
H1	H2	H3
B2	K1	K2
D4	D5	D6
D7	D8	D9
D10	D11	D12
S1	M0/1	E2

- These octets (K1, K2) are used to send messages between two nodes when a failure is detected between them.
- Messages are sent both ways around the ring (if possible).
- Last four bits of K1 specifies the address of the addressed node.
- First four bits of K2 specifies the address of the sending node.
- Means that there can only be 16 nodes on a ring.

Squelching

- If a node fails, as opposed to a fiber, it's important to "squelch" the links that terminated on that node.
- Otherwise, traffic for that node will continue to enter the ring and could cause congestion.
- Squelching is signaled by sending AIS in the channels that terminated on the failed node.
- The sending node(s) will stop sending traffic for the failed node until they receive a message that the node is back up.

SONET Performance Monitoring

SONET Performance Monitoring

- **SONET performance monitoring falls into the following categories:**
  – Anomaly
  – Defect
  – Failure

Anomaly

- **An anomaly is the lowest level of problem report and usually does not cause an interruption of communications.**
- **Some anomaly conditions are:**
  - B1 Bit Interleaved Parity (BIP)
  - B2 BIP
  - Path B3 BIP
  - Remote Error Indication (REI). Indicates to the downstream node that an error occurred.

Bit Interleaved Parity

- For Bit Interleaved Parity, the bits that make up the bytes of the block to be check are added up position-wise. That is, all of the bits in the first position are added, all of the bits in the second position are added, etc.
- If the number of 1 bits in any position are odd, the parity bit is 1 to make the count even.
- If the number of 1 bits in any position are even, the parity bit is 0.

Example of BIP-8 Parity

"Monitored portion"

8-bits word	8-bits word	8-bits word	8-bits word	8-bits word
Byte 5	Byte 4	Byte 3	Byte 2	Byte 1

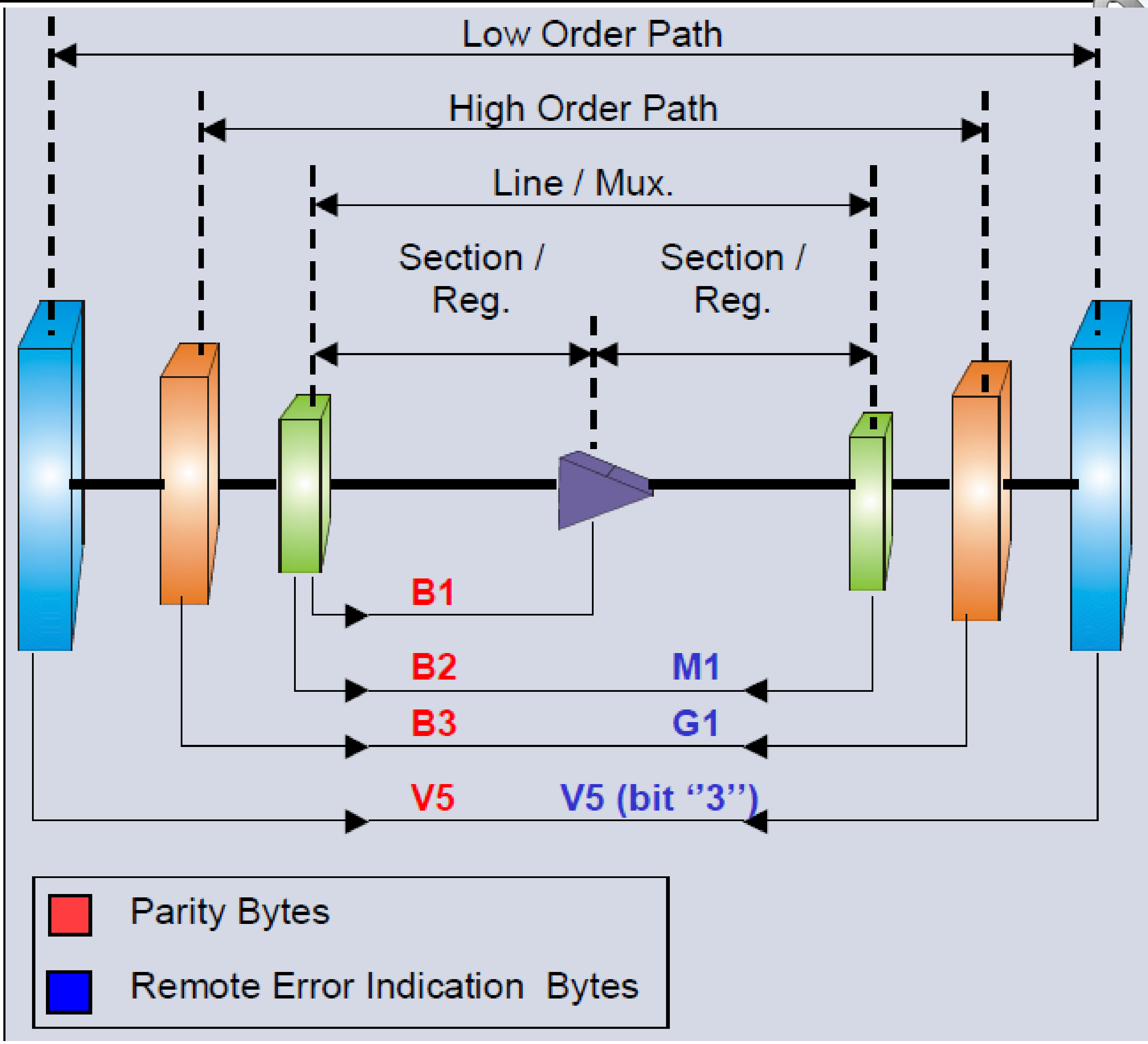
Bit position	1	2	3	4	5	6	7	8
Word 1	1	0	1	0	1	0	1	0
Word 2	0	1	0	1	0	1	0	1
Word 3	1	0	1	0	1	0	1	0
Word 4	0	1	0	1	0	1	0	1
Word 5	1	1	0	0	1	1	0	0
«1's» count	3	3	2	2	3	3	2	2
Odd/Even	Odd	Odd	Even	Even	Odd	Odd	Even	Even

An odd total in the "1's" count row causes a binary "1" to be placed in the same position below.

BIP-8 code	1	1	0	0	1	1	0	0

BIP- 8 calculation process

Where BIP-8 is Used



B1 Parity Check

A1	A2	J0
B1	E1	F1
D1	D2	D3
H1	H2	H3
B2	K1	K2
D4	D5	D6
D7	D8	D9
D10	D11	D12
S1	M0/1	E2

- Does a bit interleaved parity (BIP-8) over the entire (previous) frame.
- Even parity.
- Because it covers so many bits, it does not give a good estimate of BER for larger frames (STS-48/STM-16 or larger).

B2 Parity Check

A1	A2	J0
B1	E1	F1
D1	D2	D3
H1	H2	H3
B2	K1	K2
D4	D5	D6
D7	D8	D9
D10	D11	D12
S1	M0/1	E2

- Does a bit interleaved parity (BIP-8) over the STS-1 of the previous frame.
- Even parity.
- Since it covers less bits than the B1 octet, it gives a better estimate of the BER.
- One B2 octet for each STS-N.
- For SDH, there are three B2 octets for an STM-1, giving a BIP-24.
- The Section Overhead (SONET) or Regenerator Section Overhead (SDH) is not included in the B2 check

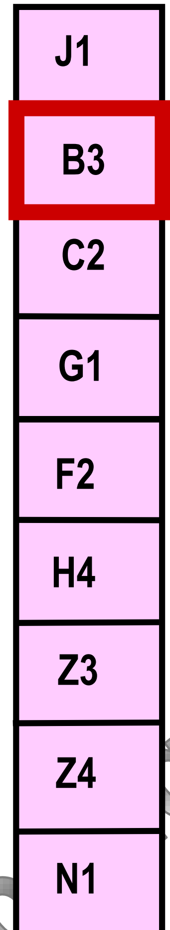
Interleaving of SONET Signals

270 Columns

A1	A1	A1	A2	A2	A2	J0	Z0	Z0
B1	X	X	E1	X	X	F1	X	X
D1	X	X	D2	X	X	D3	X	X
H1	H1	H1	H2	H2	H2	H3	H3	H3
B2	B2	B2	K1	X	X	K2	X	X
D4	X	X	D5	X	X	D6	X	X
D7	X	X	D8	X	X	D9	X	X
D10	X	X	D11	X	X	D12	X	X
S1	Z1	Z1	M0/1	Z2	M2	E2	X	X

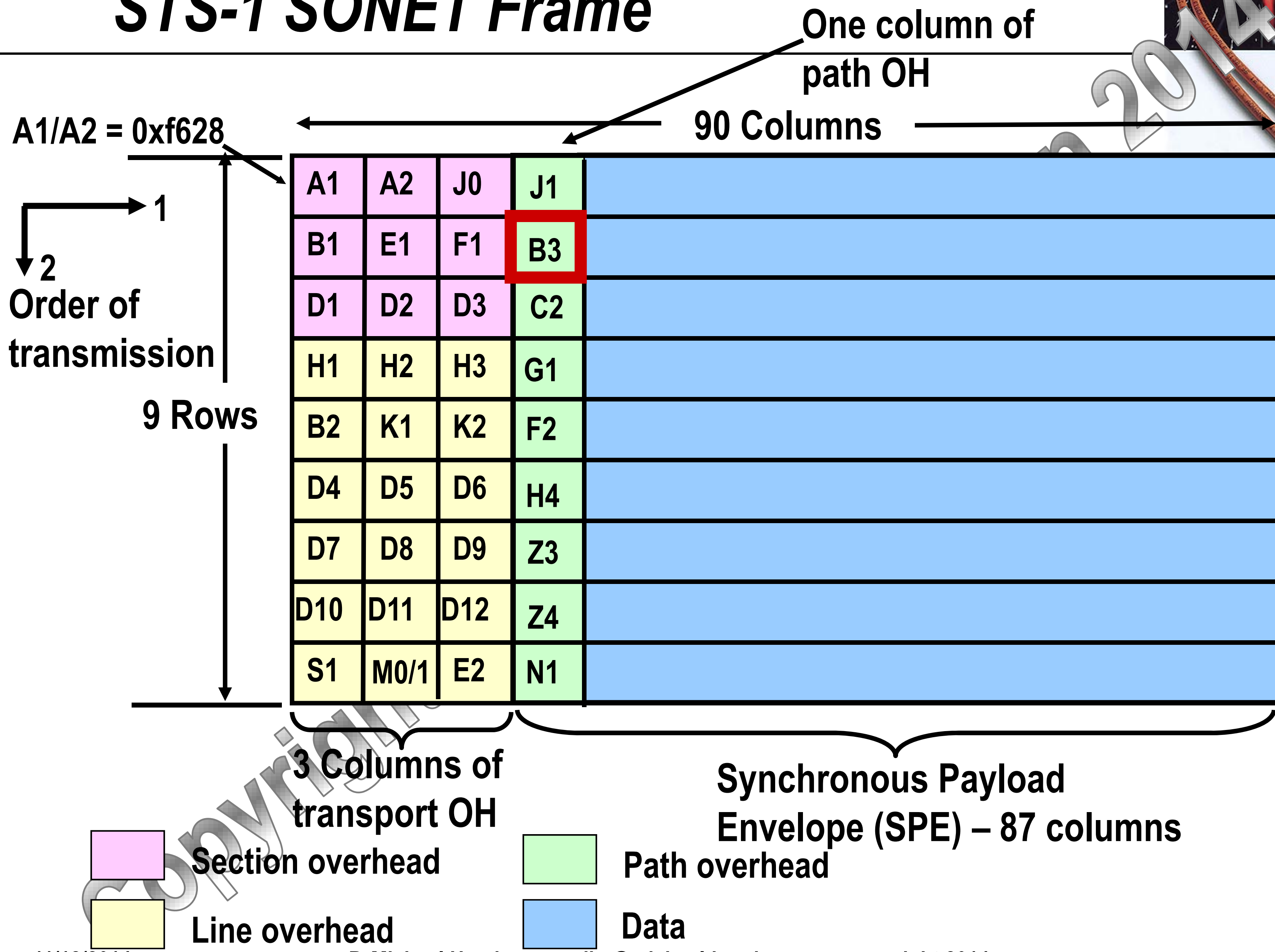
First STS-1 Second STS-1 Third STS-1

B3 Parity Check



- BIP-8 parity check over the payload, only.
- Even parity.
- Parity is for *previous* payload, not this one.
- For SDH, the check is over the VC-4

STS-1 SONET Frame

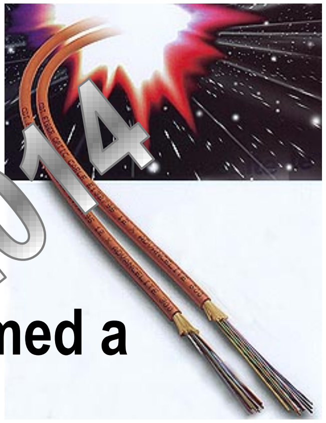


Remote Error Indication

A1	A2	J0
B1	E1	F1
D1	D2	D3
H1	H2	H3
B2	K1	K2
D4	D5	D6
D7	D8	D9
D10	D11	D12
S1	M0/1	E2

- M0/M1 is used to send back to the sender, the error status of the received signal (determined by the Bx octets). Number of bit errors detected.
- M0 is only used by SONET.
- M1 is used by both SONET and SDH to send a count of the number of errors found by the B2 octets. For STM-1 in SDH, B2 will be a BIP-24, so the number of errors could be 0 to 24.
- This value is truncated at 255 for STM levels greater than STM-4, STS levels greater than STS-12.

Defect



- **When a more serious status condition occurs, it is termed a "defect".**
- **Defects are as follows:**
  - Out Of Frame Alignment (OOF)
  - Loss of Frame (LOF)
  - Loss of Pointer (LOP)
  - Loss of Superframe (LOS)
- **These may result in the following status signals:**
  - Alarm Indication Signal (AIS)
  - Remote Defect Indication (RDI)

Out of Frame Alignment

- OOF state occurs when several consecutive SONET frames are received with invalid framing patterns (A1 and A2 bytes).
- The maximum time to detect OOF is 625 microseconds (5 frames).
- OOF state clears within 250 microseconds when two consecutive SONET frames are received with valid framing patterns.

Loss of Frame Alignment

- **LOF state occurs when the OOF state exists for a specified time in microseconds.**
- **The LOF state clears when an in-frame condition exists continuously for a specified time in microseconds.**
- **The time for detection and clearance is normally 3 milliseconds.**
- **Indicated downstream by sending AIS.**

Loss of Pointer

- **LOP state occurs when N consecutive invalid pointers are received or N consecutive New Data Flags (NDF) are received.**
- **LOP state is cleared when three equal valid pointers are received.**
- **LOP can be identified as:**
  - LOP-P (Loss of the H1/H2 pointer to the Payload).
  - LOP-VT (Loss of the V1/V2 pointer to the Virtual Tributaries).

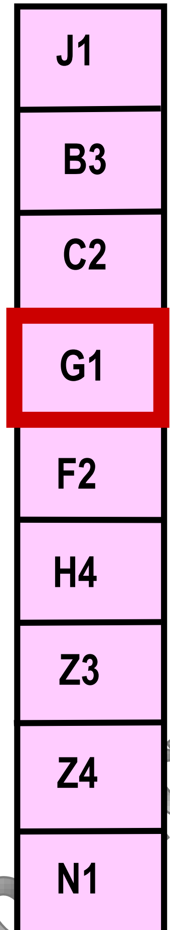
Loss of Superframe

- Remember that the last two bits of H4 in the POH count 00, 01, 10, 11, 00, etc. This produces a superframe of four frames.
- If there's an error in the count of those last two bits, for example, if they go 00, 11, 10, 11, 00, the system will indicate loss of superframe.

Alarm Indication Signal

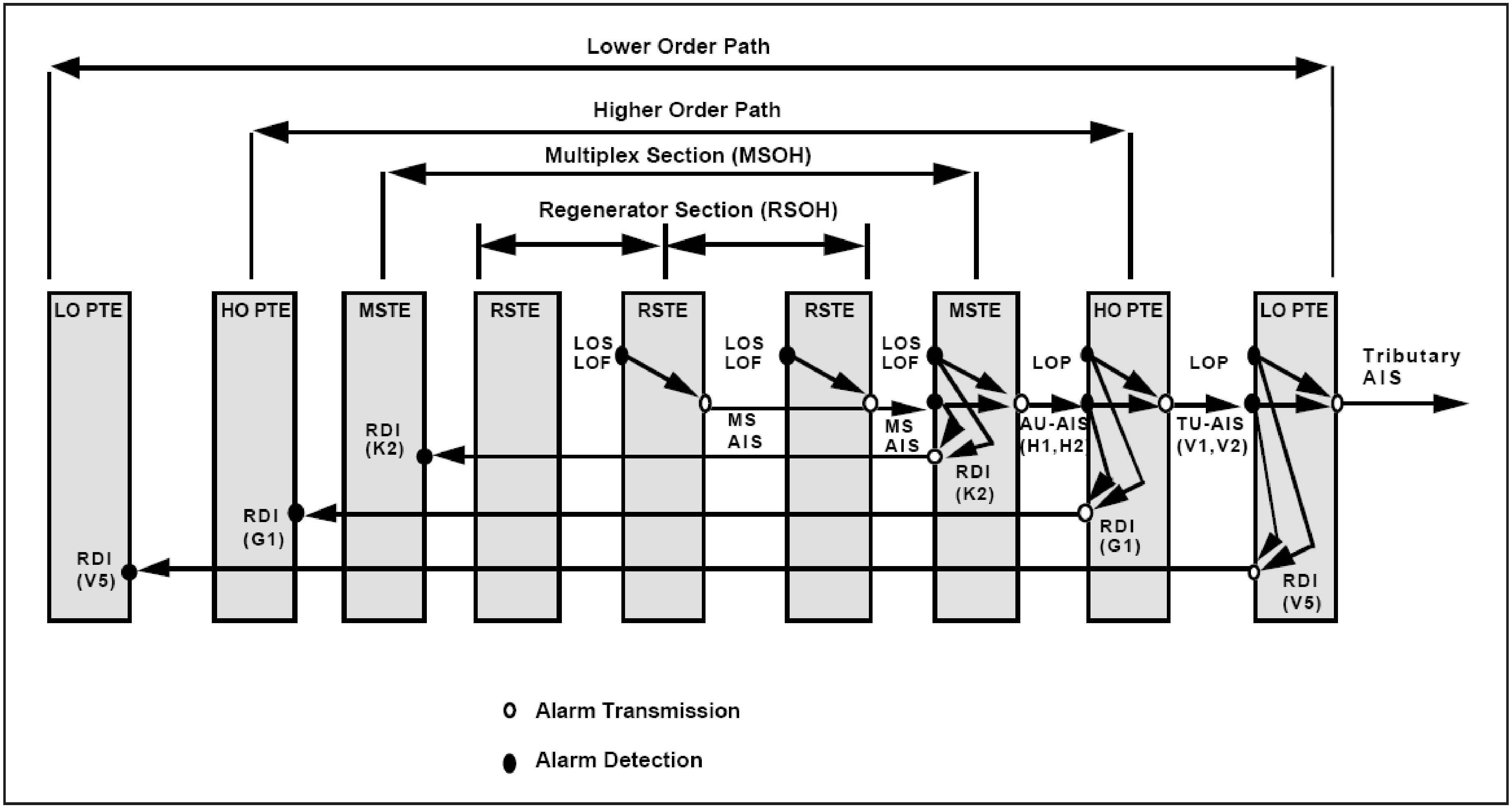
- When a signal is lost – perhaps one that is carried within the SONET payload – that signal is replaced with all ones to alert the equipment down line that the failure occurred.

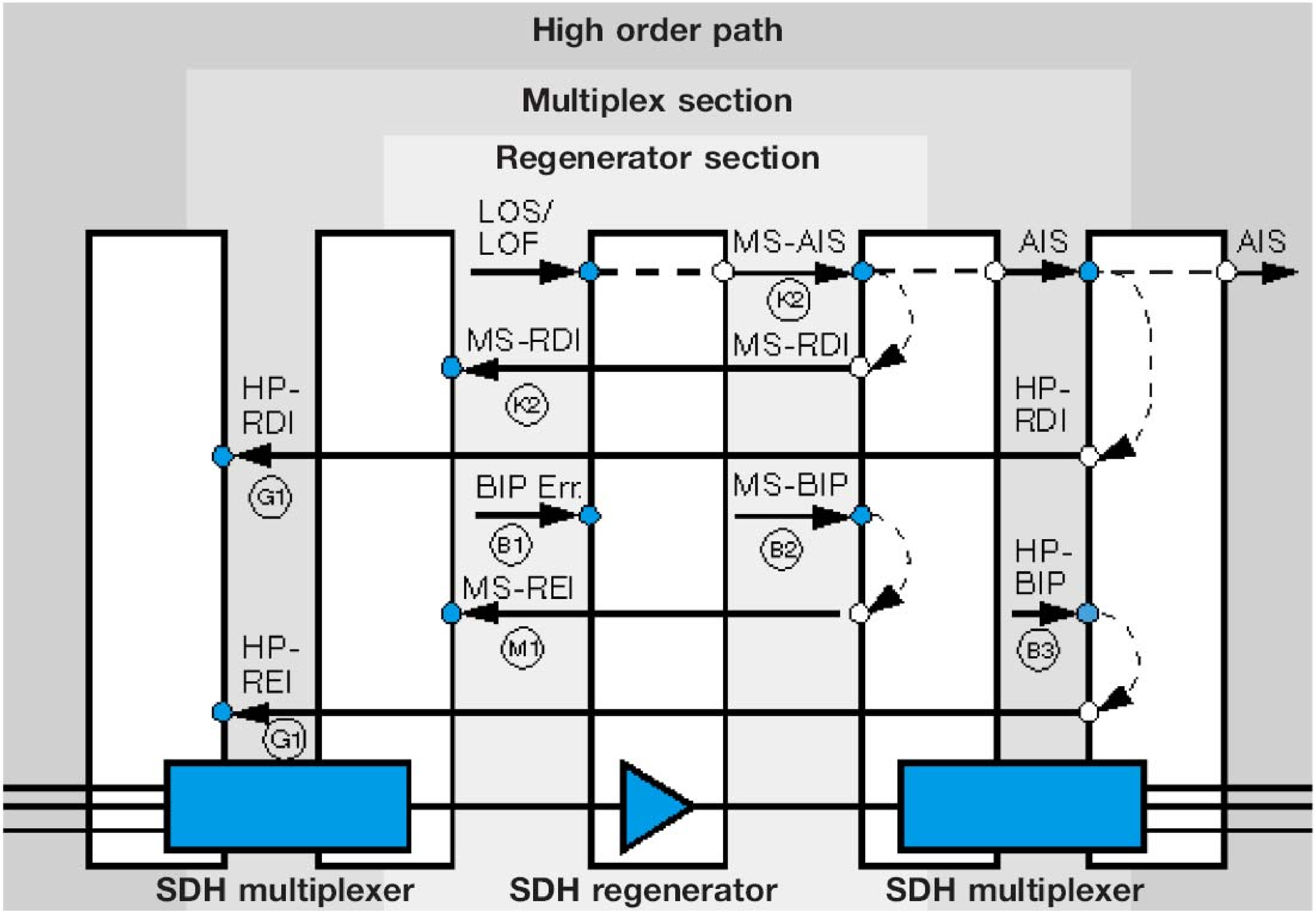
Remote Defect Indication (RDI)



- The G1 octet (Path Status) sends information back to the sender indicating the number of parity errors, or if a complete failure is detected.
- Can send back indication of Loss of Signal, Loss of Frame, or AIS received.
- Previously known as Far End Receiver Failure (FERF).

Signals in Response to Defects



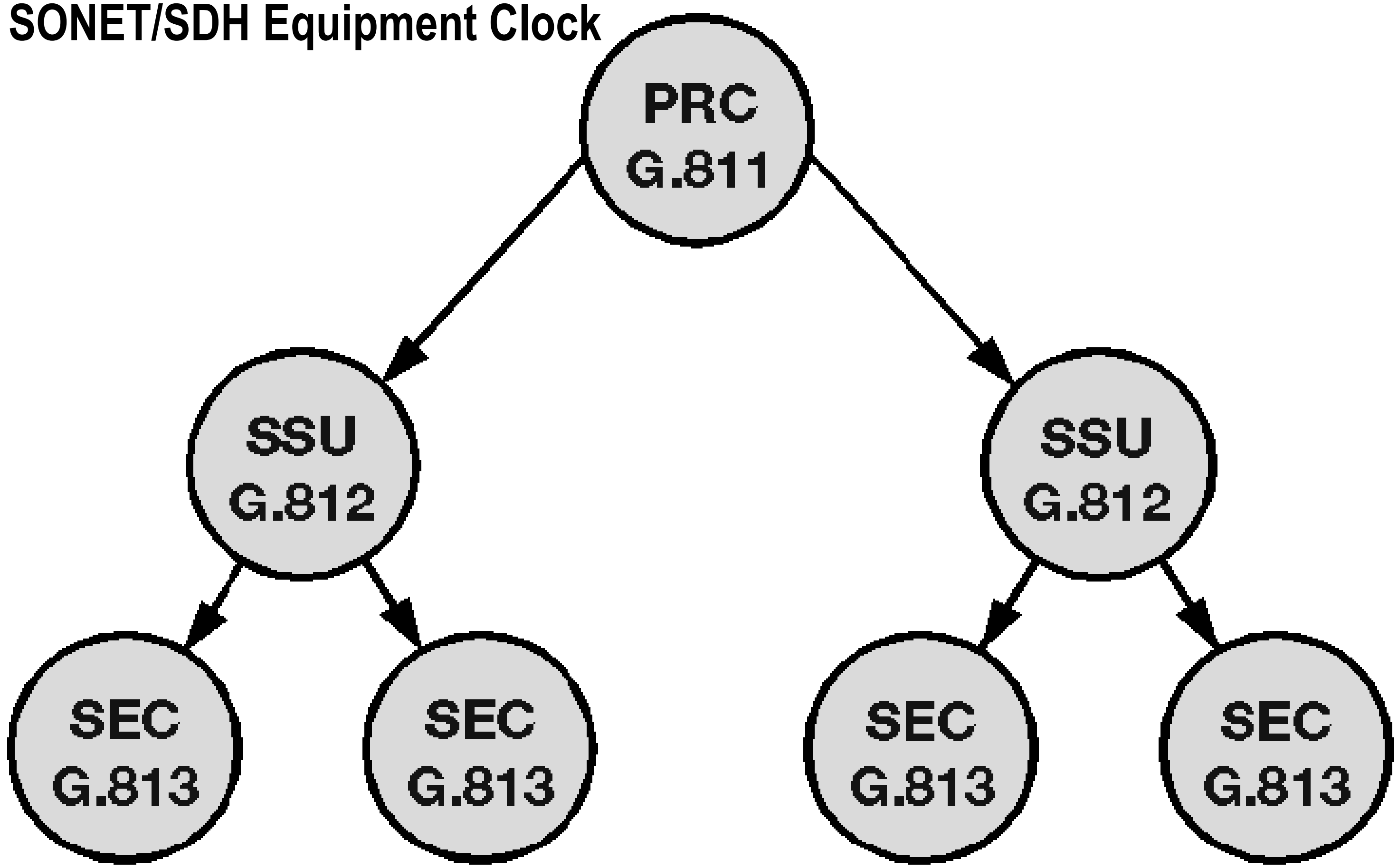

High order path
Multiplex section
Regenerator section
LOS/
LOF
MS-AIS
K2
AIS
AIS
MS-RDI
MS-RDI
K2
HP-
RDI
HP-
RDI
G1
BIP Err.
MS-BIP
B1
B2
HP-
BIP
MS-REI
M1
B3
HP-
REI
G1
SDH multiplexer
SDH regenerator
SDH multiplexer

Clocking in SONET/SDH

- **SONET/SDH is a synchronous system, meaning that all nodes need to operate off the same clock.**
- **There are no "stuff bits" in SONET/SDH to accommodate lines with slightly different clocks.**
- **SONET/SDH nodes, therefore, need to operate off the same clock, or off of different clocks that are extremely accurate and synchronized.**
- **An individual network is usually synchronized from a Primary Reference Clock (PRC), a stratum 1 clock.**
- **That clock is fed to other nodes by using the data stream. This introduces some jitter in the clock so there's a limit to how many levels clock can be fed.**

Clock Supply Hierarchy

PRC = Primary Reference Clock
SSU = Synchronization Supply Units
SEC = SONET/SDH Equipment Clock



Synchronization Status Byte

A1	A2	J0
B1	E1	F1
D1	D2	D3
H1	H2	H3
B2	K1	K2
D4	D5	D6
D7	D8	D9
D10	D11	D12
S1	M0/1	E2

- **S1 is used for synchronization status.**
- **Bit usage of the S1 octet is shown in the next slide.**

Synchronization Status Byte

S1 Byte		SONET synchronization quality level description
Bit 7 - Bit 4	Bit 3 - Bit 0	
Reserved	0000	Synchronized-tracebility unknown
Reserved	0001	Stratum 1 traceable
Reserved	0100	Transit node clock traceable
Reserved	0111	Stratum 2 traceable
Reserved	1010	Stratum 3 traceable
Reserved	1100	SONET minimum traceable
Reserved	1101	Stratum 3E traceable
Reserved	1110	Reserved for network synchronization
Reserved	1111	Do not use for synchronization

Clock Synchronization

- Clock should be distributed in a tree fashion.
- It is essential that timing loops not be formed.
- The tree must be rooted on a PRC.
- The tree should be of minimal depth.
- Note: An example of a timing loop is where the PRC fails and the nodes have to decide where to take clock. A provides clock to B, who provides clock to C, who provides clock to A. In this case performance will rapidly deteriorate.

Switching

Switching

- Now that we understand how digital voice traffic is carried over the network, let's take a look at how it's switched.
- The ITU defines switching as:

  "The establishing, on-demand, of an individual connection from a desired inlet to a desired outlet within a set of inlets and outlets for as long as is required for the transfer of information."

Why Switches?

- But why do we use switches? Switches are expensive. Why not just use more circuits?
- Switches allow reduction in overall network cost by reducing the number and/or the cost of transmission links required to enable a given user population to communicate.
- Limited number of physical connections implies need for sharing of transport resources, which means better utilization of transport circuits.
- Switching systems are central components in communications networks.

Circuit Switching

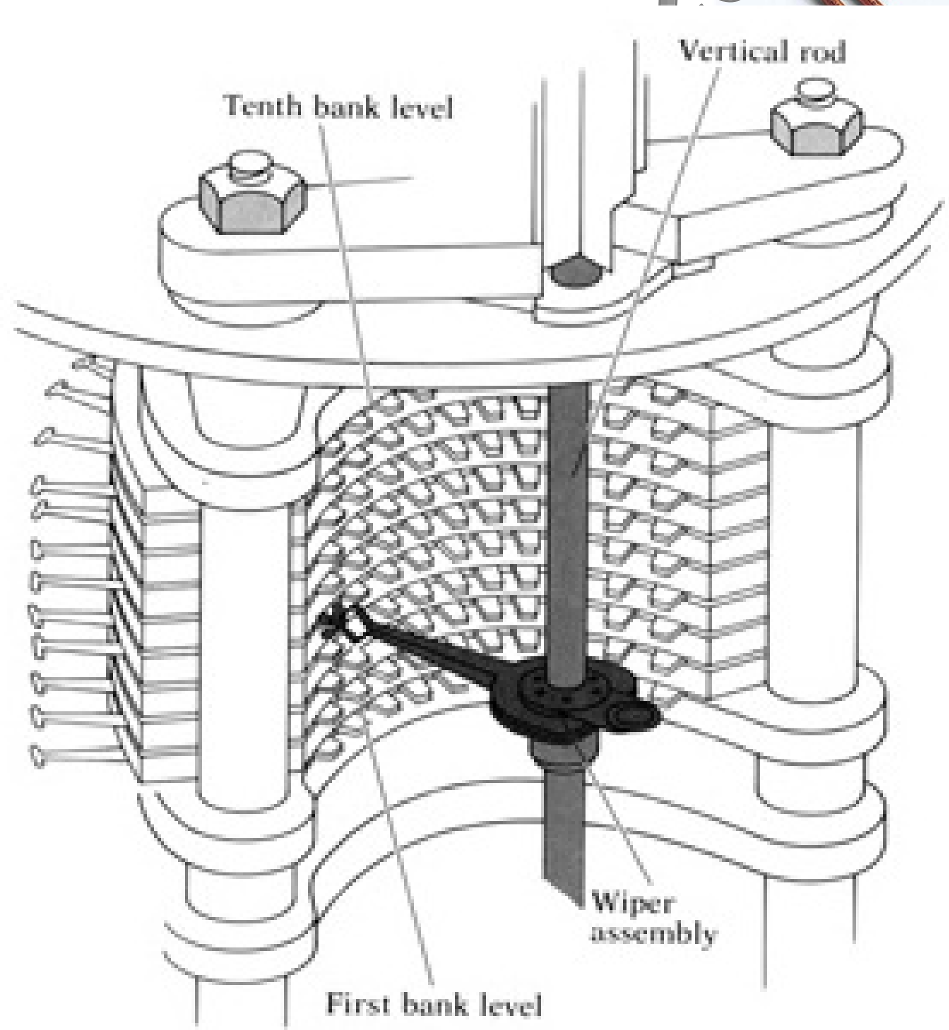
- Historically, the first mechanical telephone switch was invented by Almon Brown Strowger in about 1892.
- Strowger type switches were used in the network for a long time – the last step-by-step office in the US was not replaced until 1999.
- The first electronic switches entered the network about 1976.

Circuit Switching for Voice Calls

- A switch is a device which connects signals from one port to another port.
- Here, I'm going to describe switching of voice channels in the digital telephone system.
- I'll start by describing several of the switching technologies, then look at how they are implemented in a switch today.
- Then we'll look at switches in the network and a bit at how they're controlled with Signaling System 7 (SS7).

Strowger Switch

- The first automatic switching mechanism was the Strowger switch, invented about 1888.
- It survived in the US network until 1999 (La Crosse, Indiana).
- These switches only handle analog traffic.



Strowger Switch

- Rotary dialing interrupted the loop current, producing pulses on the line.
- The pulses caused the Strowger switch to step to a position that reflected the digit dialed.
- After that digit was latched in, the next digit stepped the next Strowger switch.
- By the time all the digits were dialed, the call was connected to the calling party.
- The switches were noisy – you had difficulty talking and being understood in a central office.
- The racks of switches were also BIG. Central offices had to be large to house all the equipment. Most have a lot of empty space now.

Strowger Switch

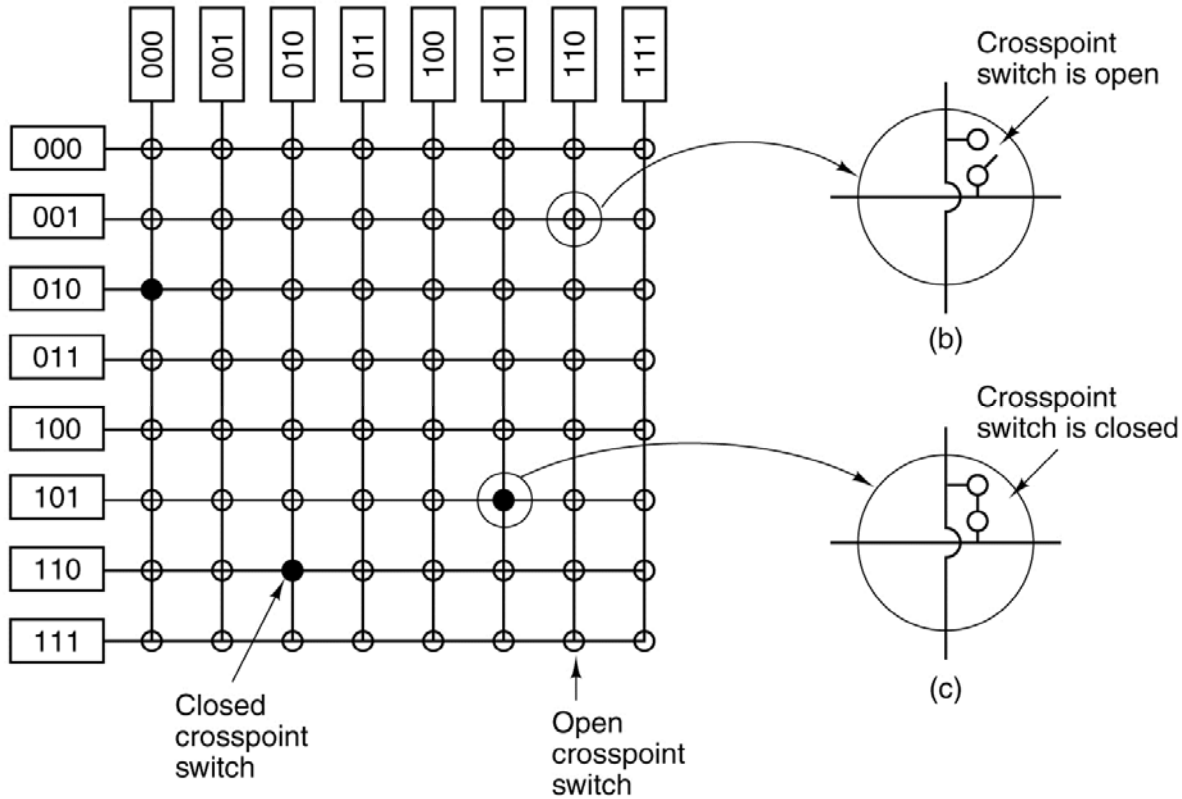
- **Just FYI, here's a picture of Strowger switches, with covers on, rack mounted in a central office.**
- **These switches were used for local calls only. Operators handled set up of long distance calls until the 1940's.**

Crossbar Switch

- The mechanical crossbar switch was first patented in 1913 by Bell Labs, but was developed by Ericsson in Sweden.
- Ericsson installed the first crossbar switch in 1926, while AT&T did not install their first crossbar switch until 1935 – after seeing the improvements made by Ericsson.

Crossbar Switch

- The mechanical crossbar switch had contacts at the intersection of the vertical and horizontal bars of the switch.
- For an *m* by *n* fabric there were *m* x *n* crosspoints.
- For example, for a 1,000 by 1,000 crossbar switch there would be a million crosspoints.

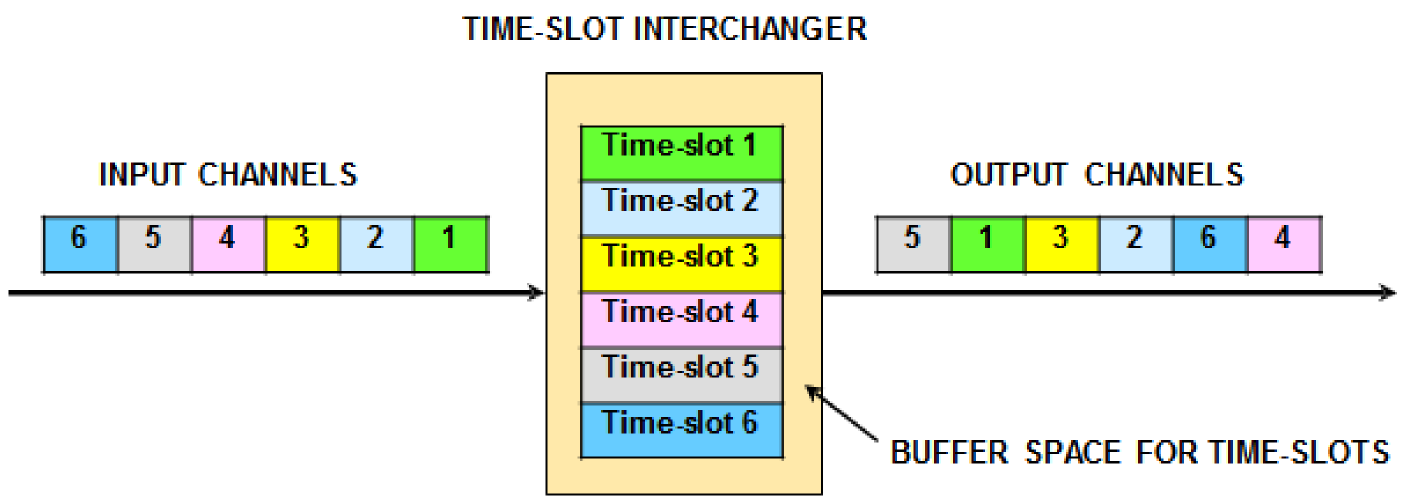


Crossbar Switches

- **Once semiconductor technology was available, the crosspoints were semiconductor switches.**
- **Crossbar switches are know as "space" switches.**

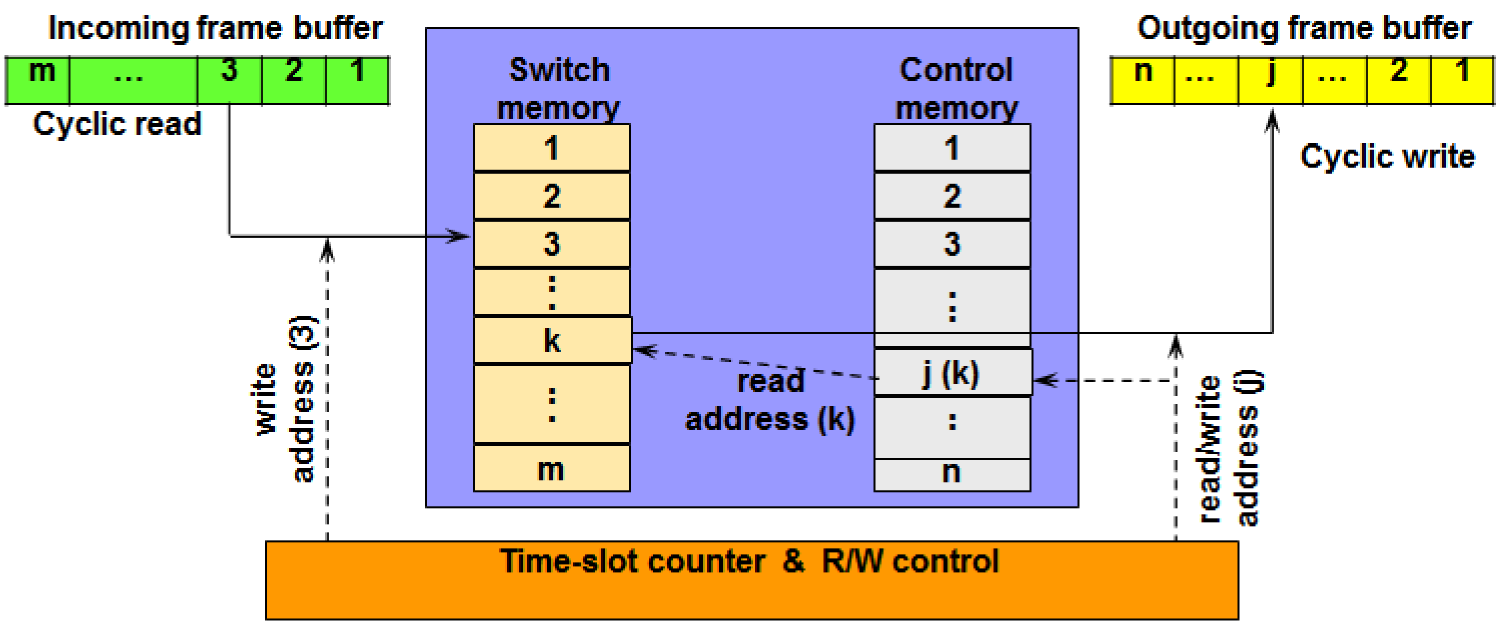
Time Slot Interchanger (TSI)

- The time slot interchanger switch works only in the digital domain.
- It’s fed by a digital line, let’s say a T1 line for this example.
- The 24 bytes of voice are read and stored into memory, then read out to an outgoing T1 with the voice channels in different time slots.
- Requires accurately synchronized input and output.
- TSI switches are know as “time” switches.



Time Slot Interchanger

- Incoming time-slots are written cyclically into switch memory.
- Output logic cyclically reads control memory, which contains a pointer for each output time-slot.
- Pointer indicates which input time-slot to insert into each output time-slot.



Limitations on the TSI switch

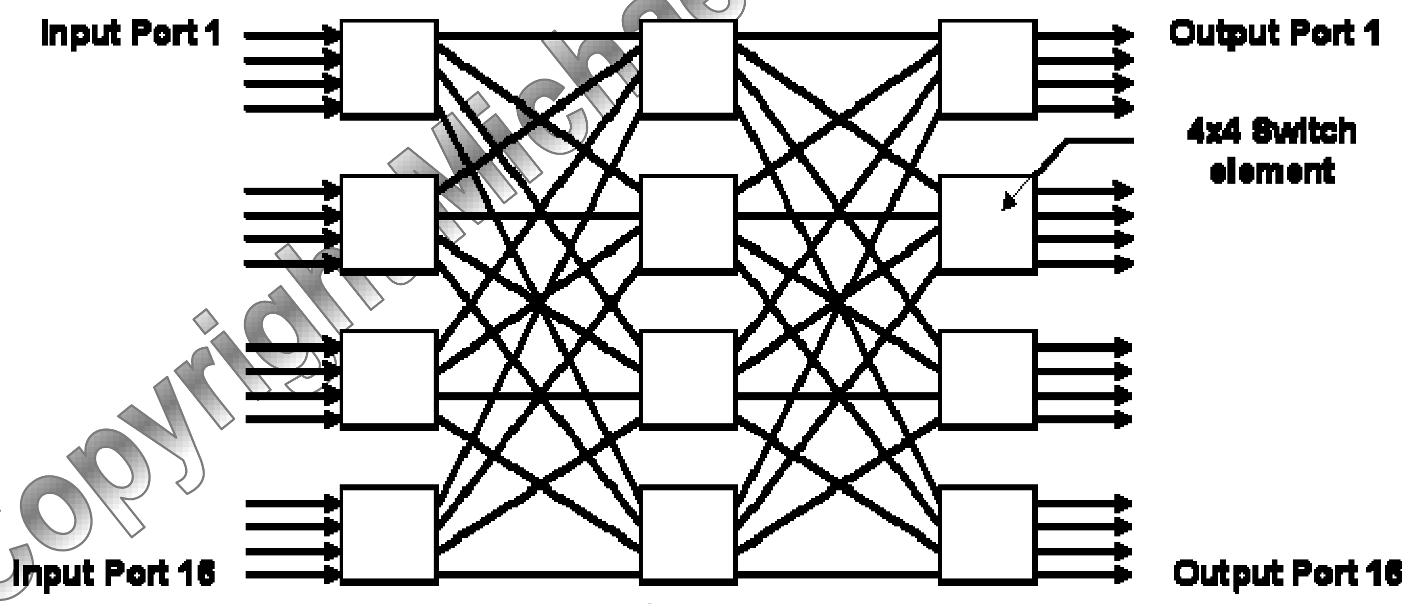
- **The steps in the TSI process are:**
  – Speech samples read in and placed in memory.
  – Processor reads control memory to determine where the data for an output time slot is located.
  – That data in switch memory is read and placed in the output frame.
- **Eventually, as the number of inputs increase, the memory and processor speeds are not fast enough to serve the incoming data.**
- **TSI switches are, therefore, built as modules and put together for form larger switches.**

Switch Fabric

- **Switches are usually built up from smaller, modular, units for several reasons:**
  - A switch fabric can be made with less crosspoints by building a multi-stage switch fabric.
  - Modular switch components are less expensive than custom large switches and can be built in volume.
  - Using modular switch components allows a switch unit to be configured (sized) to the needs of the service area.
  - Modular switch components makes servicing easier. One module can be replaced without taking the whole switch our of service. Modules can be kept as spares.

Switch Fabric

- **Here's an example of a 16 by 16, three stage switch made up of 4 by 4 modules. Note that any input port can reach any output port.**
  - A single 16 by 16 switch would contain 256 crosspoints.
  - Each 4 by 4 module contains 16 cross points.
  - There are 12 modules for a total of 192 crosspoints, a 25% reduction.

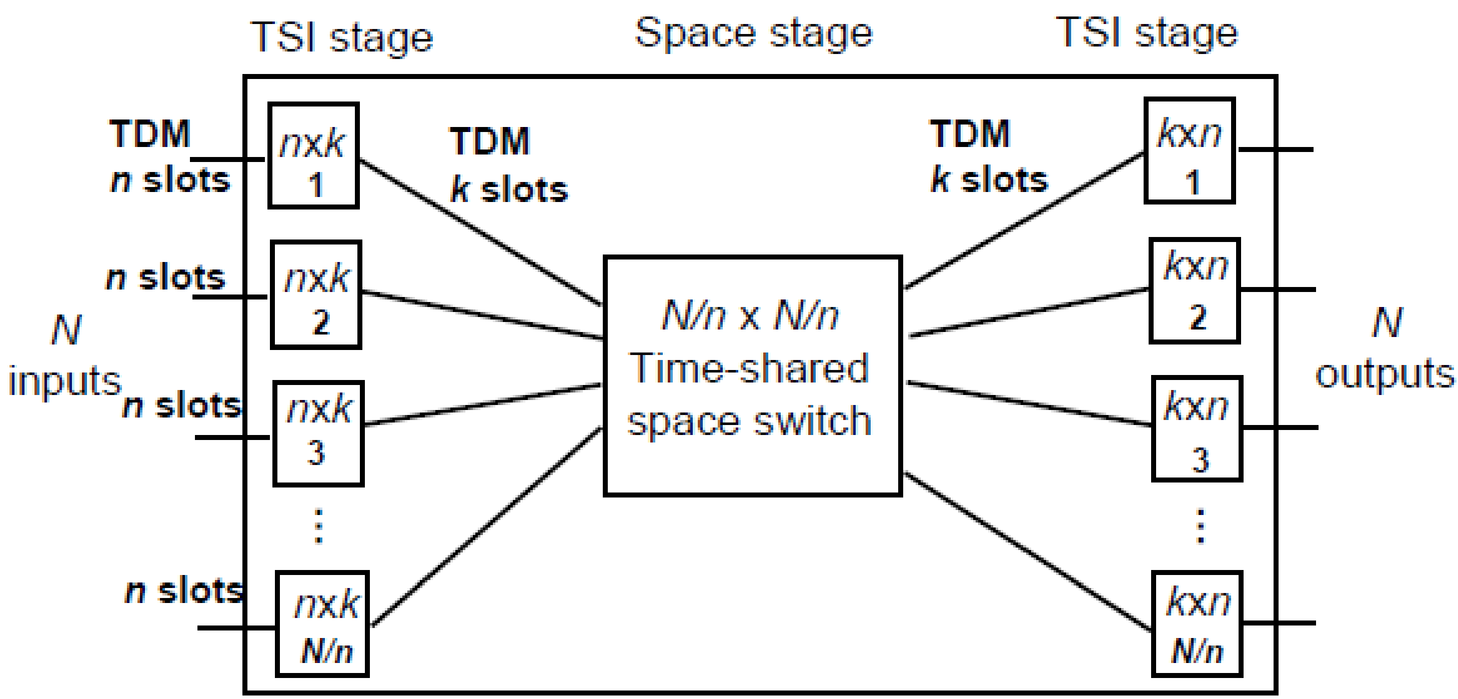


Switch Fabric

- **For a switch serving the local loops, a switch needs many ports on one side (to serve the loops) and significantly less on the other side to serve the trunks.**
  - The amount of concentration (input ports to output ports) depends on the amount of expected traffic and the blocking probability.
  - Was originally analyzed with queuing theory but is now usually done via simulation.
  - Traffic is measured in Erlangs (60 minutes of calls/hour).
- **Tandem offices, which serve trunks on both sides of the switch will have approximately the same number of ports on both sides of the switch.**

Time-Space-Time Switch

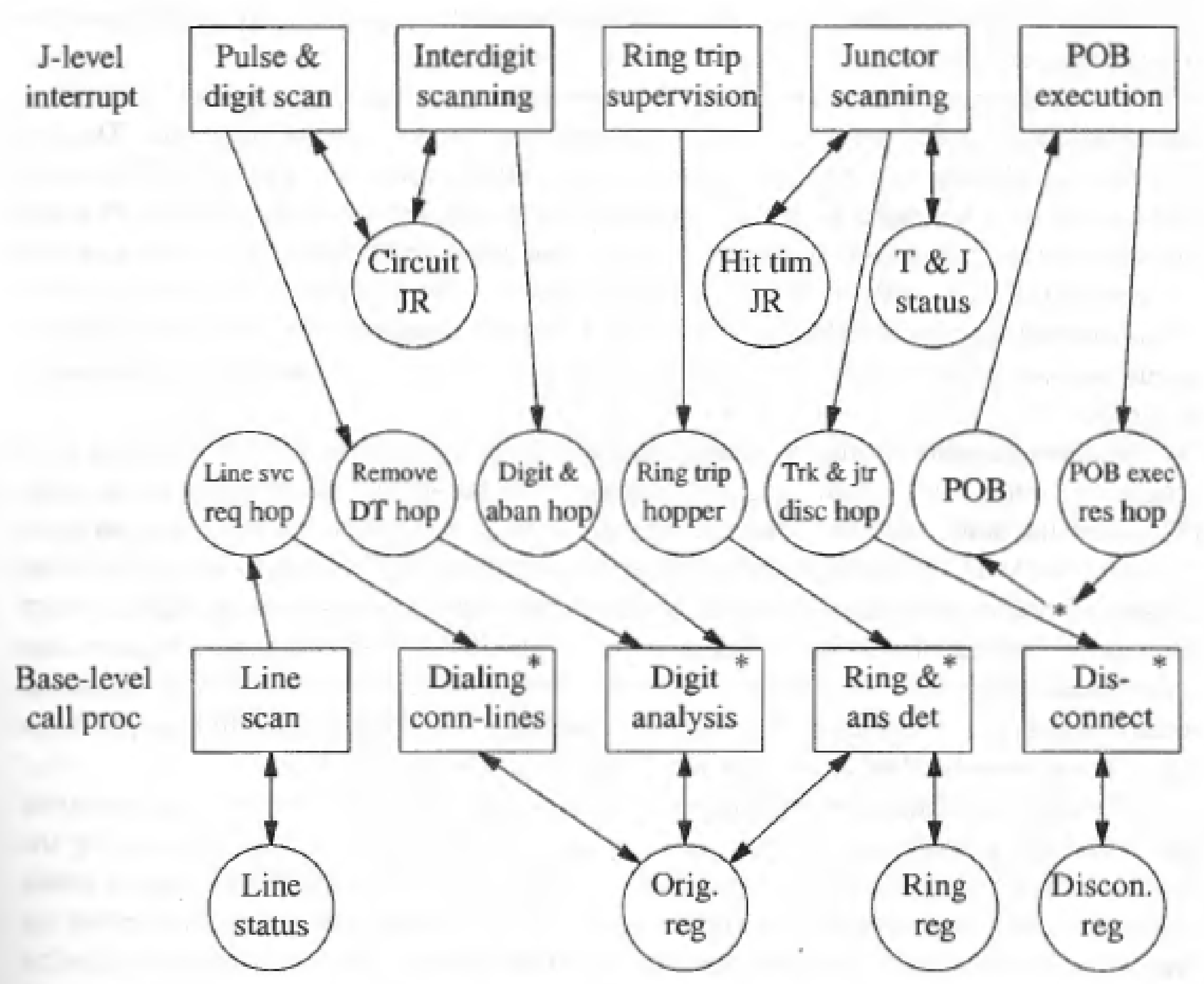
- Building a mulit-stage all-time switch is complex.
- Solution was to build a time-space-time switch.
- The space switch has semiconductor crosspoints and the interconnection pattern is reconfigured each time slot.
- Very compact design.



Switch Control

- In 1965, AT&T installed the first #1ESS switch, the first computer controlled switch.
- Selection of the route through the switch fabric was completely under the control of the computer. Route selection was computationally intensive and improved as processor speed improved.
- Over time, the computer control evolved to distributed processors under the control of a central processor.
- The control process is quite complex, involving many steps and precise timing – too much to address here.
- A block diagram of some of the processes is shown on the next slide.

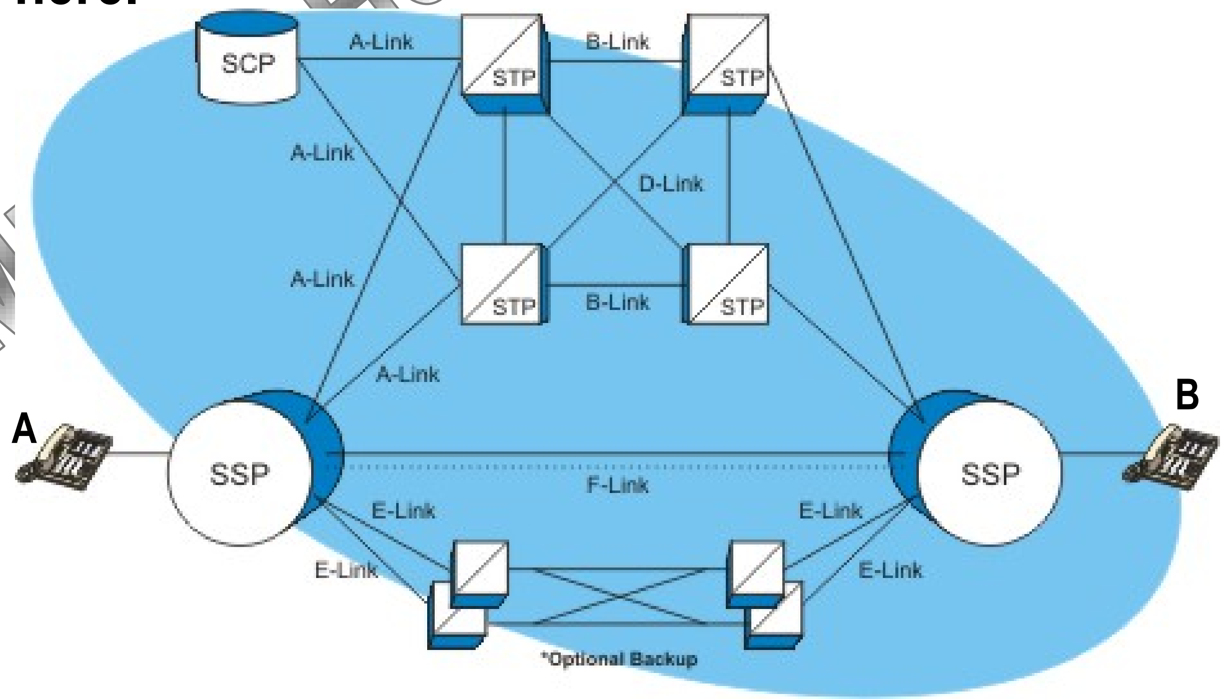
Switch Control



Signaling System 7

- **Signaling System 7 is defined in the Q.700 series of ITU recommendations and is used on essentially all telephone networks worldwide. Adopted 1975.**
- **The figure below attempts to show some of the control overlay on the network. SS7 has it's own terminology, some of which is shown here.**

SCP = Service Control Point
STP = Signal Transfer Point
SSP = Service Switching Point
A-Link = Access link
B-Link = Bridge Link
C-Link = Cross Link
D=Link = Diagonal link
E-Link = Extended Link
F-Link = Fully Associated Link



Setting up a Call

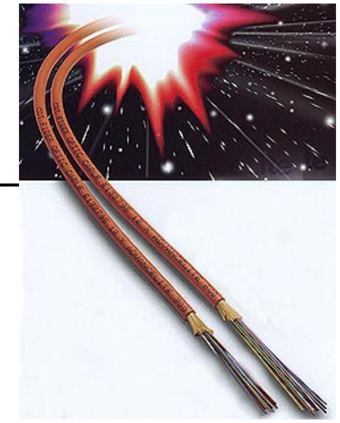
- **Looking at the previous network diagram, let's track what happens when subscriber A calls subscriber B.**
  - Switch A analyzes the dialed digits and determines that the call is intended to switch B. This is possible because of the hierarchical nature of telephone numbers.
  - Switch A selects an idle trunk between it and Switch B and then sends a SS7 message with the call information to Switch B through the SS7 network.
  - Switch B checks to see if the called phone is available (not busy) and if so, starts to build the circuit backwards towards Switch A, rings the called phone, and puts ringing tone on the backwards circuit. It also sends a SS7 message back to Switch A.

Setting up a Call (continued)

- When Switch A receives the acknowledgement message from Switch B, it completes the call towards the calling subscriber – it connects the subscriber to the trunk so the ringing tone can be heard by the calling subscriber.
- When Subscriber B picks up the phone, Switch B sends a message to Switch A to confirm the completion of the call. The lines are connected to the trunk by this time so conversation can begin.
- When one of the subscribers hangs up, that switch sends a message to the other switch and they both disconnect the lines from the trunk and put the trunk back into idle status, ready for the next call.

Special Calls

- Certain "special" numbers, such as 800, 888, 877, or 900 numbers, do not really exist as subscriber numbers.
- Those numbers must be translated to real subscriber numbers, usually a hunt group.
- This is done by the calling central office making a request to the Service Control Point (SCP) to get the real number associated with a special number.
- Once the real number is obtained, the call is routed as described earlier.

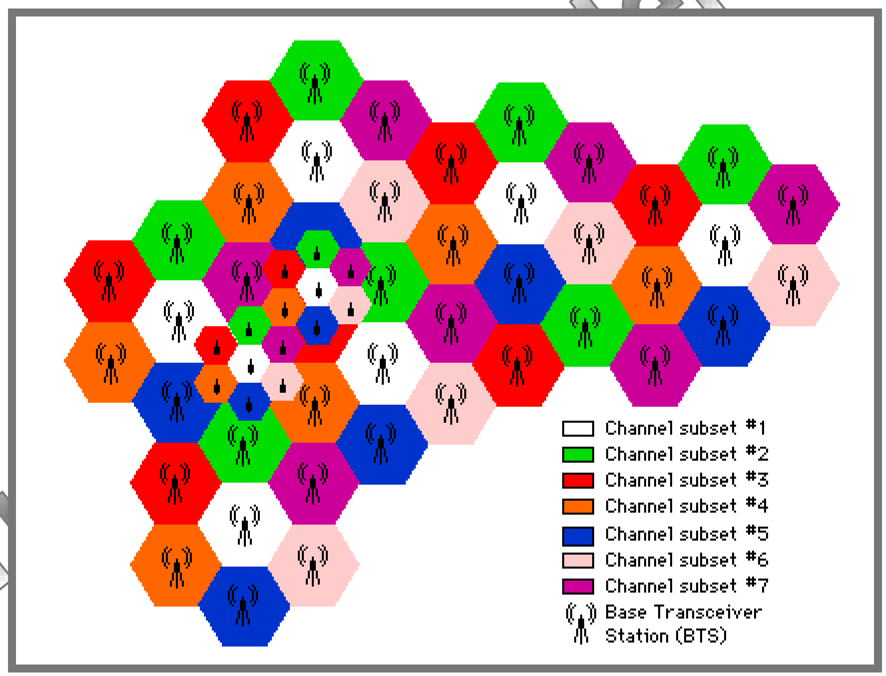


Cellular Networks

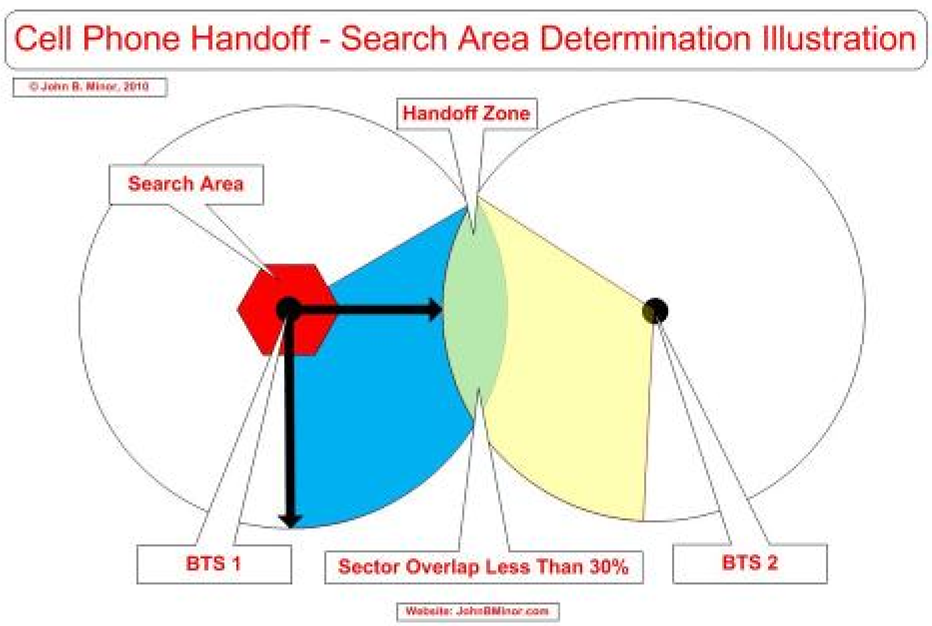
Cellular Communications

- **Cellular systems are adjunct to the PSTN. They cannot operate without the PSTN to carry the traffic.**
- **They are composed of the following components:**
  - Cell towers and cell coverage areas.
  - Mobile Telephone Switching Office (MTSO) .
  - Cellular telephones

- **ad**

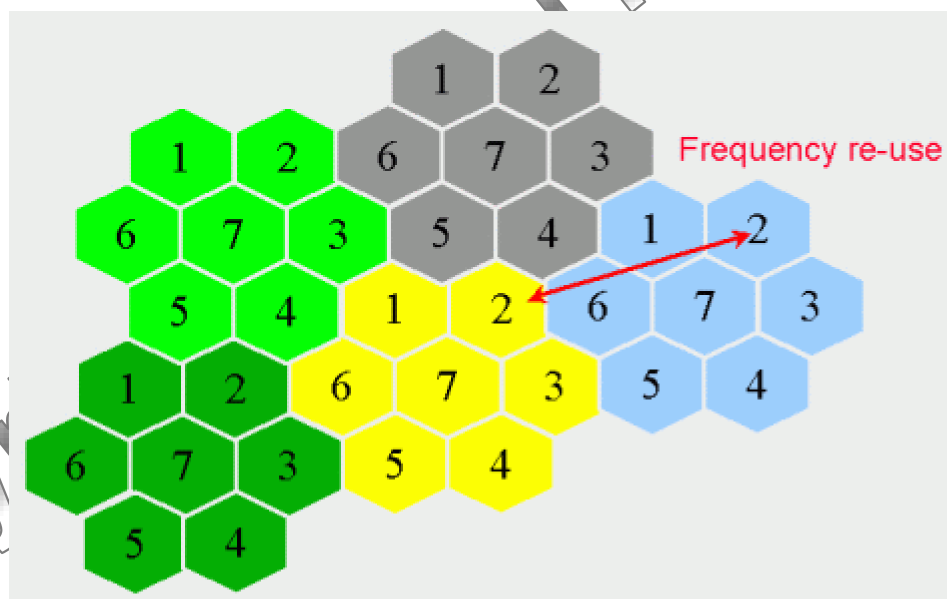

Channel subset #1
Channel subset #2
Channel subset #3
Channel subset #4
Channel subset #5
Channel subset #6
Channel subset #7
Base Transceiver Station (BTS)

* fad

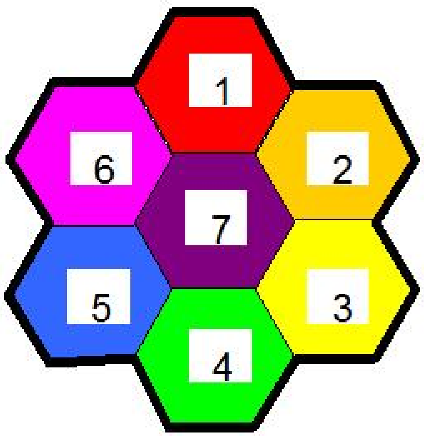


Cell Phone Handoff - Search Area Determination Illustration
© John B. Minor, 2010
Handoff Zone
Search Area
BTS 1
Sector Overlap Less Than 30%
BTS 2
Website: JohnBMinor.com

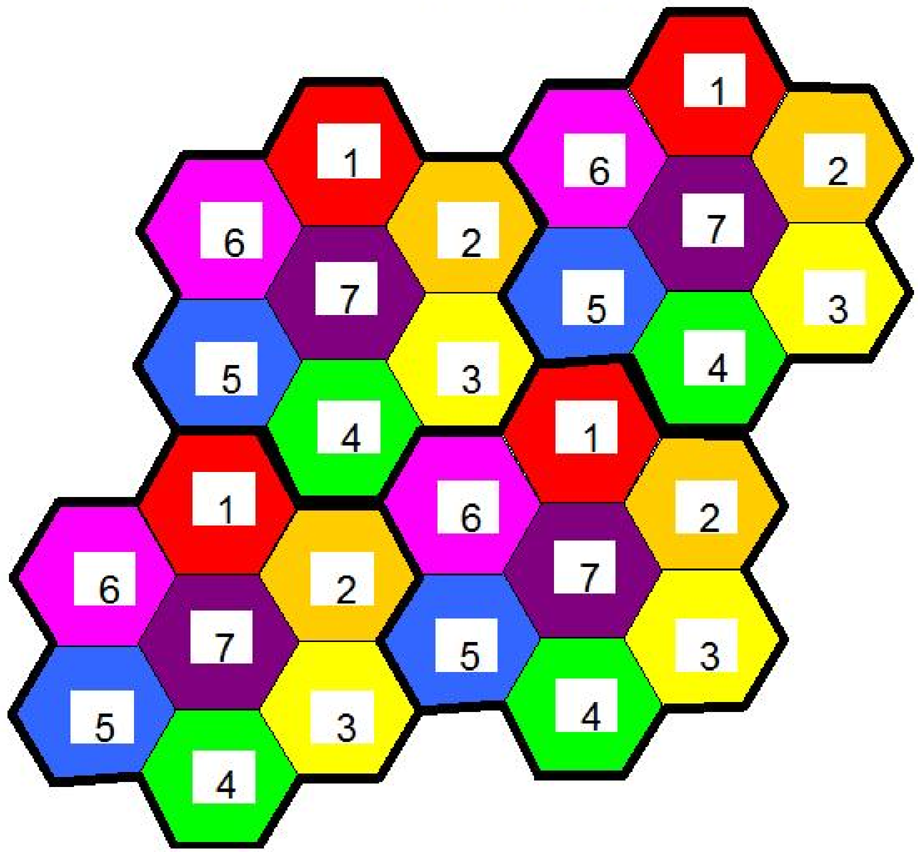
- dfad



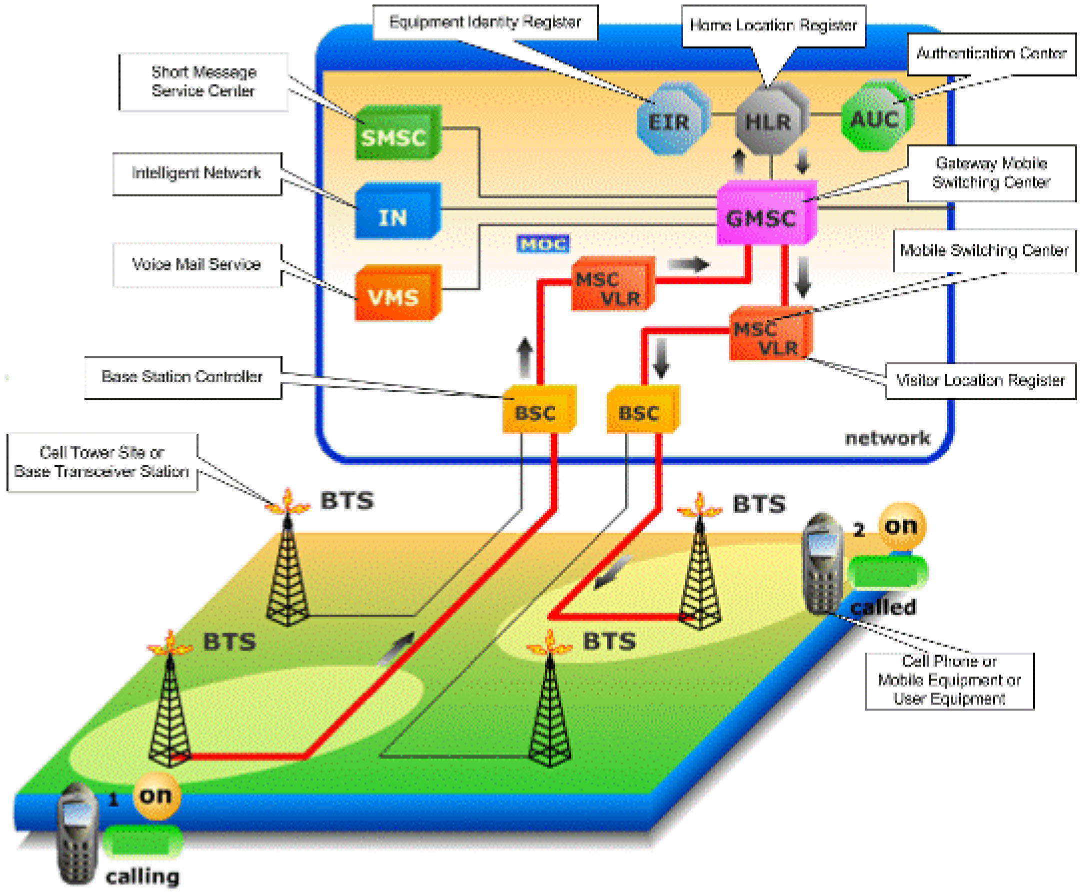
7-cell
cluster



Coverage area ‘tiled’
with 7-cell clusters

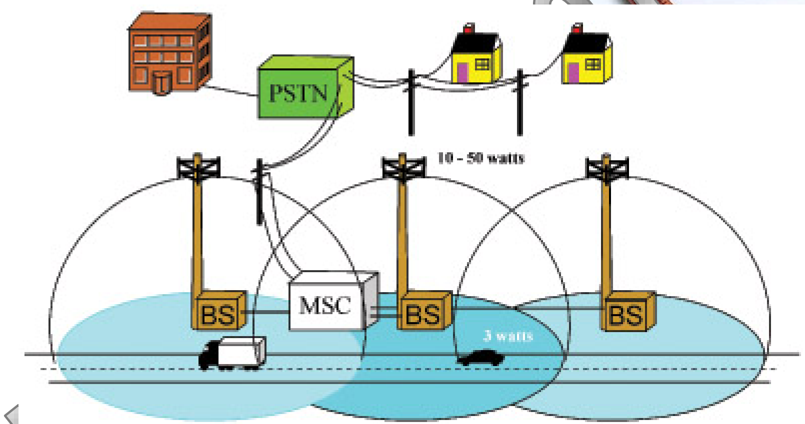


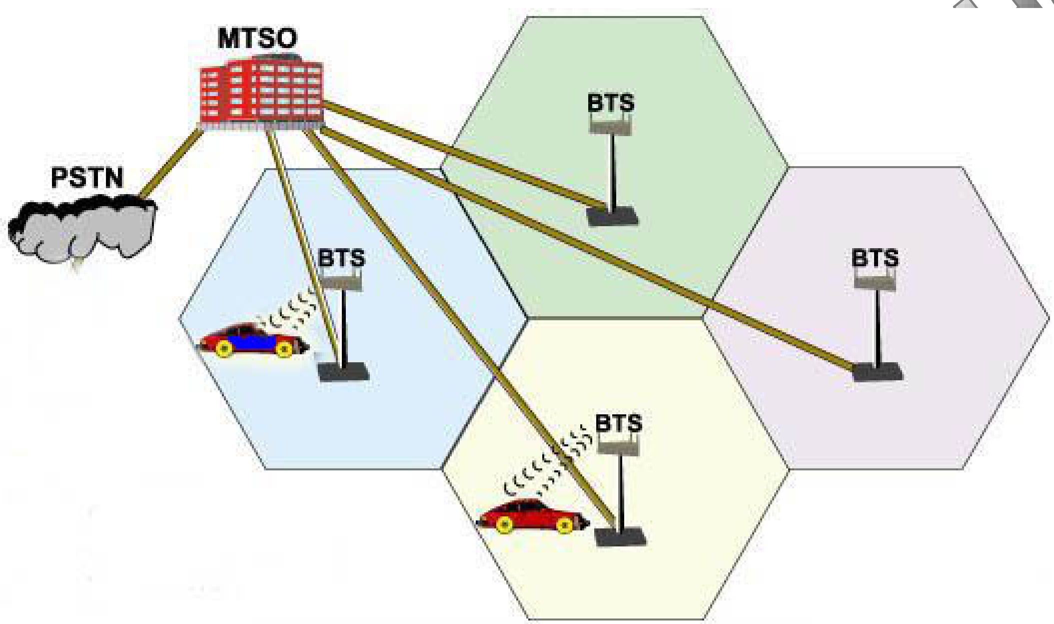
- fad

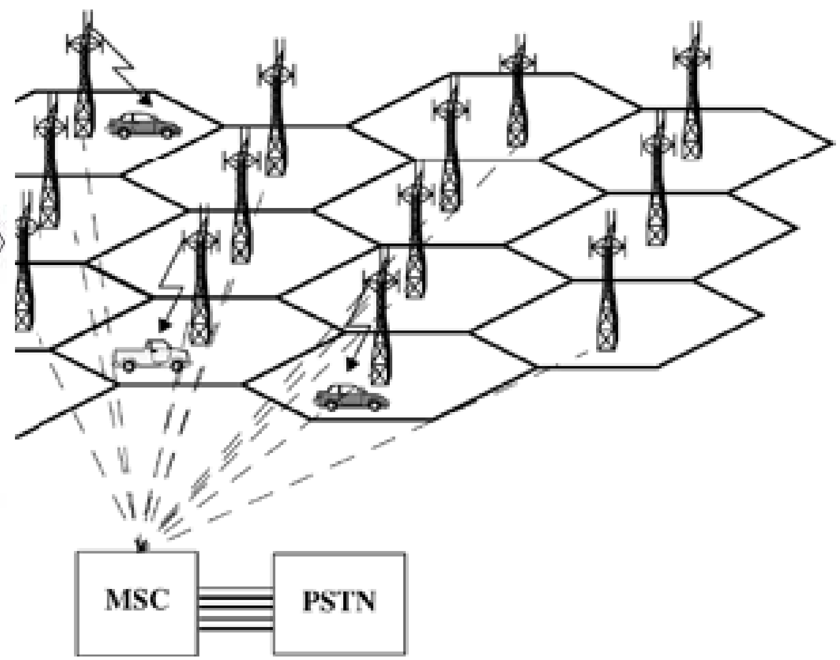


Equipment Identity Register
Home Location Register
Authentication Center
Short Message Service Center
Intelligent Network
Voice Mail Service
Gateway Mobile Switching Center
Mobile Switching Center
Visitor Location Register
Base Station Controller
Call Tower Site or Base Transceiver Station
Cell Phone or Mobile Equipment or User Equipment
SMSC
EIR
HLR
AUC
IN
GMSC
MOC
VMS
MSC VLR
MSC VLR
BSC
BSC
network
BTS
BTS
BTS
BTS
1 on
calling
2 on
called

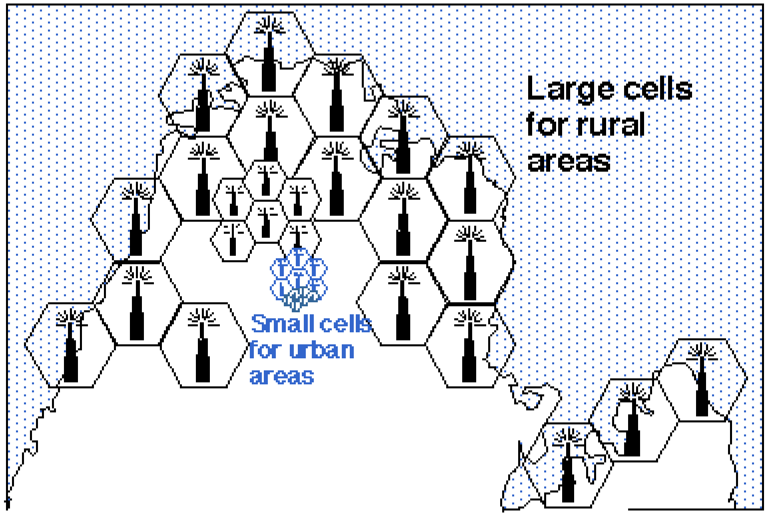
- **fad**


PSTN
10 - 50 watts
MSC
BS
BS
BS
3 watts


MTSO
PSTN
BTS
BTS
BTS
BTS


MSC
PSTN

- dafds



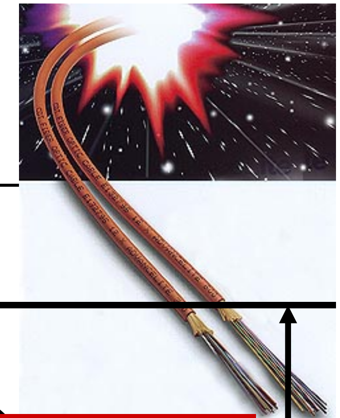
Questions?

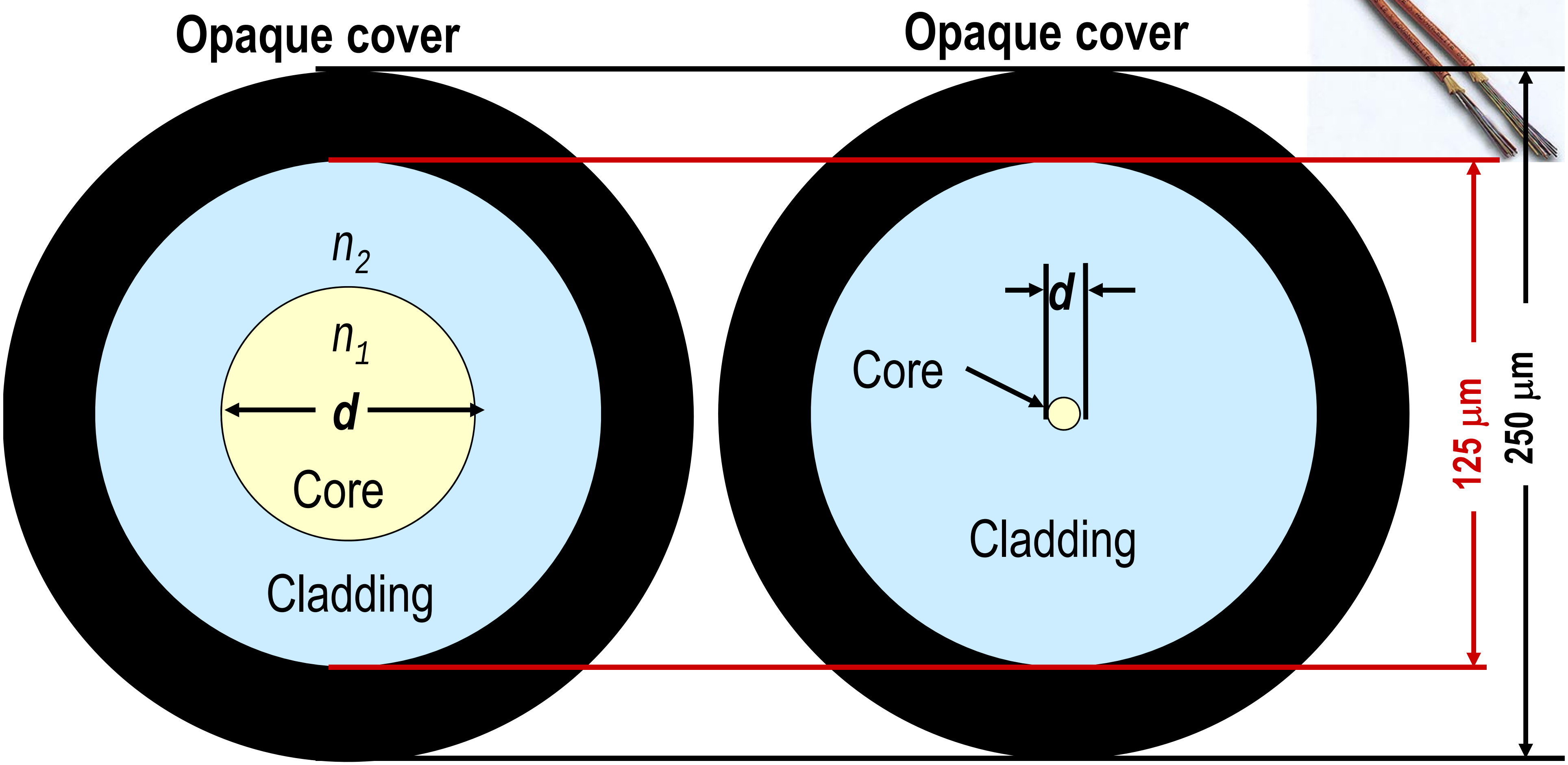

USWEST
AMERITECH
NYNEX
BELL ATLANTIC
PACBELL
SOUTHWESTERN BELL
BELLSOUTH

RBOCs Today

AT&T
Qwest
Verizon
AT&T
Cincinnati Bell
AT&T
Bell South
Alaska Power and Phone Company
Hawaiian Telecom
Washington
Oregon
California
Nevada
Idaho
Montana
Wyoming
Utah
Arizona
Colorado
New Mexico
North Dakota
South Dakota
Nebraska
Kansas
Oklahoma
Texas
Minnesota
Iowa
Missouri
Arkansas
Louisiana
Wisconsin
Illinois
Michigan
Indiana
Ohio
Kentucky
Tennessee
Mississippi
Alabama
Georgia
Florida
South Carolina
North Carolina
Virginia
West Virginia
Pennsylvania
New York
Maine
New Hampshire
Vermont
Massachusetts
Rhode Island
Connecticut
New Jersey
Delaware
Maryland
Alaska
Hawaii
Note: Many of these companies compete with other local phone companies in the regions they provide service to.

Multimode and single mode fiber





Multimode fiber

d **= 50 or 62.5 µm**

Single mode fiber

d **= ~8 µm**

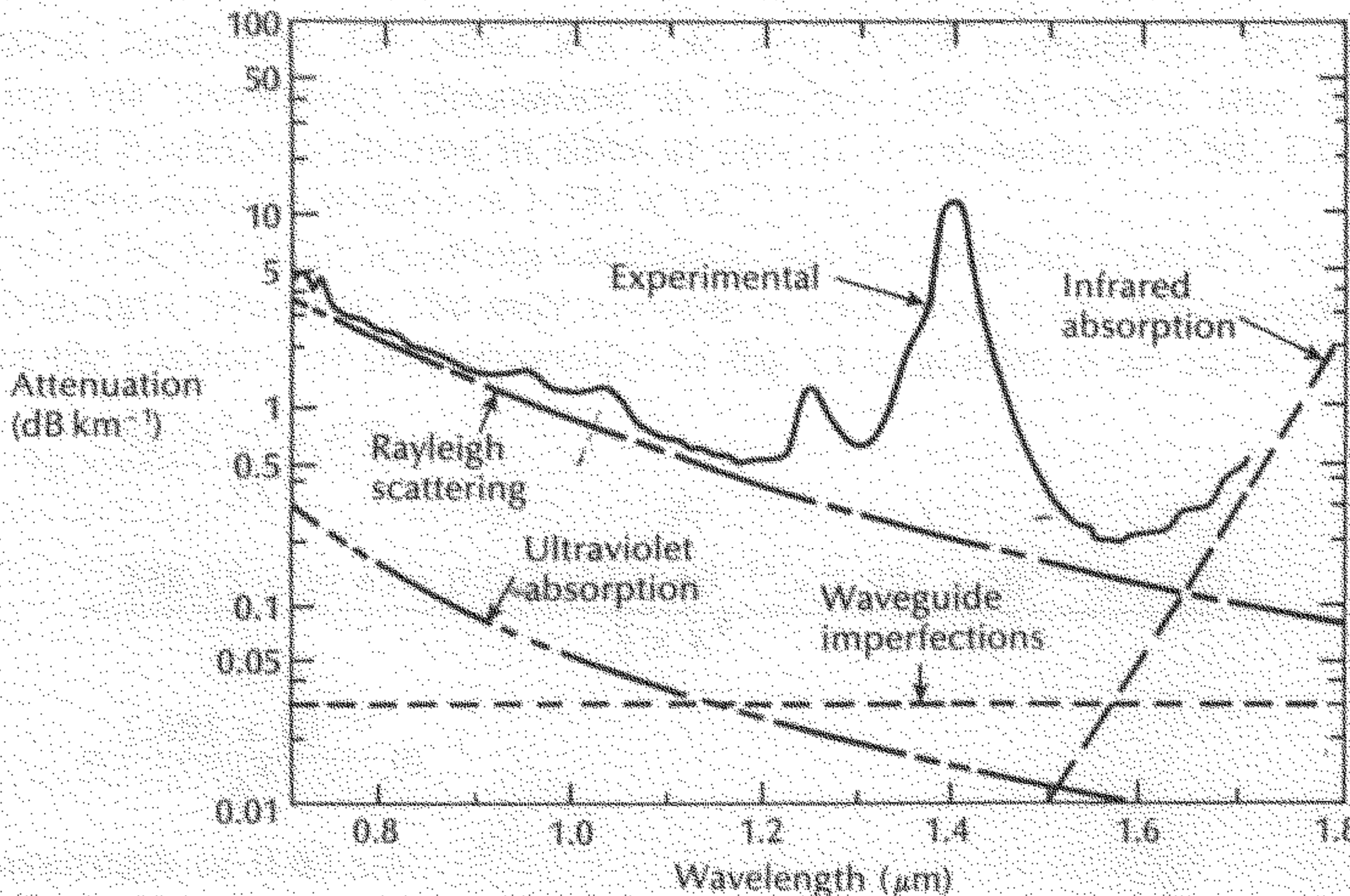
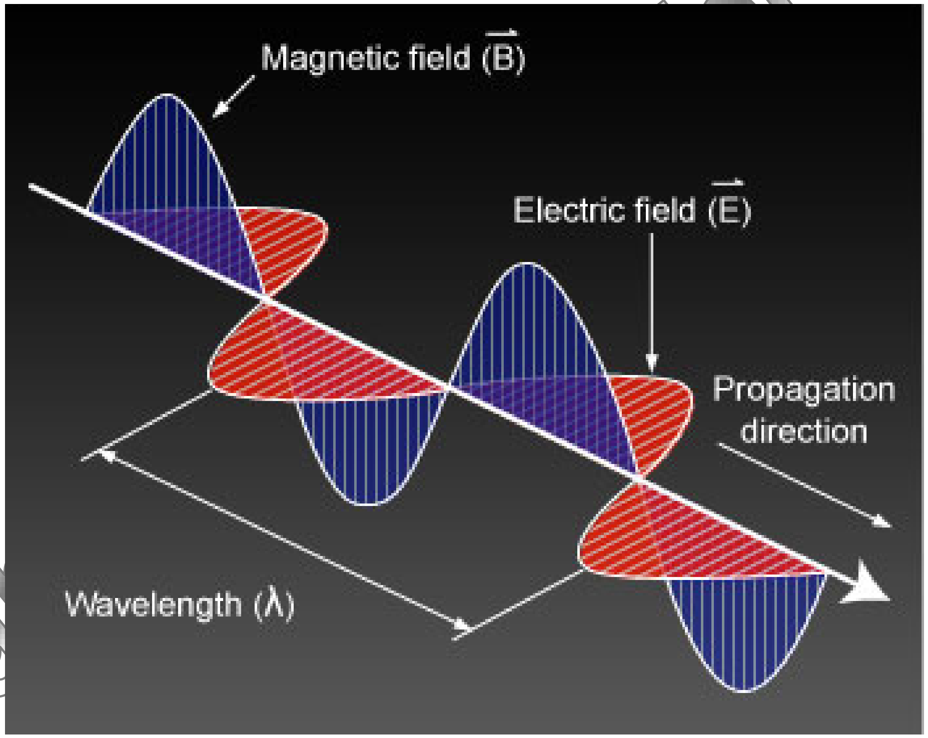
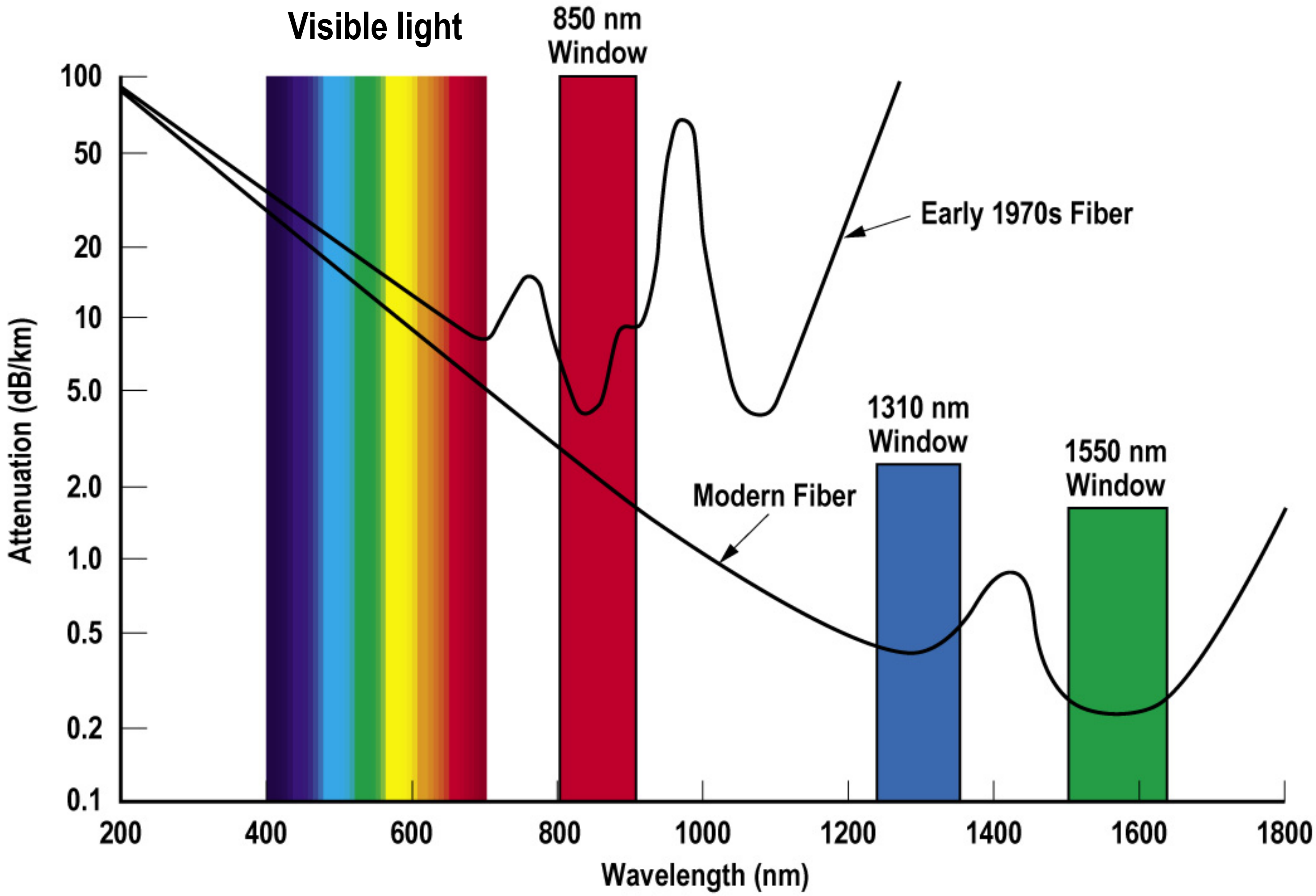
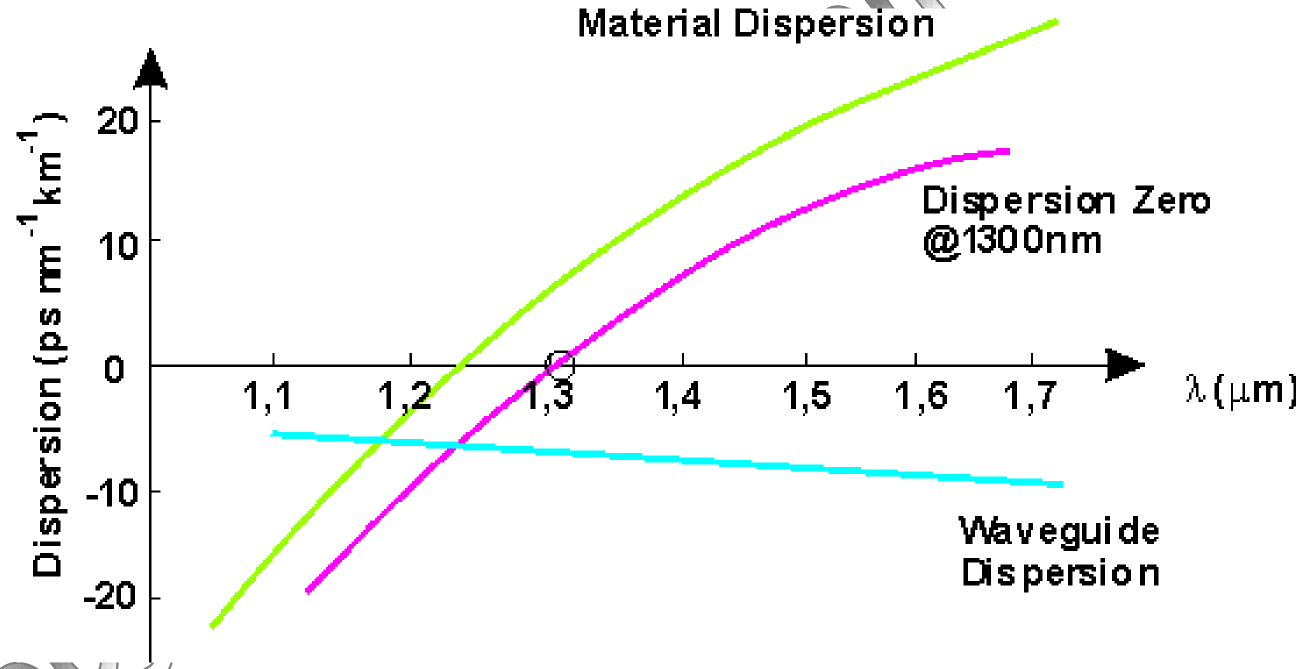


Figure 3.3 The measured attenuation spectrum for an ultra-low-loss single-mode fiber (solid line) with the calculated attenuation spectra for some of the loss mechanisms contributing to the overall fiber attenuation (dashed and dotted lines) [Ref. 3].

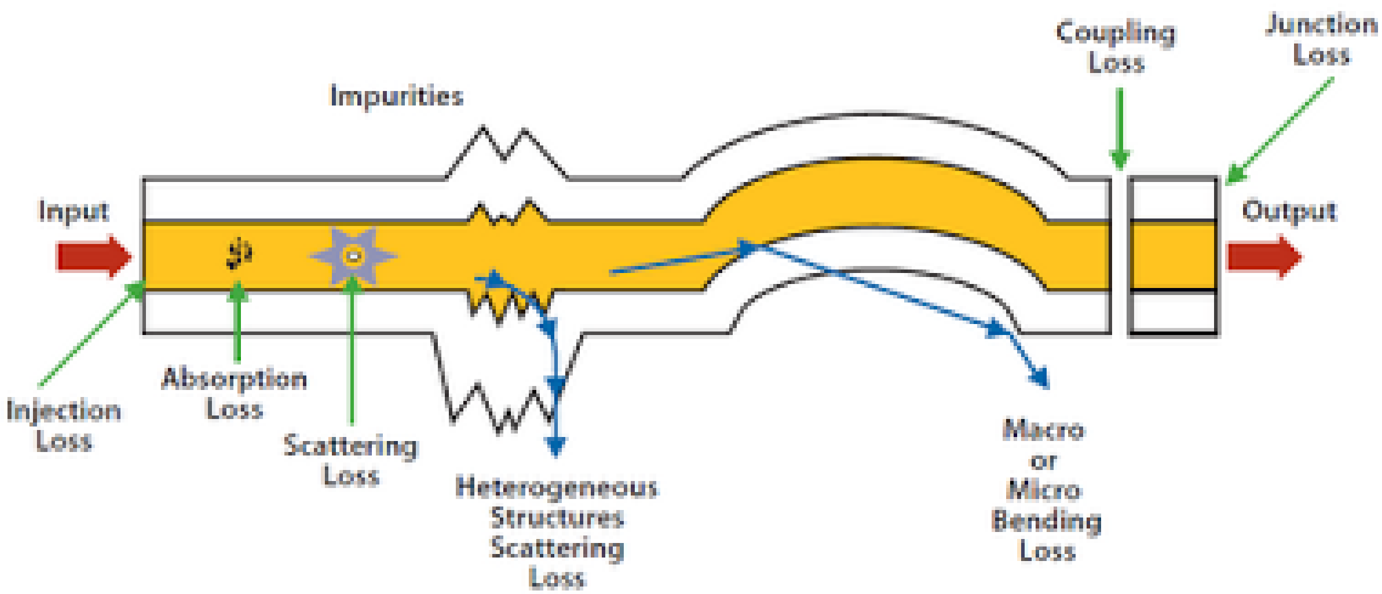

Magnetic field (B)
Electric field (E)
Propagation direction
Wavelength (λ)

Optical Spectrum




Material Dispersion
Dispersion Zero
@1300nm
Waveguide
Dispersion
Dispersion (ps nm-1 km-1)
20
10
0
-10
-20
1,1
1,2
1,3
1,4
1,5
1,6
1,7
λ (μm)

- adasd

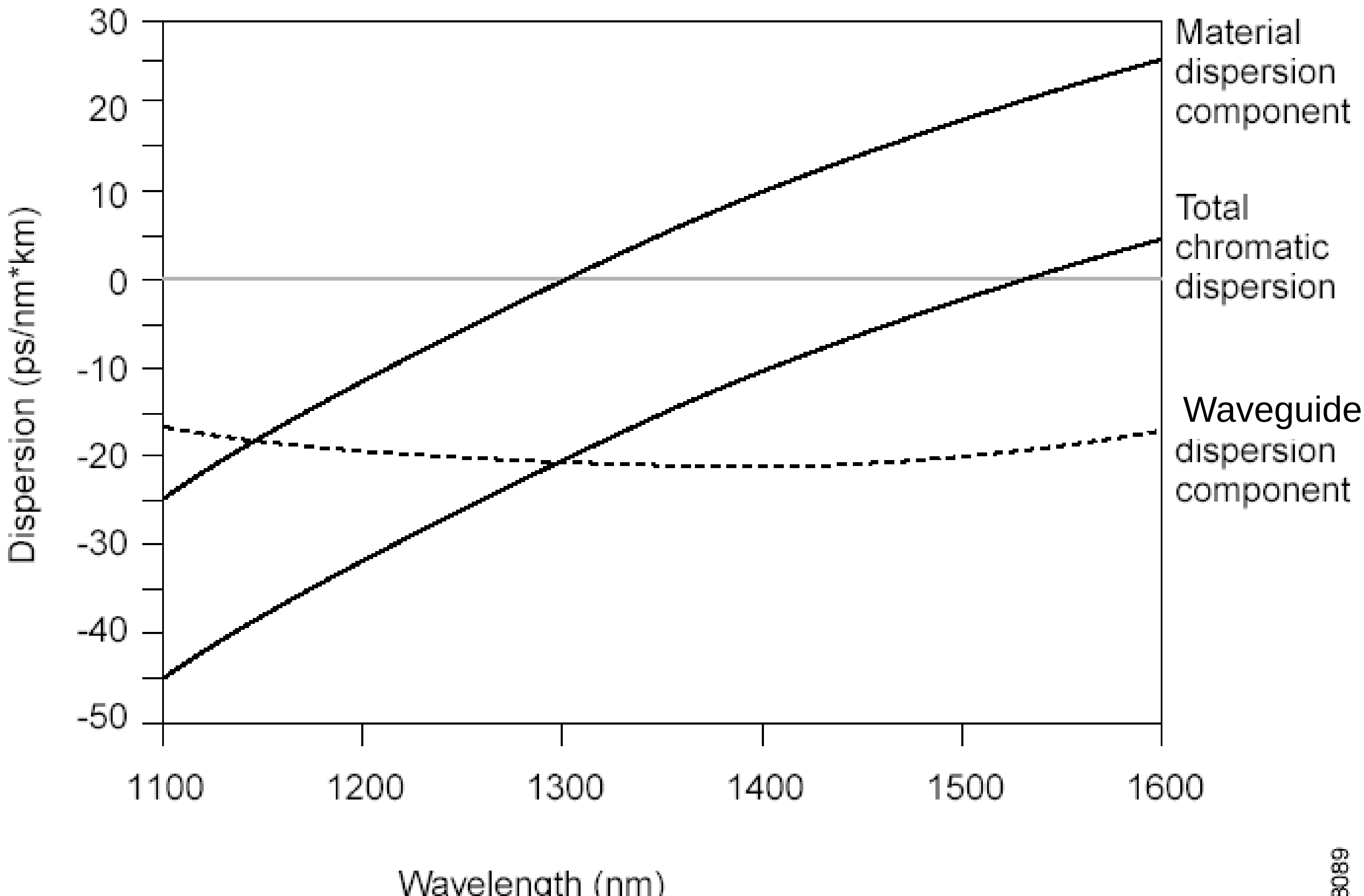


Link loss mechanisms

Zero Dispersion Point

- G.652 Fiber is said to have a zero dispersion point in the 1310 nm band. What does this mean?
- Material Dispersion causes the pulse to spread in the positive direction (positive ps/nm-km).
- Waveguide Dispersion causes the pulse to spread in the negative direction (negative ps/nm-km).
- The frequency (or wavelength) where they cancel each other is the zero dispersion point.
- Can be modified by doping of the fiber, producing Dispersion Shifted Fiber.

Zero Dispersion Point (here about 1550nm)



- **fdgfd**

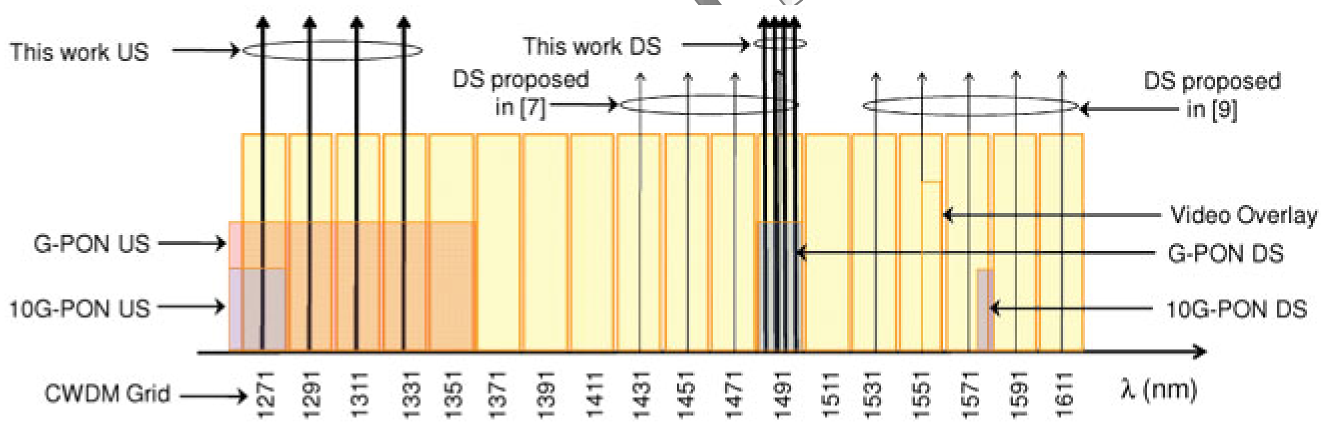
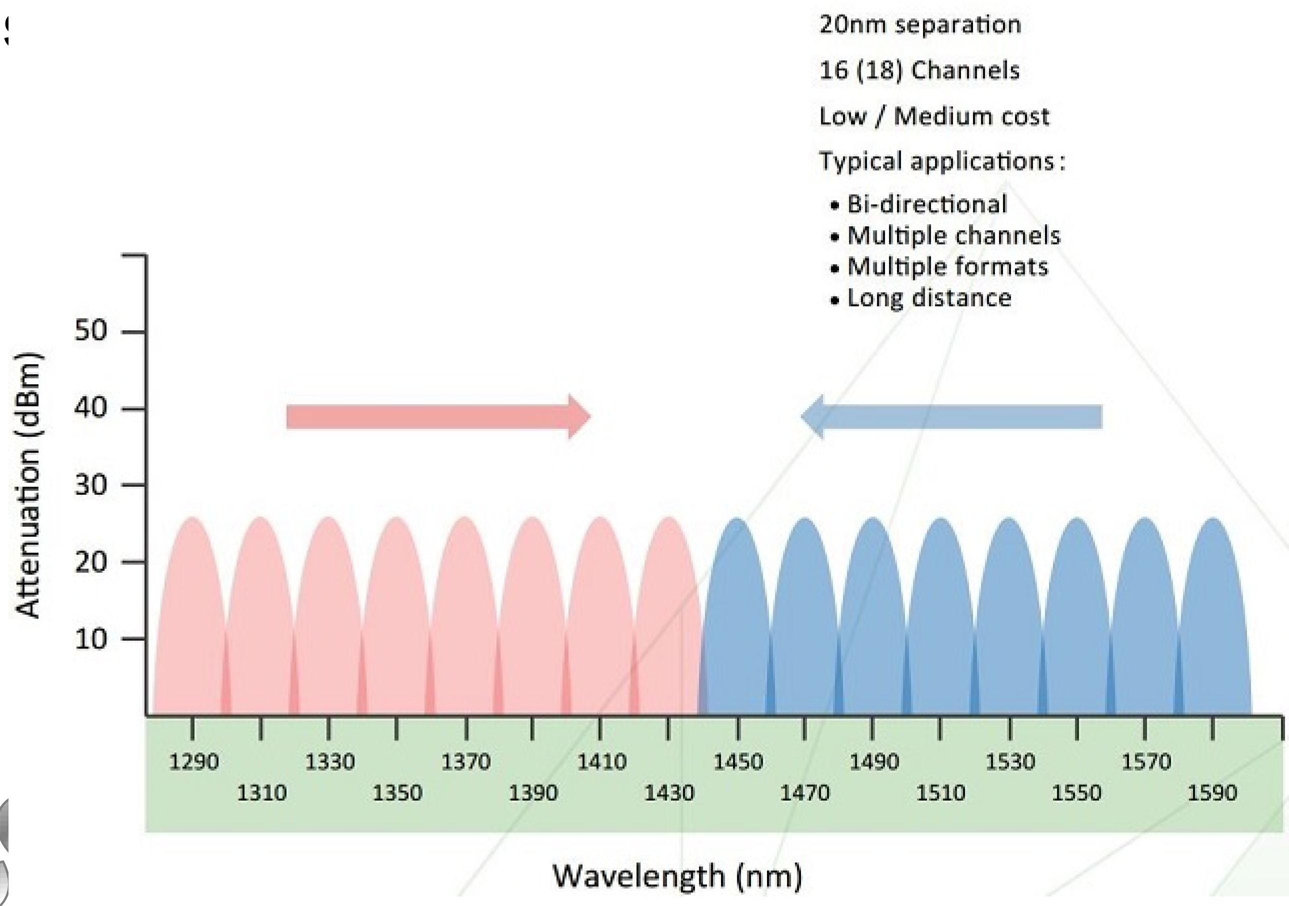


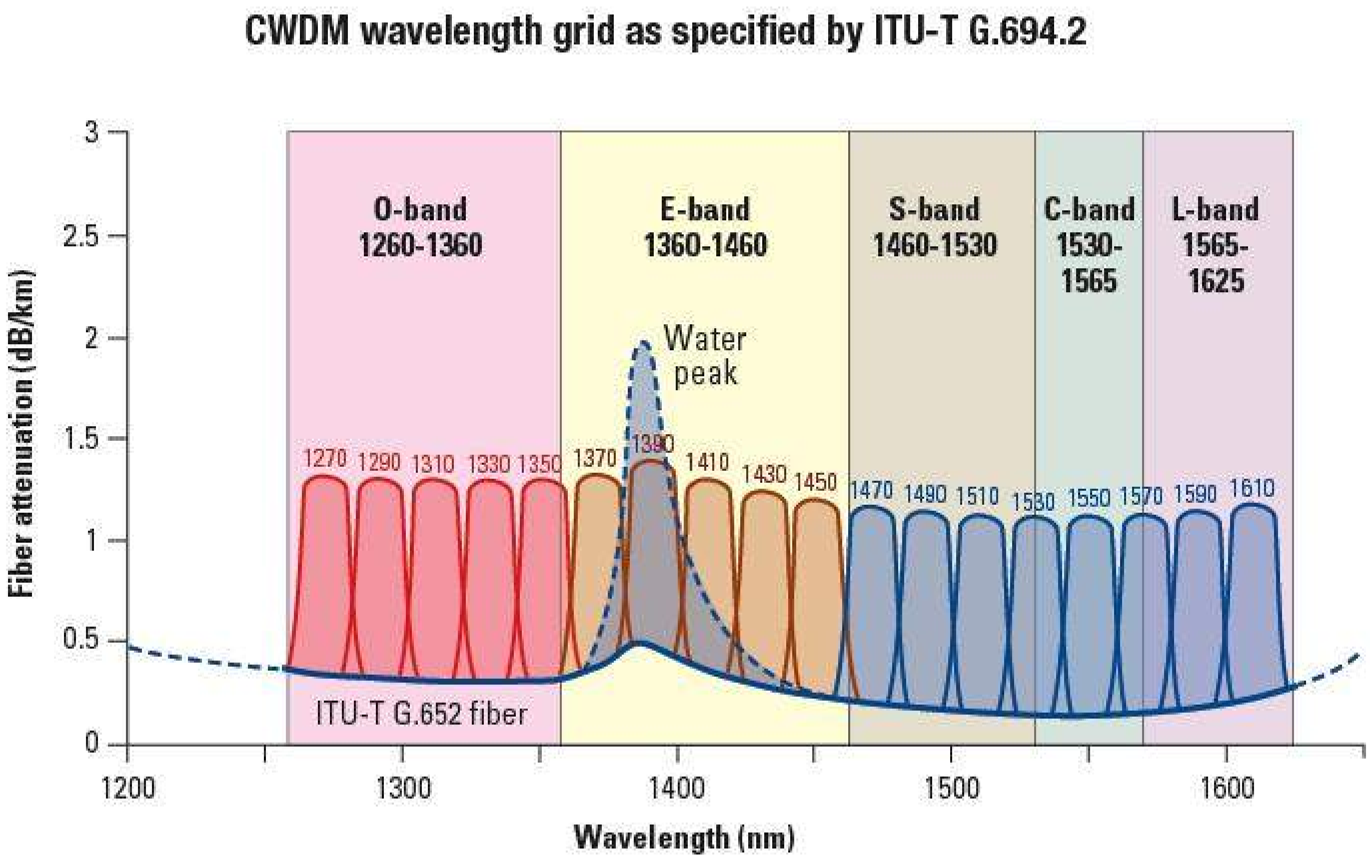
Fig. 2. Available spectrum grid for a WDM PON overlay.

- d

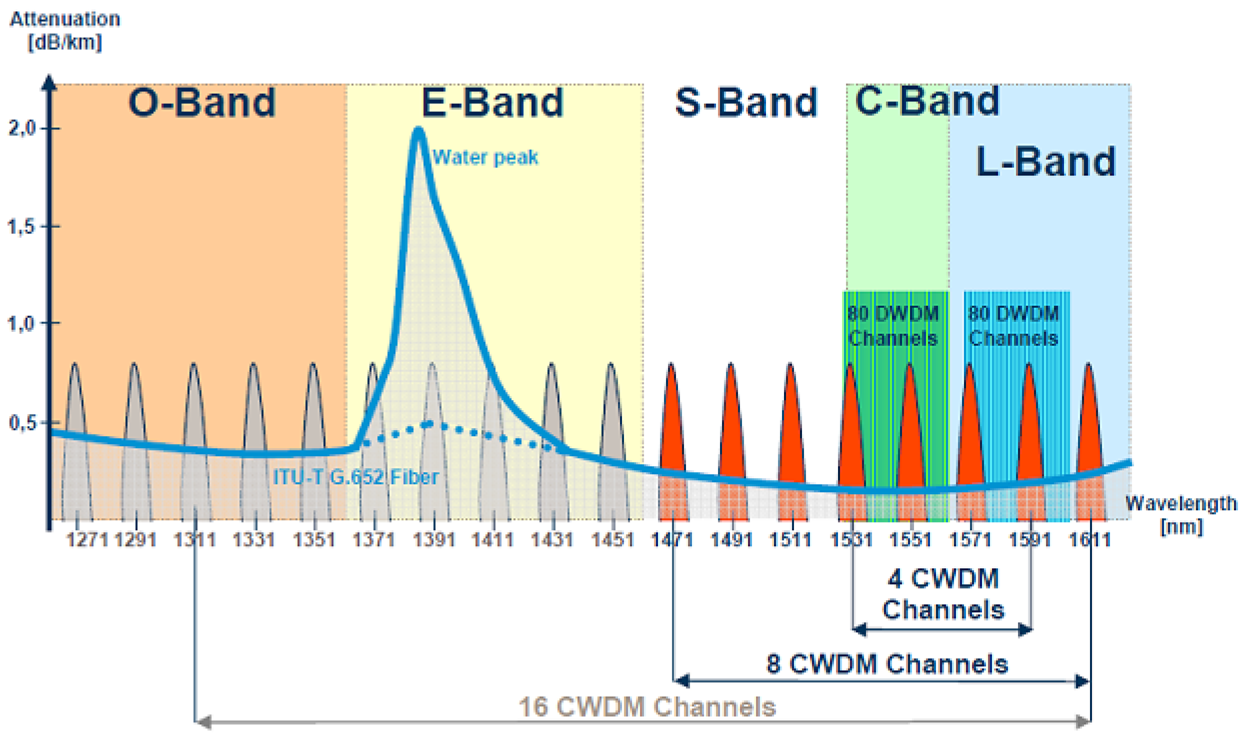


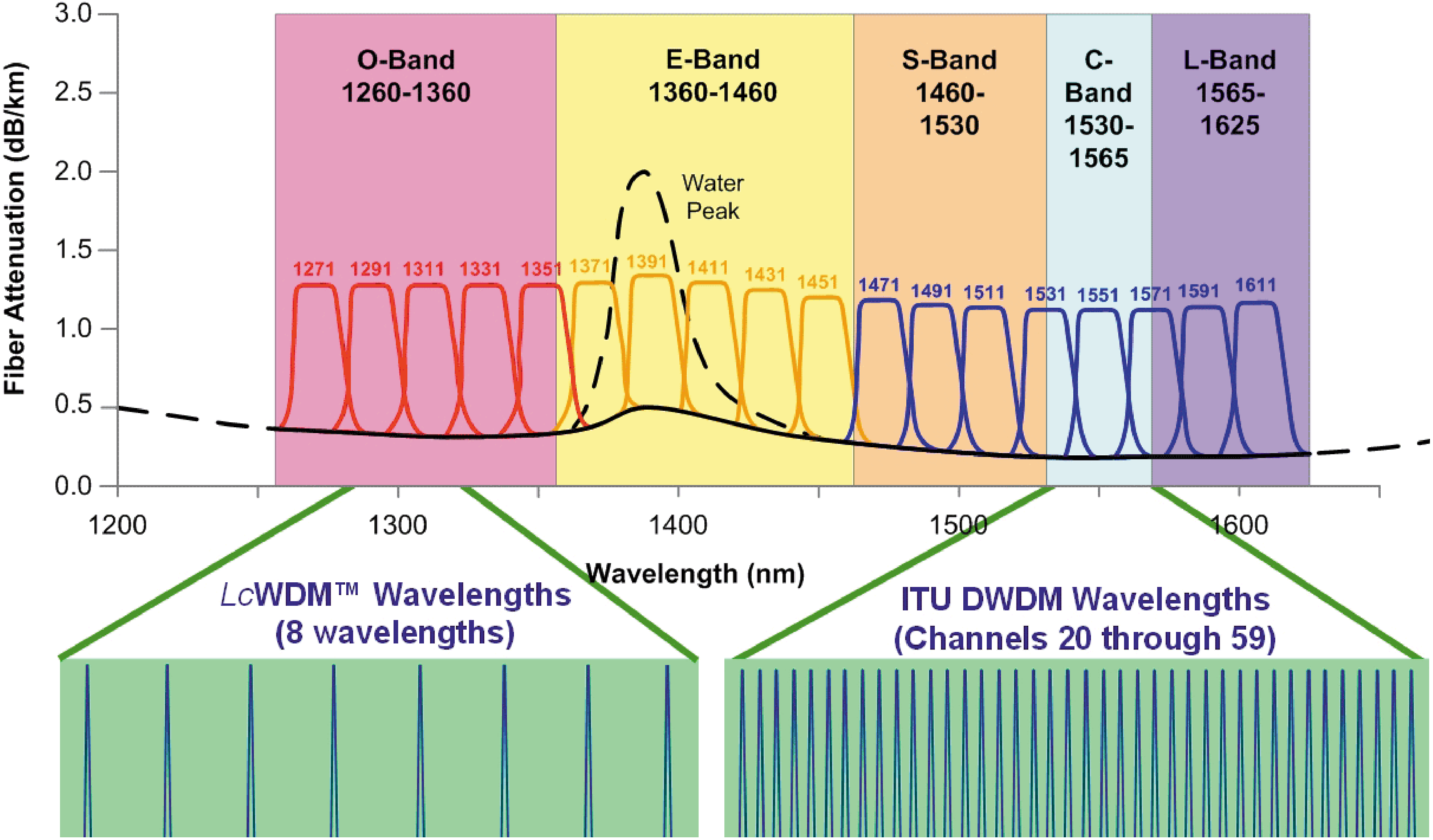
20nm separation
16 (18) Channels
Low / Medium cost
Typical applications:
• Bi-directional
• Multiple channels
• Multiple formats
• Long distance
Attenuation (dBm)
50
40
30
20
10
1290
1310
1330
1350
1370
1390
1410
1430
1450
1470
1490
1510
1530
1550
1570
1590
Wavelength (nm)

- fsdads


CWDM wavelength grid as specified by ITU-T G.694.2
O-band
1260-1360
E-band
1360-1460
S-band
1460-1530
C-band
1530-
1565
L-band
1565-
1625
Water
peak
1270 1290 1310 1330 1350 1370 1390 1410 1430 1450 1470 1490 1510 1530 1550 1570 1590 1610
ITU-T G.652 fiber
Fiber attenuation (dB/km)
0 0.5 1 1.5 2 2.5 3
1200 1300 1400 1500 1600
Wavelength (nm)

- **dfsSD**

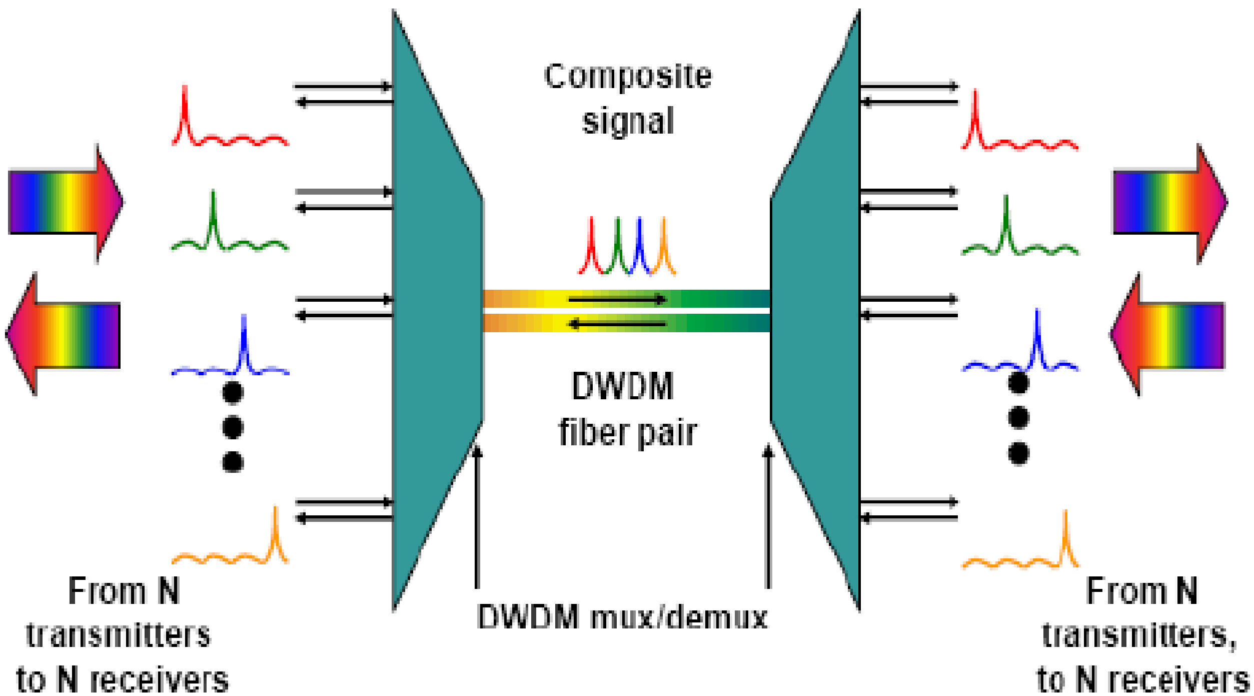

Attenuation [dB/km]
2,0
1,5
1,0
0,5
O-Band
E-Band
S-Band
C-Band
L-Band
Water peak
80 DWDM Channels
80 DWDM Channels
ITU-T G.652 Fiber
Wavelength [nm]
1271 1291 1311 1331 1351 1371 1391 1411 1431 1451 1471 1491 1511 1531 1551 1571 1591 1611
4 CWDM Channels
8 CWDM Channels
16 CWDM Channels



Fiber Attenuation (dB/km)
3.0
2.5
2.0
1.5
1.0
0.5
0.0
1200
1300
1400
1500
1600
Wavelength (nm)
O-Band
1260-1360
E-Band
1360-1460
S-Band
1460-
1530
C-
Band
1530-
1565
L-Band
1565-
1625
Water
Peak
1271 1291 1311 1331 1351
1371 1391 1411 1431 1451
1471 1491 1511 1531 1551 1571 1591 1611
*Lc*WDM™ Wavelengths
(8 wavelengths)
ITU DWDM Wavelengths
(Channels 20 through 59)

DWDM

- Multiple signals are input, each on a different wavelength, called a lambda.
- The lambdas are combined on the fiber and transmitted to the receiver, where they are demuxed to separate the lambdas.



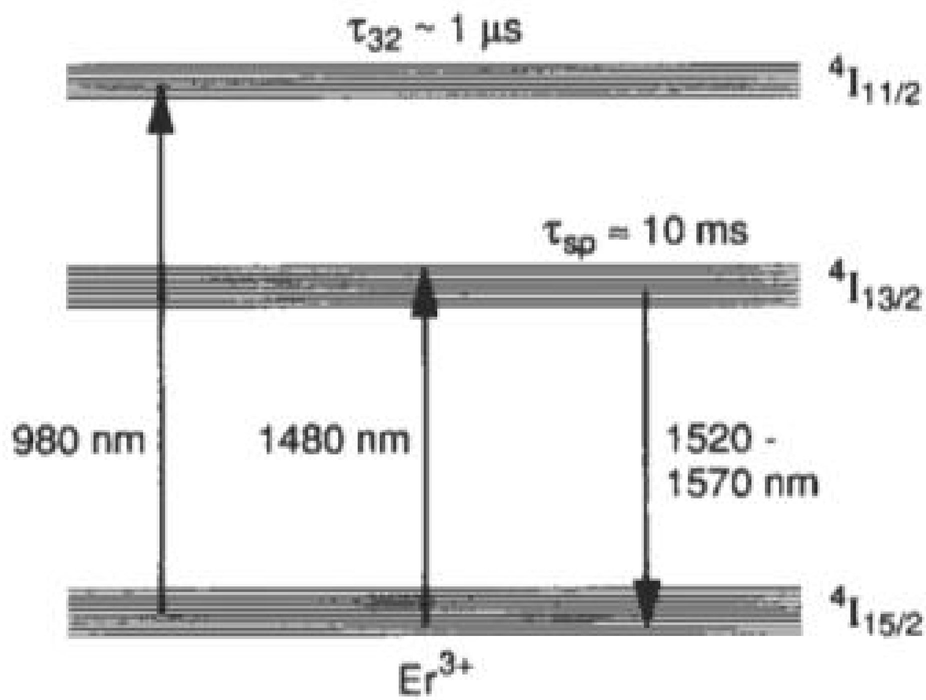
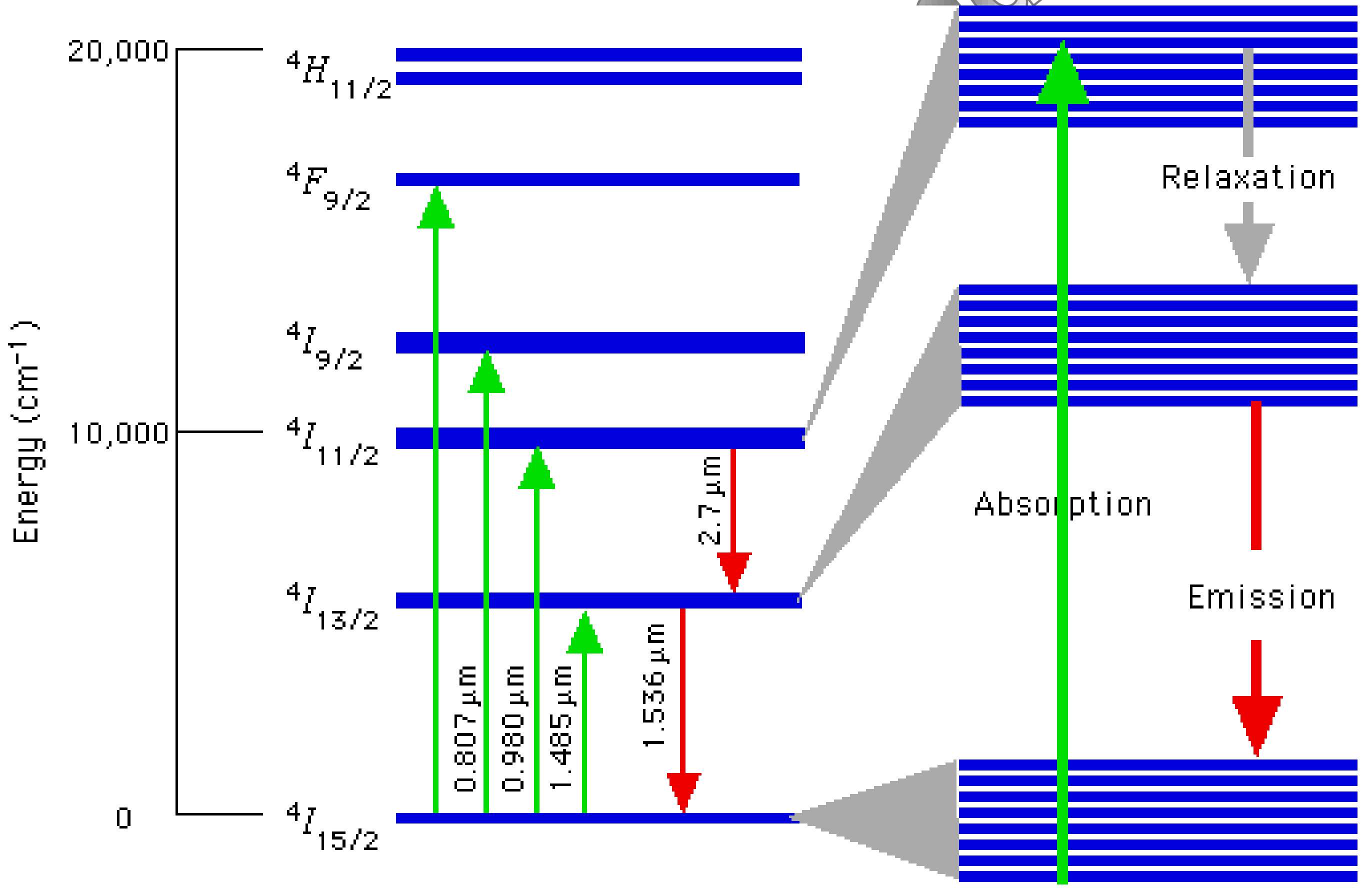


Fig. 2.2 Erbium energy-level scheme.

EDFA

- Erbium is a three level system, where a 980nm laser will pump electrons

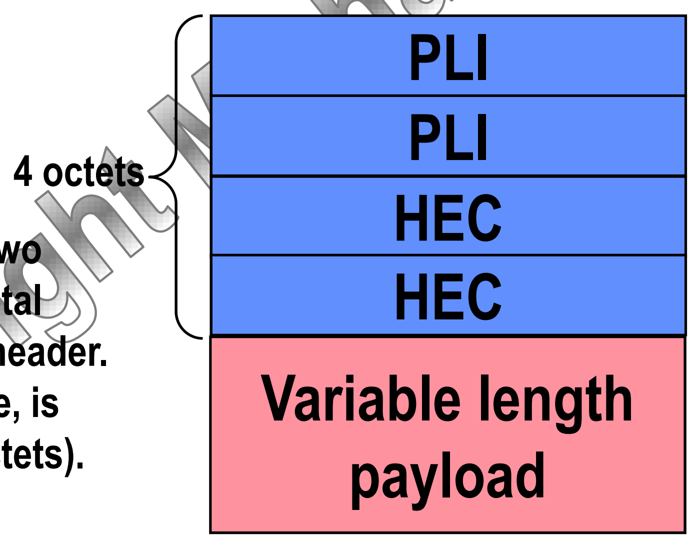


Support for ATM, POS, and GFP

- **Asynchronous Transfer Mode (ATM), Packet over SONET (POS) and Generic Framing Procedure (GFP) are simply mapped into the SPE as a serial data stream, octet aligned with the SONET/SDH octets.**
- **When traffic is mapped into an STS-1 payload, columns 30 and 59 are not used for payload (fixed stuff).**
- **ATM, POS and GFP can be mapped into higher speed concatenated payloads.**
  - Mapping is simply done by putting the ATM, POS or GFP octets into the concatenated SPE. SONET/SDH does not examine the payload octets in any way.

Generic Framing Procedure

- **A way of framing variable length data without a framing character.**
  - To overcome the problem of “shielding” in POS.
- **GFP is being defined in the ANSI T1X1.5 committee.**
  - Can be thought of a variable length ATM type of framing.
  - Header has header check octets and payload length. GFP frame delineation is similar to finding ATM header.



Length specified in the two octets of the PLI is for total frame length, including header. Max frame size, therefore, is (64K-1) octets (65,535 octets).

PLI = Payload length indicator
HEC = Header error control (CRC-16)