Computer Networks

Chapter 2: Physical Layer (2/2)

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Outline

Essence: Provide the means to transmit bits from sender to receiver $\Rightarrow$ involves a lot on how to use (analog) signals for digital information

- Theoretical background: signal transmission and Fourier analysis
- Transmission media (wires and no wires)
- Modulation techniques (the actual encoding), multiplexing, and switching

数据通信的理论基础

- Fourier Analysis: 周期为T的函数g(t)可如下表示:

$$g(t) = \frac{1}{2}c + \sum_{n=1}^{\infty} a_n \sin(2\pi n f t) + \sum_{n=1}^{\infty} b_n \cos(2\pi n f t)$$

where f=1/T, a_n and b_n are the sine and cosine amplitudes of the nth harmonics (terms), and c is a constant.

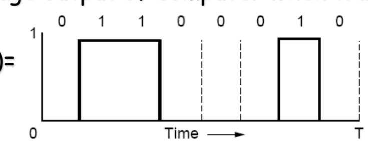
$$a_n = \frac{2}{T}\int_0^T g(t)\sin(2\pi n f t)\,dt \qquad b_n = \frac{2}{T}\int_0^T g(t)\cos(2\pi n f t)\,dt \qquad c = \frac{2}{T}\int_0^T g(t)\,dt$$

Continue…

- Example:

The voltage output of computer when transmitting 'b',

g(t)=



The Fourier analysis of this signal yields the coefficients:

$$a_n = \frac{1}{\pi n}[\cos(\pi n/4) - \cos(3\pi n/4) + \cos(6\pi n/4) - \cos(7\pi n/4)]$$

$$b_n = \frac{1}{\pi n}[\sin(3\pi n/4) - \sin(\pi n/4) + \sin(7\pi n/4) - \sin(6\pi n/4)]$$

$$c = 3/4$$

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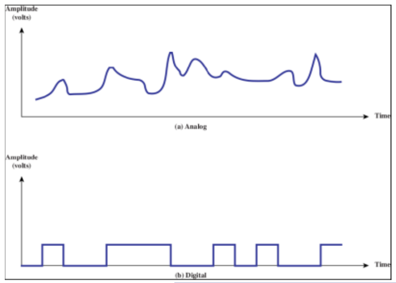
- 这意味着什么呢?
  - 传输的数字信号 可以看作是由 无限多个周期模拟信号叠加而成.



模拟信号

数字信号

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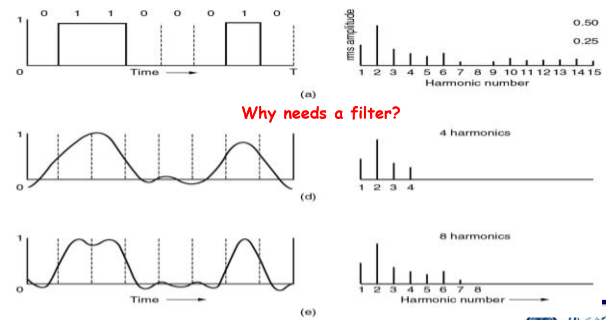
- Th1: $a_n^2 + b_n^2$ 正比于传输相应频率分量所需能量。



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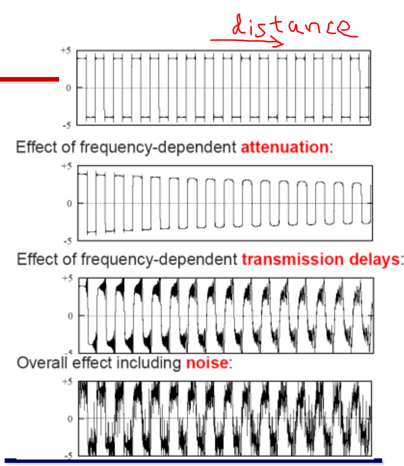
- 传输介质

1) 不存在无损能量的传输介质, 而且不同的付立叶分量能量损耗还不一样。有信号衰减。attenuation(衰减)

2) 不同的分量在介质中的传播速度还不一样。这会导致在接收方发生信号畸变。distortion(畸变)



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- 人为引入过滤器

比如: 电话公司过滤器频率范围0~3100 Hz.
  - 这个范围足够满足语音通信的需要,
  - 并且能提高全系统资源的利用率（介质频分通信）

Def. 通常, 0~fc 频率的the amplitudes of harmonics是不会改变的, 0~fc was called the bandwidth. 实际上带宽被定义成0到能还有一半能量通过的频率处。
  - 带宽是介质的固有属性, 取决于材料、粗细和长度。
  - 3100Hz 也称为语音级线路带宽。
    - 假设发送8位时间为T=8/b秒, 因此1次谐波频率b/8Hz
    - 这意味着通过的最大谐波号约是 3000T=3000*8/b. //nf=3000 → n=3000T

数据传输速度和介质带宽的关系

Bps b	T (msec)	First harmonic (Hz)	# Harmonics sent
300	26.67	37.5	80
600	13.33	75	40
1200	6.67	150	20
2400	3.33	300	10
4800	1.67	600	5
9600	0.83	1200	2
19200	0.42	2400	1
38400	0.21	4800	0

Assumption: We are using a simple encoding technique based on the fact that the line supports only two signal values.

Observation: Most telephone carriers cut off the highest frequency at 3000 Hz ⇒ we can never transmit at a higher speed than 9600 bps.

Continue…

Improvement: If there are four signal values available, we could encode 2 bits at a time:

$00 \to 0$ volt $01 \to 2$ volt
$10 \to 4$ volt $11 \to 6$ volt

The number changes in a signal per second is called the **baud**. vs bps, bandwidth

Example 2: A 2400 bauds line (modem) can make a bit rate of 9600 (14.4) bps provided it uses 16 (2^4) (2^6) signal values:

S	bits	S	bits	S	bits	S	bits
0	0000	4	0100	8	1000	12	1100
1	0001	5	0101	9	1001	13	1101
2	0010	6	0110	10	1010	14	1110
3	0011	7	0111	11	1011	15	1111

□ 问题：

可以无限制的去提高介质数据传输率？

Nyquist and Shannon

Nyquist showed that if the cut-off frequency is H Hz, the filtered signal can be reconstructed by making $2H$ samples. No more, no less. **Consequence:**

maximum transmission rate = $2H\log_2 V$ bps

(where V is the number of signal values)

Shannon showed that a *noisy* channel with a signal-to-noise ration S/R, has a limit with respect to the bit rate:

maximum transmission rate = $H\log_2(1+S/R)$ bps

Example: A telephone line with H = 3000 and $10\log_{10}(S/R) = 30$ dB, can do no better than 30 kbps, no matter how you do your encoding (excluding compression).

56kbps modem?

Continue…

- 56kbps modems use a 8000 baud line (4000Hz) with 8 bits per sample (1 bit is reserved for control purpose)

  //$H\log_2(1+S/R)$，R降低一半

- ADSL uses up to 224 4-kHz channels, With 15 bits/baud and 4000 baud, the downstream bandwidth would be 13.4 Mbps (more details later on)

传输介质---磁带

- Never underestimate the bandwidth of a station wagon full of tapes hurtling down the highway
  - Take a standard videotape that can carry about 7GB of data.
  - A box of 50*50*50cm can hold about 1000 tapes, which corresponds to 7000GB.
  - Sending such a box can be done within 24 hours, worldwide.

We've got a transmission rate of 648 Mbps!

Question: What is overlooked in this reasoning?

铜线(1/2)

- Twisted pair: 两根绝缘铜线, twisted like a DNA string (reduces electrical inference). Often, twisted pairs go by the bundle. Comparable to telephone wiring at home.

(a)

(a) Category 3 UTP(16MHz)

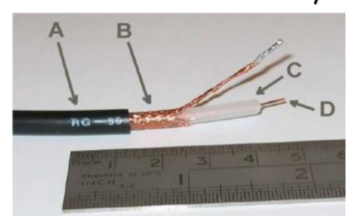
(b)

(b) Category 5 UTP(100MHz)

- Further distinction between shielded (STP) and unshielded (UTP) versions, but the shielded ones used to be primarily used only with IBM installations; now also for local networks (Cat 7,600MHz)

铜线(2/2)

- Coaxial cable: like the one you use for your TV Set:

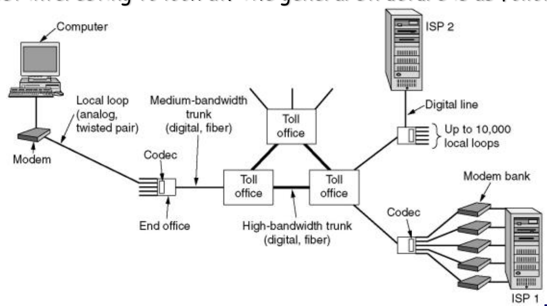


D=Copper core
C=insulating material
B=Braided outer conductor
A=Protective plastic covering

- Coax is better than twisted pair when you need more bandwidth (in that it has better shielding, up to 1GHz), mainly used in MAN and Cable TV, but is now rapidly being replaced with fiber.

The local loop（本地回路）

- Observation: when it comes the telephone system, from a networking perspective the local loop (a.k.a. the last mile) is the most interesting to look at. The general structure is as follows:



调制技术(1/5)

- Problem: How can we encode our signals when we can effectively use only a single frequency (or better: small frequency range)？

 Answer: Apply modulation techniques:

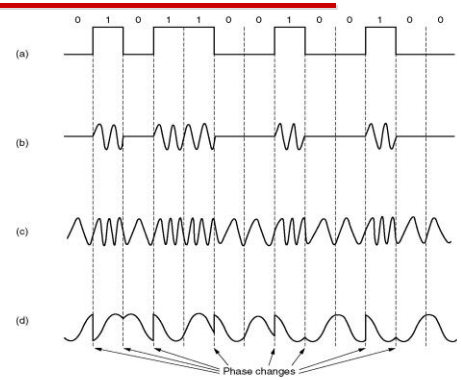
  - Change the amplitude (strength) of the signal: changing amplitude means a binary 1, constant amplitude a binary 0.
  - Use different frequencies to encode your bits (these frequencies can be put “on top” of your base frequency).
  - Change the phase of the wave (cf. sine and cosine) to do signal encoding.

调制技术(2/5)

(a) A binary signal
(b) Amplitude modulation

(c) Frequency modulation
(d) Phase modulation

分绝对相移和相对相移

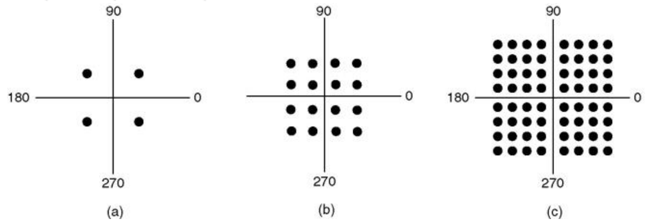


调制技术(3/5)

- 一些概念
  - 实践中,大多数Modem采样频率2400Hz,焦点在于:每次采样如何表达更多的位信息.
  - 每秒钟采样的次数用波特(Baud)计量
  - 在一个波特中,发送的是一个码元
  - 一个码元可编码m位数据
  - 数据传输速率就是波特率的m倍
- 组合几种调制技术，以便在每个波特中传输尽可能多的位
  - 描述工具：星座图

调制技术(4/5)

- QPSK，QAM-16，QAM-64



正交相移键控：4种相移，每个码元2位→4800bps

正交振幅调制（QAM）：3种振幅12种相移，每个码元4位→9600bps

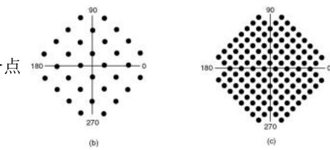
每个码元6位→14.4kbps

- **星座中的点越多，干扰引起传输错误的可能性越大**

调制技术(5/5)

- V.32 for 9600 bps. V.32 bis for 14,400 bps 带1位奇偶校验位

32个点

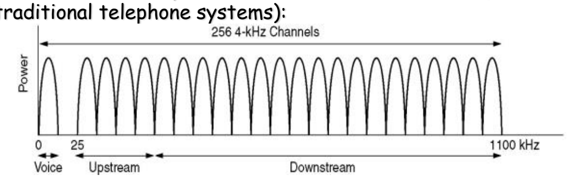


128个点

- V.34: 12个数据位，28.8kbps；V.34 bis: 14个数据位，33.6kbps
- 香农modem限制35kbps，56kbps modem何来？
  - 减少一个本地回路，S/N增大，最大速率可以增加一倍→70kbps
  - 为什么不是70kbps，而是56kbps?
  - 在美国，电话信道带宽4kHz，采样8000波特，每次采样的位数是8，1位用于控制，所以56kbps。
  - 标准名为：V .90（33.6kbps上行，56kbps下行）
  - V.92（48kbps上行，56下行kbps）

Asymmetric DSL(1/3)

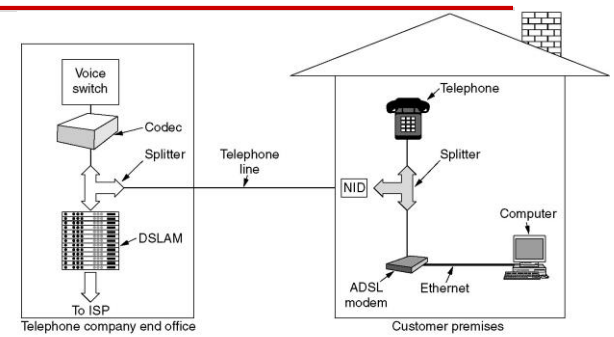
- Essence: considering that the local loop has a 1.1MHz spectrum, we can divide the spectrum into 256 4kHz channels (like in traditional telephone systems):



使用离散多信道调制技术（DMT）的ADSL频谱划分方案

- Note: it is up to the provider to decide how it will arrange its channels. Different combinations are possible.
- 每条信道使用QAM（正交振幅调制）调制方案，但采样频率4000Hz

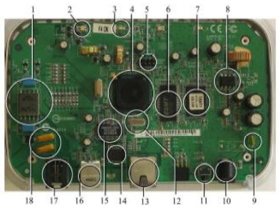
Asymmetric DSL(2/3)



NID	Network Interface Device
DSLAM	DSL Access Multiplexer

A typical ADSL equipment configuration.

Asymmetric DSL(3/3)



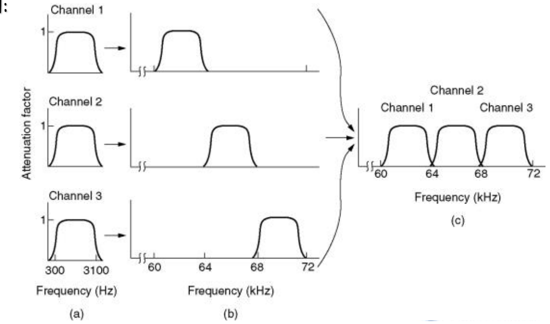
ADSL Modem

4:	CPU	5:	JTAG interface
6:	8 Mb RAM	7:	Flash memory
13:	Ethernet port	16:	USB port
17:	Telephone port		

DSLAM

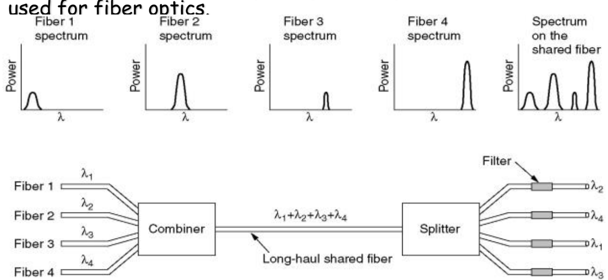
复用 Multiplexing: FDM

- Problem: considering that the bandwidth of a channel can be huge, wouldn't it be possible to divide the channel into sub-channels?
- Frequency Division Multiplexing: Divide the available bandwidth into channels through frequency filtering, and apply modulation techniques per channel:



复用 Multiplexing: WDM*

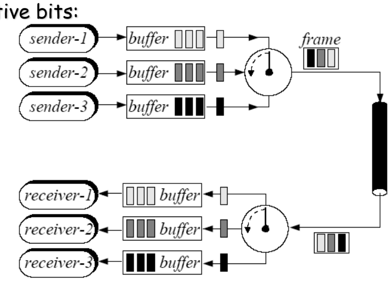
- Wavelength Division Multiplexing: actually the same as FDM, but used for fiber optics.



- Observation: light waves have their own frequency range; they are simply combined and separated using standard (de)fraction properities.

复用 Multiplexing: TDM (1/4)

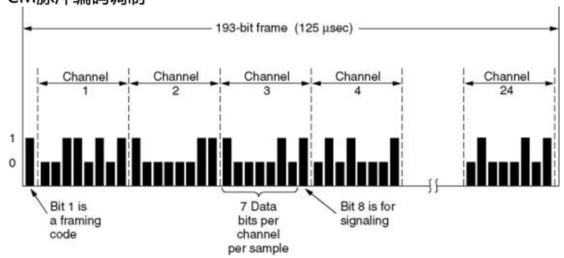
- Time Division multiplexing: simply merge/split streams of digital data into a new stream. Data is handled in frames-a fixed series of consecutive bits:



- Observation: this is full-digital solution in contrast to FDM and WDM（完全数字化的复用线路方案）

复用Multiplexing: TDM(2/4)

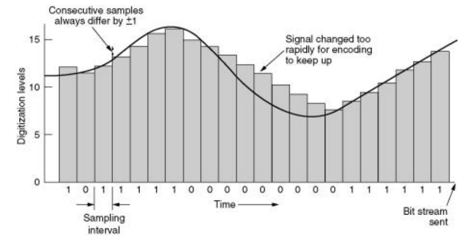
- Example: the T1 system samples at 8000Hz, and encodes each sample as a 7-bit number (i.e. 128 different values). With some extra control bits, we merge samples into 193-bit frames, every 125usec.
  PCM脉冲编码调制



- Observation:T1 supports a total of 1.544Mbps

复用Multiplexing: TDM (3/4)

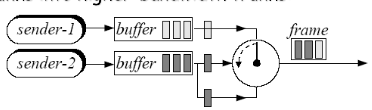
- 压缩方法：减少每条信道所需要的位数
  - 差分PCM:传输前后采样值之差，5位足够，无需7位。
  - 增量调制：+-1较前一采样是高了，还是低了。1位。

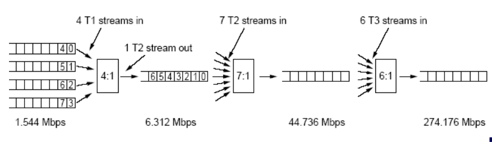


  - 预测编码：对实际与预测的差值编码

复用Multiplexing: TDM (4/4)

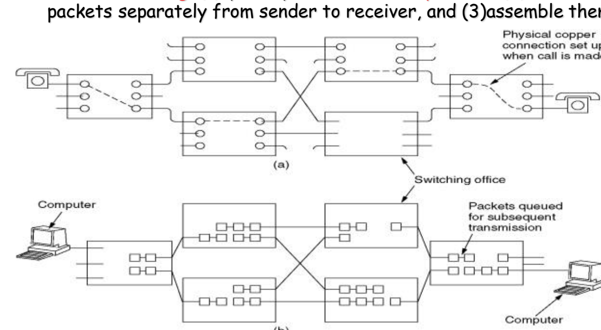
- Observation: TDM also makes it easy to offer individual senders higher bandwidth, by simply putting more data into a frame, or to combine several trunks into higher-bandwidth trunks:





交换Switching(1/2)

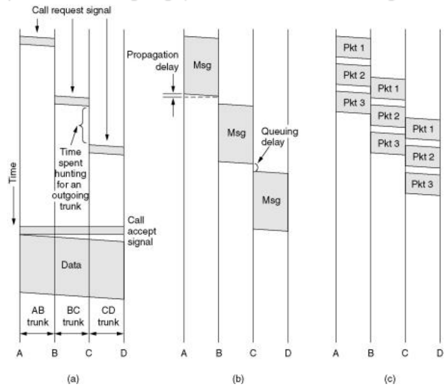
- Circuit switching: make a true physical connection from sender to receiver. This is what happens in traditional telephone systems.
- Packet switching: (1)split any data into small packets, (2) route those packets separately from sender to receiver, and (3)assemble them again.



交换Switching (2/2)

- Variation: message switching – a message is completely received at a router, stored, and then put into an outgoing queue for further routing.

(a) Circuit switching
(b) Message switching
(c) Packet switching



交换Switching: Comparision

Item	Circuit-switched	Packet-switched
Call setup	Required	Not needed
Dedicated physical path	Yes	No
Each packet follows the same route	Yes	No
Packets arrive in order	Yes	No
Is a switch crash fatal	Yes	No
Bandwidth available	Fixed	Dynamic
When can congestion occur	At setup time	On every packet
Potentially wasted bandwidth	Yes	No
Store-and-forward transmission	No	Yes
Transparency	Yes	No
Charging	Per minute	Per packet

CDMA(1/2,码分多路访问)

- Code Division Multiple Access allows transmissions to be interleaved, but avoids interference. Note that this means inherently no message collision.
- Principle: assign a chip sequence to a station, which is just an m-bit code. Make sure that all chip sequences are pairwise orthogonal:
  - Rewrite a binary 0 as -1, and a binary 1 as +1
  - Send a 1 bit as your chip sequence, and a 0 bit as the inverse
  - Just transmit your bits when a new bit time slot starts → the chip sequences时间片序列 (possibly inversed补码) are just added.
  - Getting the original value means taking the inner product of the original chip sequence with the signal sent.

CDMA(2/2,码分多路访问)

A: 0 0 0 1 1 0 1 1
B: 0 0 1 0 1 1 1 0
C: 0 1 0 1 1 1 0 0
D: 0 1 0 0 0 0 1 0

(a)

4个移动点的chip sequence

A: (−1 −1 −1 +1 +1 −1 +1 +1)
B: (−1 −1 +1 −1 +1 +1 +1 −1)
C: (−1 +1 −1 +1 +1 +1 −1 −1)
D: (−1 +1 −1 −1 −1 −1 +1 −1)

(b)

芯片序列的双极表示

Six examples:

– – 1 –	C	S_1 = (−1 +1 −1 +1 +1 +1 −1 −1)
– 1 1 –	B + $\bar{C}$	S_2 = (−2 0 0 0 +2 +2 0 −2)
1 0 – –	A + $\bar{B}$	S_3 = (0 0 −2 +2 0 −2 0 +2)
1 0 1 –	A + $\bar{B}$ + C	S_4 = (−1 +1 −3 +3 +1 −1 −1 +1)
1 1 1 1	A + B + C + D	S_5 = (−4 0 −2 0 +2 0 +2 −2)
1 1 0 1	A + B + $\bar{C}$ + D	S_6 = (−2 −2 0 −2 0 −2 +4 0)

(c) 6个传输例子

$S_1 \bullet C$ = (1 +1 +1 +1 +1 +1 +1 +1)/8 = 1
$S_2 \bullet C$ = (2 +0 +0 +0 +2 +2 +0 +2)/8 = 1
$S_3 \bullet C$ = (0 +0 +2 +2 +0 −2 +0 −2)/8 = 0
$S_4 \bullet C$ = (1 +1 +3 +3 +1 −1 +1 −1)/8 = 1
$S_5 \bullet C$ = (4 +0 +2 +0 +2 +0 −2 +2)/8 = 1
$S_6 \bullet C$ = (2 −2 +0 −2 +0 −2 −4 +0)/8 = −1

(d) 接收方恢复C的信号过程

AMPS & GSM: 频段→信道→时槽
CDMA:
1、在整个频段范围内发送信号
2、利用编码技术分离叠加信号