Data Communication and Computer Networks

COMP 445
Department of Computer Science
Concordia University
Montreal
Chapter 3
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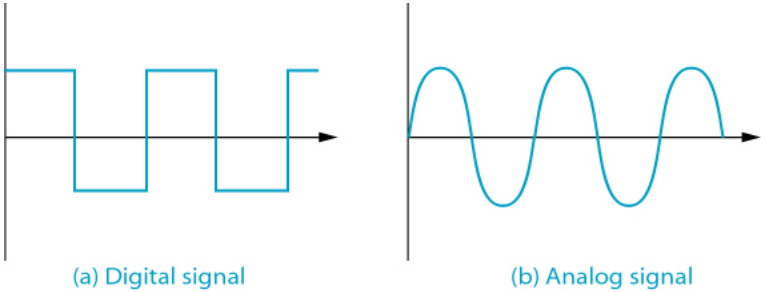
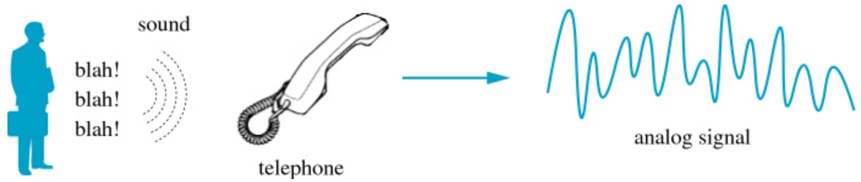
Analog Signals

- Example: Telephone Signal
- Parameters
  - Amplitude
  - Frequency
  - Phase
- Analysis methods
  - Fourier series (for periodic signals)
  - Fourier transform (for Aperiodic signals)



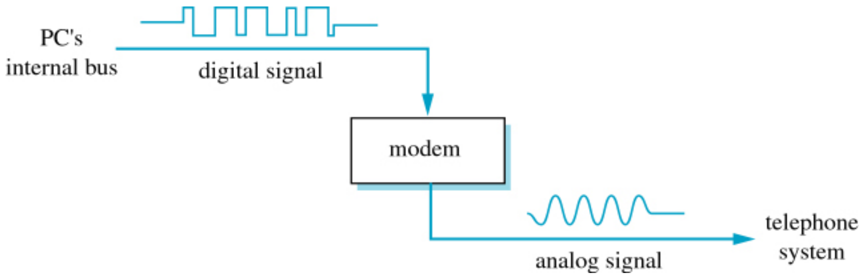
Analog Versus Digital

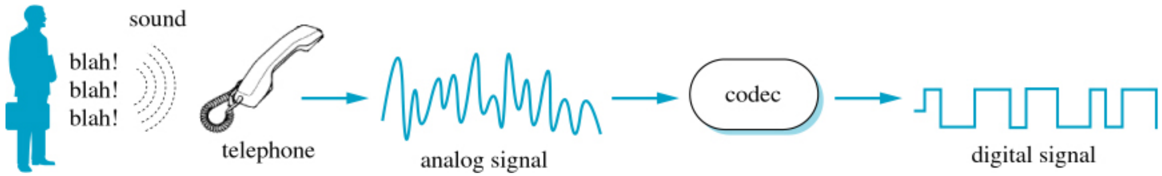
- Advantages of Digital systems
  - Repeaters
  - Ability to do Error Correction
  - Suitable for computer representation



ADC and DAC

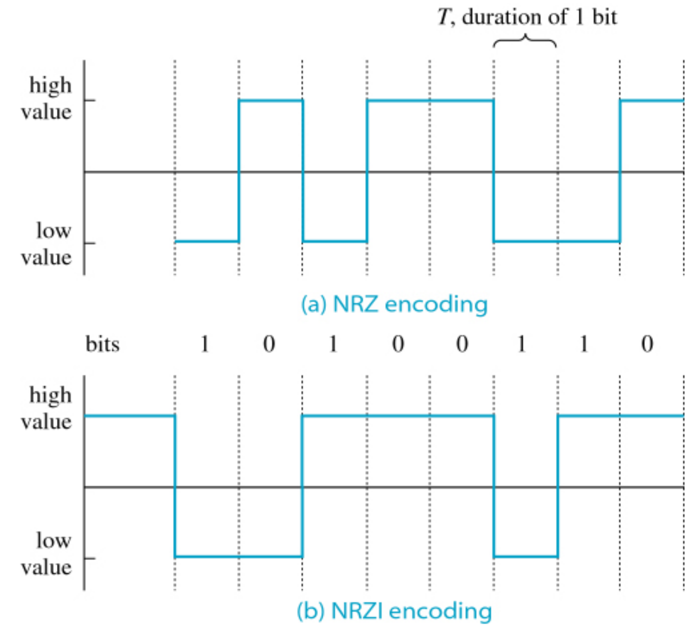
- Computer Data Transmission over telephone line (DAC)
- Voice Information transmitted digitally (ADC)

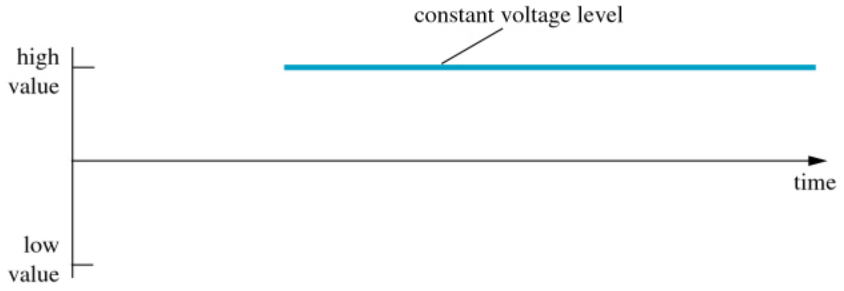




NRZ Encoding Schemes

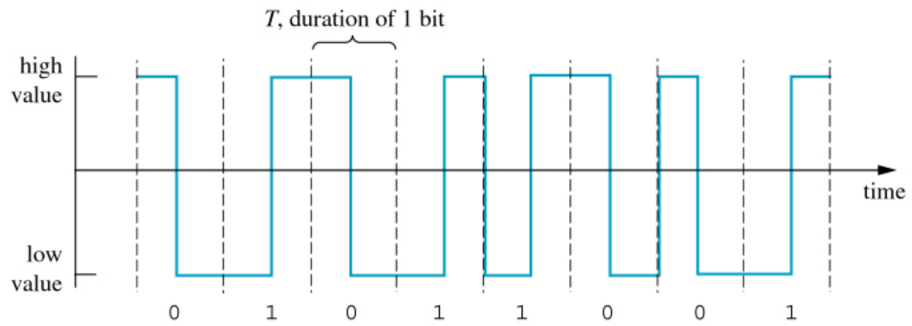
- Non Return to Zero (NRZ)
- Synchronization problems
- Solution
  - Manchester Encoding



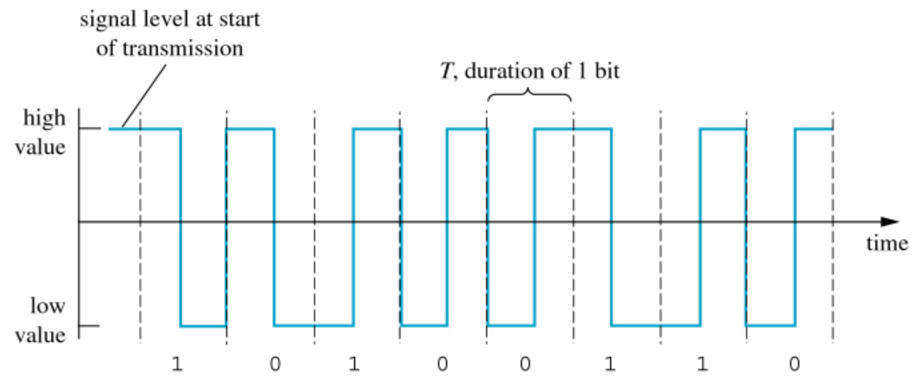


Manchester Encoding

- The signal changes in the middle of each interval
- This change allows the receiver clock to remain consistent with the transmitter clock
- Differential Manchester Encoding



Manchester Encoding



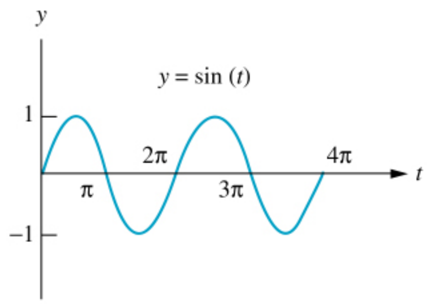
Differential Manchester Encoding

0 causes the signal to change at the start of the interval.
1 causes the signal to remain at the start of the interval.

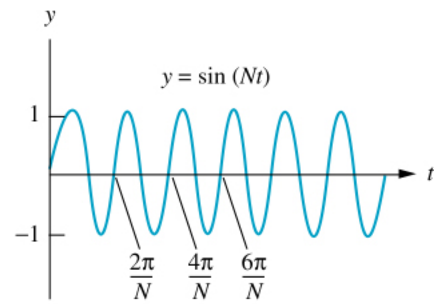
Mathematical background

- Fourier Series/Transform
- Nyquist Sampling Theorem
- Shannon (noisy channel) Theorem

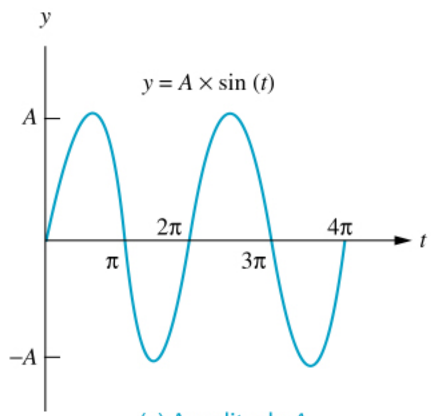
Example for Periodic Signal



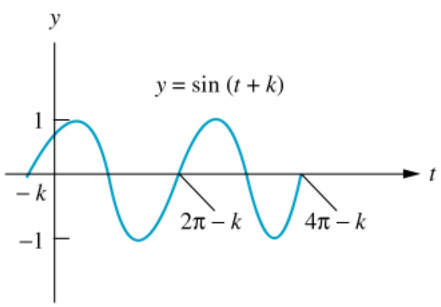
(a) Period 2π



(b) Period $\frac{2\pi}{N}$



(c) Amplitude A



(d) Phase Shift k

Fourier's Result

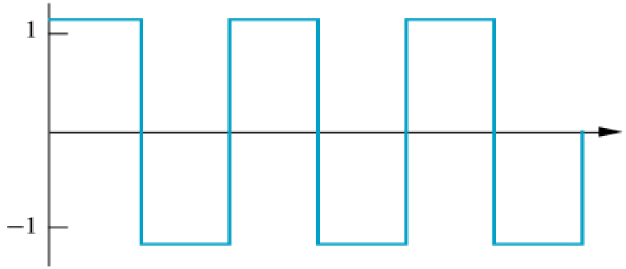
- Any periodic function can be expressed as an infinite sum of sine (and cosine) functions of varying amplitude, frequency and phase shift (Called Fourier Series)

$$s(t) = \sum_{i=1}^{\infty} a_i \times \cos(2\pi i t / P) + b_i \times \sin(2\pi i t / P),$$

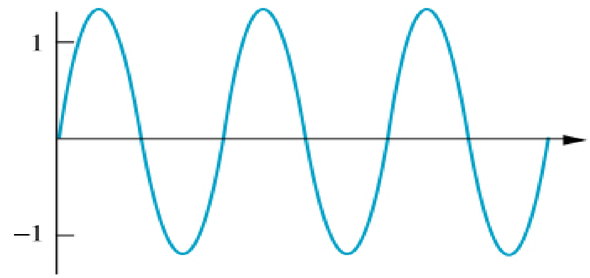
$$a_i = 2 / P \int_{-P/2}^{P/2} s(t) \cos(2\pi i t / P) d(t),$$

$$b_i = 2 / P \int_{-P/2}^{P/2} s(t) \sin(2\pi i t / P) d(t).$$

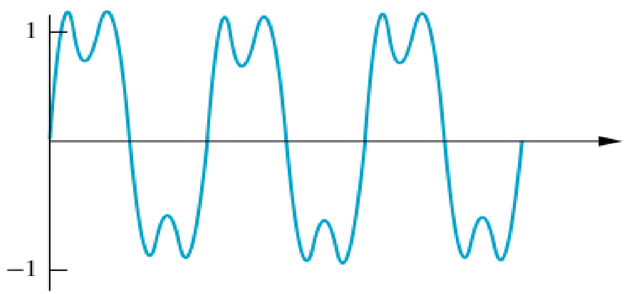
Example



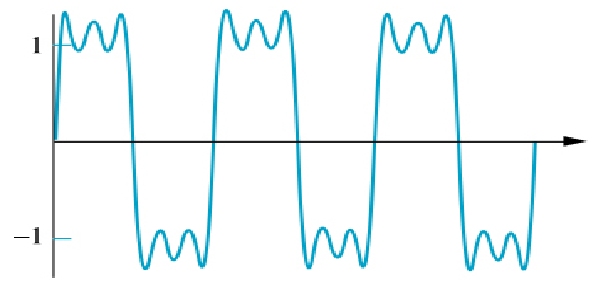
(a) Graph of $s(t)$



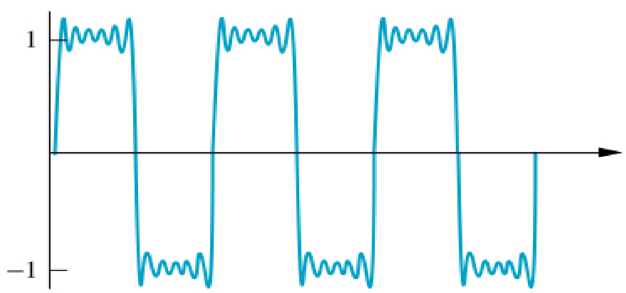
(b) 1-term Fourier approximation to $s(t)$



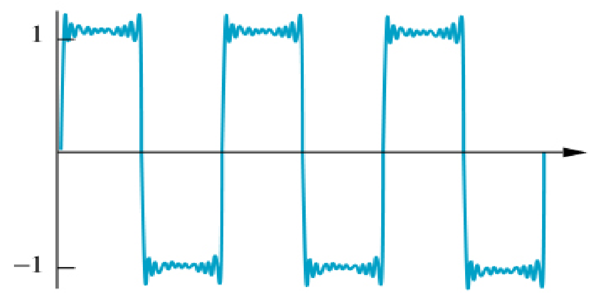
(c) 3-term Fourier approximation to $s(t)$



(d) 5-term Fourier approximation to $s(t)$



(e) 11-term Fourier approximation to $s(t)$



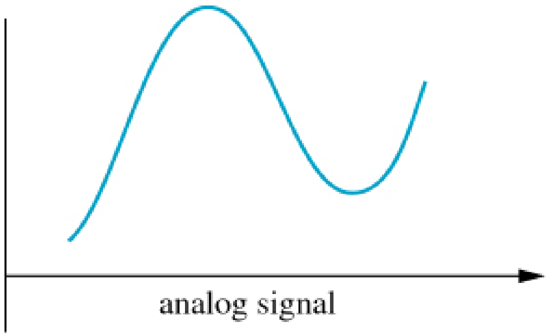
(f) 21-term Fourier approximation to $s(t)$

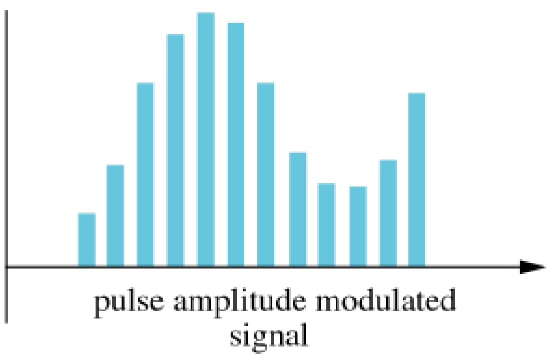
Applications of Fourier's result

- High fidelity equipments are capable of producing signals in the range between 30Hz-30KHz
- Phone: 300-3300KHz
- Filter design
- Frequency Multiplexing (e.g., Cable TV)

Nyquist Sampling Theorem

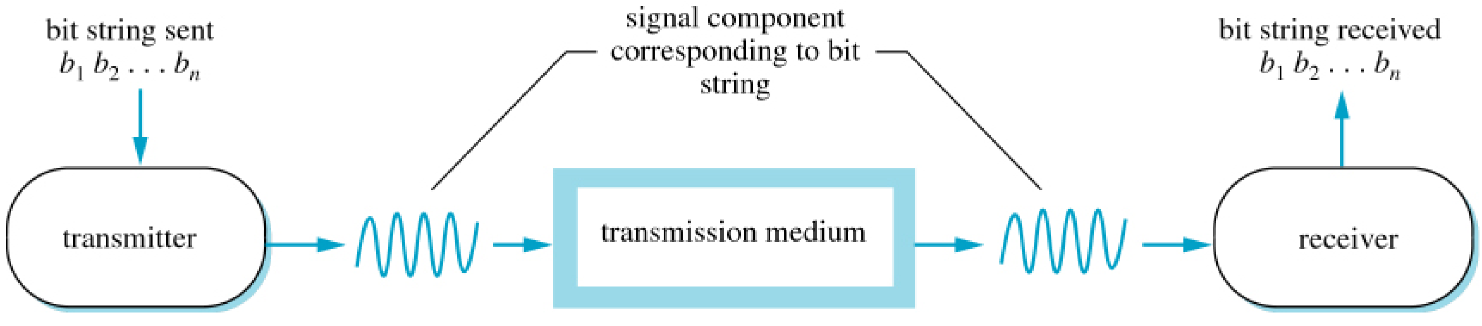
- If the signal is band limited to f_{max}
- Sampling at 2 f_{max} of allows you to reconstruct the original signal





Sending Data via Signals

- Baud Rate
- Bit Rate
- Let n=number of bits per symbol
- Bit rate=Baud Rate X n
- Ex. If the signal has 2^n possible amplitudes, then each signal can represent n bits



Example

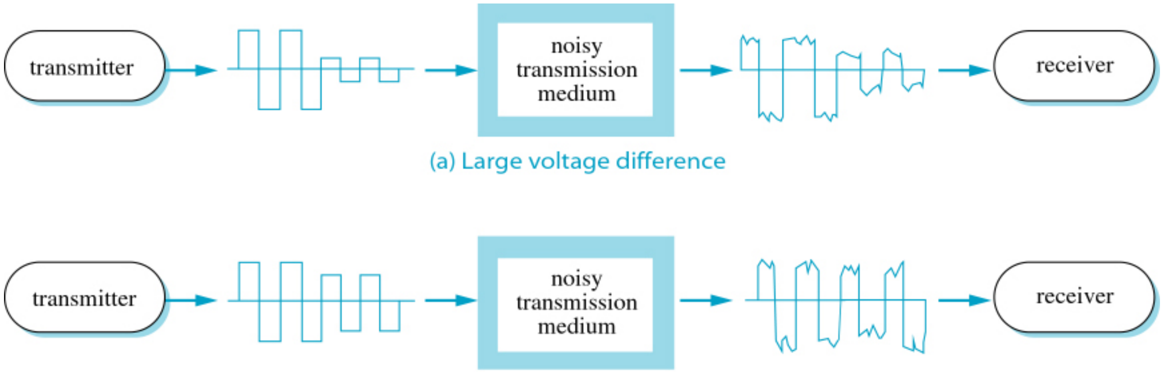
- For telephony voice grad f_{max}= 4KHz
- Sampling frequency= 2 x f_{max}= 8KHz
- Each sample is a baud
- Let the number of level/sample M=256
- Bit rate= 2 x f_{max} x $\log_2(M)$ = 2 x 8 x 8 = 128 k bit/sec

Example

- Assuming 2 bits/baud, we have four possible signals (s_1,s_2,s_3,s_4). These four signals may differ in amplitude, phase, frequency or combination
- Let s_1=>00, s_2=>01, s_3=>10, s_4=>11
- The data 010000111110 => $s_2 s_1 s_1 s_4 s_4 s_3$

Noisy Channel

- Nyquist theorem assumes noiseless channel
- According to Nyqusit theory, A higher bit rate requites more different signal components (M)
- A large M reduces the difference among them (assuming that your power is fixed)
- If M increases, original two voltage levels differ by less. Then it gets more difficult to reconstruct the original signal from the received one



(a) Large voltage difference

(b) Small voltage difference

Shannon Theorem

- Let S/N = Signal power / Noise Power, then

  Bit rate= bandwidth x $\log_2$ (1+S/N)

- The maximum possible data rate depends on the strength of the noise relative to that of the received signal
- S is usually much larger than N
  - SNR in Bels = $\log_{10}$(S/N)
  - 1 dB= 0.1 Bel
  - SNR in dB = 10 $\log_{10}$(S/N)

Example

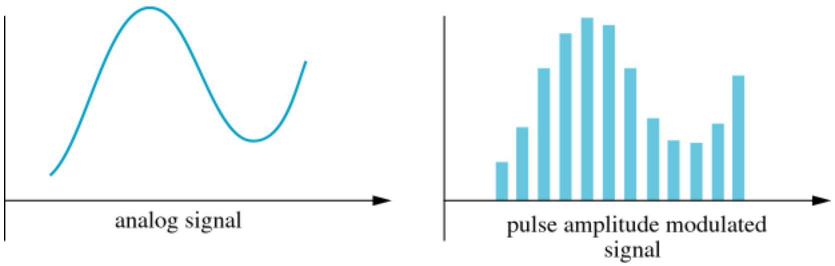
- Let S/N=35 dB, Channel BW==3000Hz then maximum bit rate is given by

  $$3000 \log_2(1+3162)=34,880 \text{ bps}$$

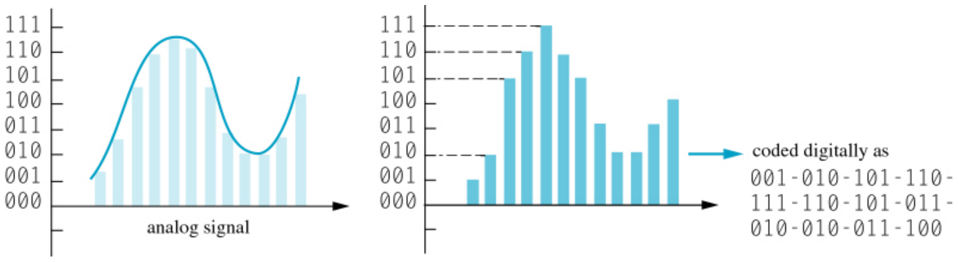
  Hint: S/N=35db=> 10log S/N=35=> $S/N=10^{3.5} = 3162$

Analog To Digital Conversion

- Pulse Amplitude Modulation (PAM)
- Pulse Code Modulation (PCM)
- Sampling @ at least $1/(2f_{max})$
- Higher sampling Rate -> Better quality
- Telephone: 4KHz-> 8K sample/sec -> 64Kbps (for 8 bit / sample)
- CD-> Higher sampling rate and 16 bit/sample



PAM

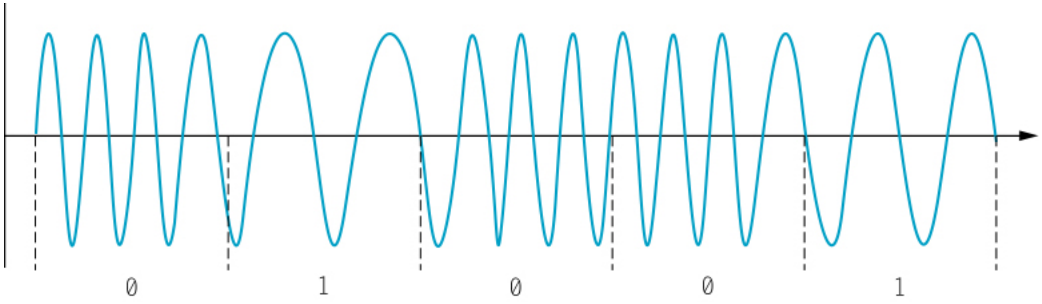


PCM

Digital Modulation Schemes

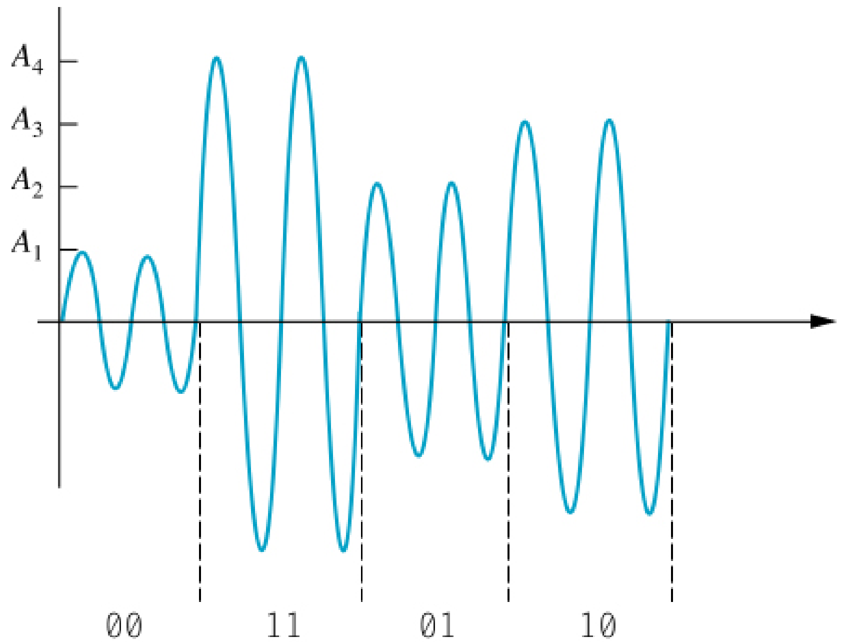
- Why need modulation
  - Antenna size
  - Multiplexing
  - Media constraints
  - Etc.
- Frequency Modulation (FSK)
- Amplitude Modulation (ASK)
- Phase Modulation (PSK)

Frequency Modulation



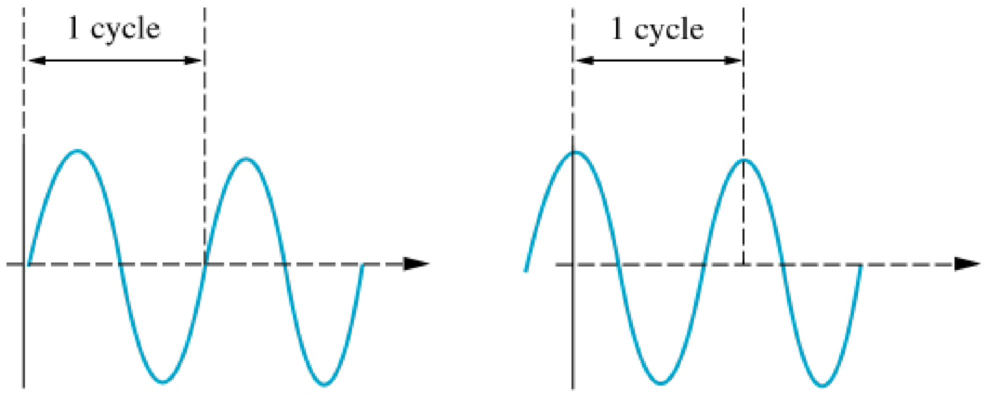
Binary FSK (one bit per baud)

Amplitude Shift Keying



Amplitude Shift Keying (Four amplitudes), two bits per Baud

Phase Shift Keying

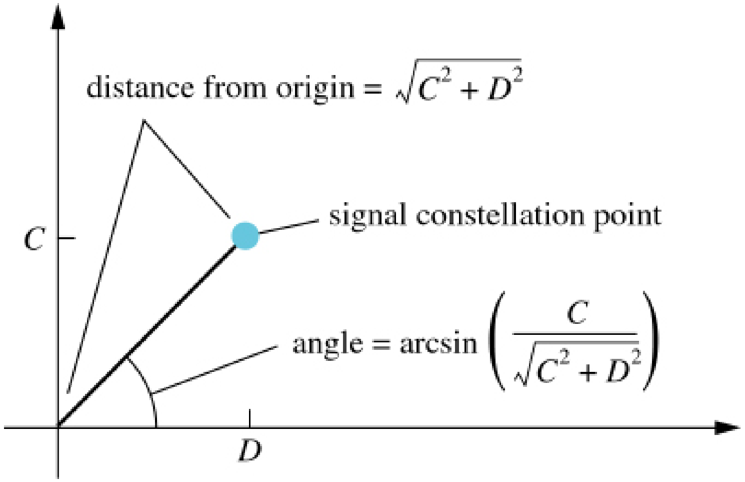


(a) No phase shift

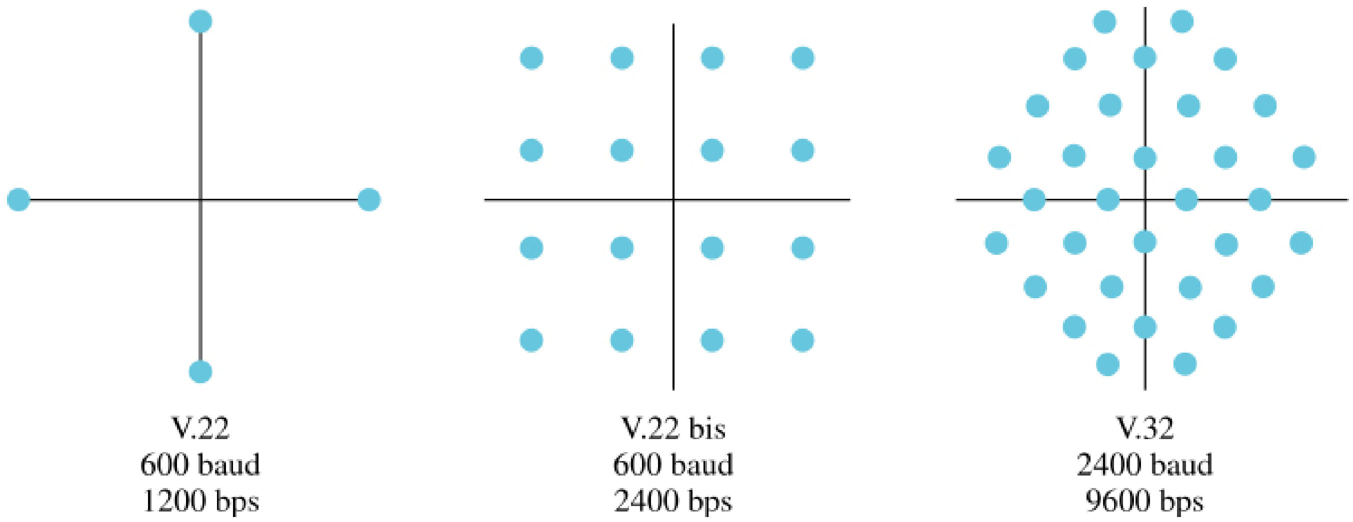
(b) Phase shift of 1/(4*f*)

Modems

- To communicate via telephone lines
- Modulate (Digital to Analog)
- Demodulate (Analog to Digital)
- CCITT standard V.xx modems
- V.21 uses FSK => 1 bit for one frequncy => bit rate=baud rate=> 300bsp
- V.22 use QPSK => 2 bit for each phase shift => 600 baud rate => 1200bsp
- V.90- modems => 56Kbps (downlink)
- Modems that modulate amplitude and phase have a signal constellation

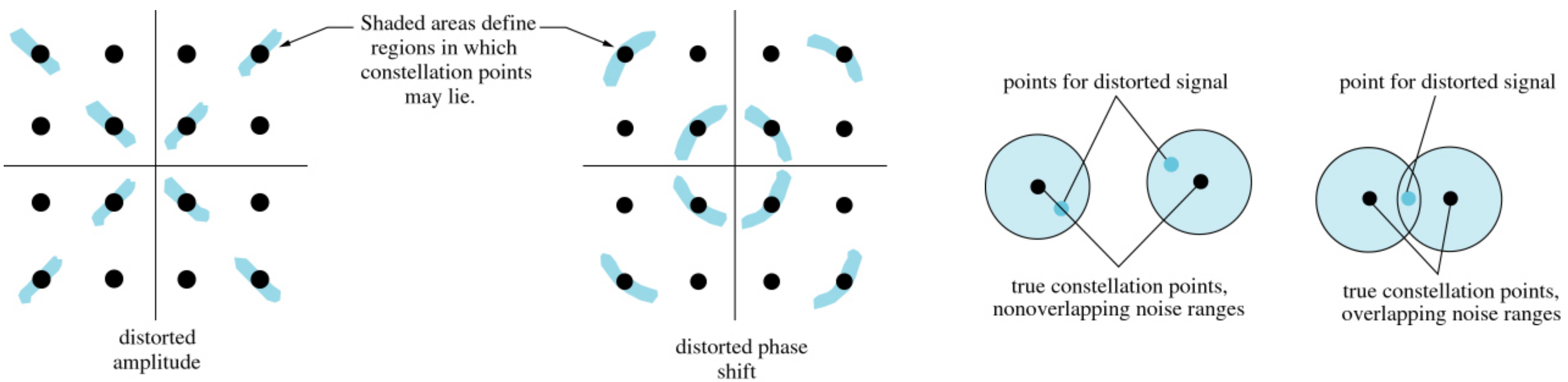


Example for Signal Constellations



Signal Constellations (Cont.)

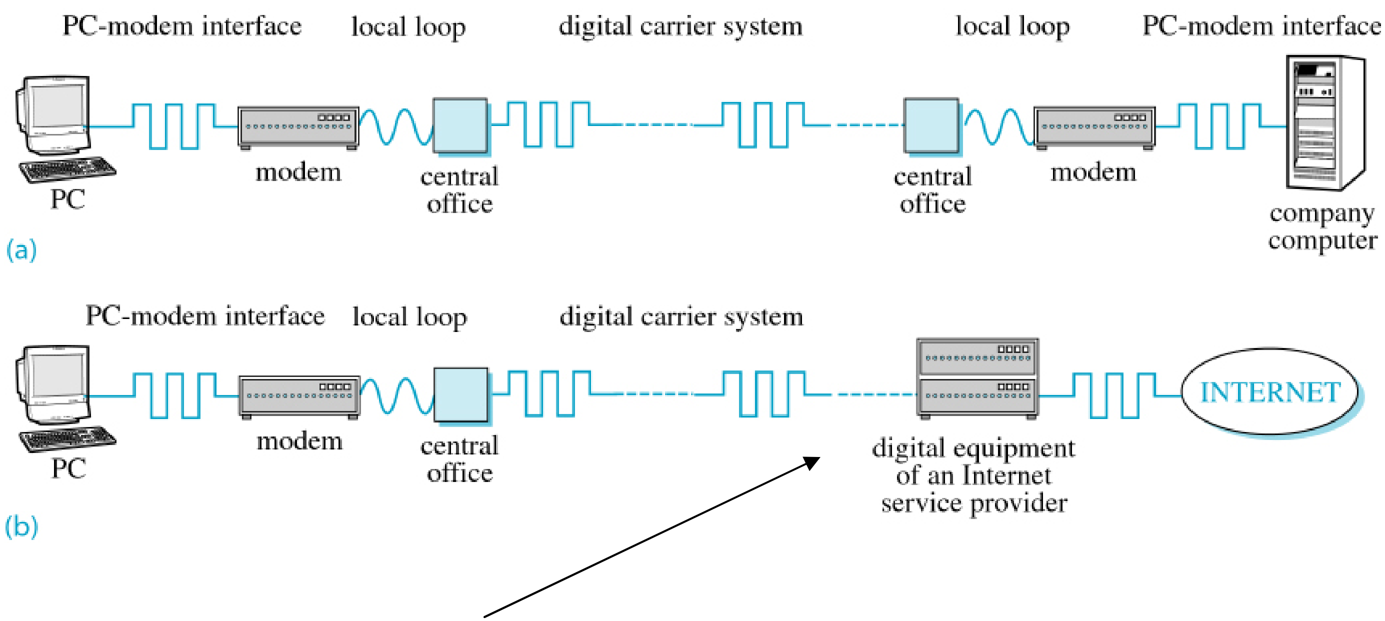
- Noise distorts the signal, i.e., the actual constellation point



Intelligent Modems

- Keywords: S/W and Compatibility
- protocol choose parameters
- More functions- dial AT commands
- Ex. To dial 555-1234 => ATDT5551234
- Call waiting

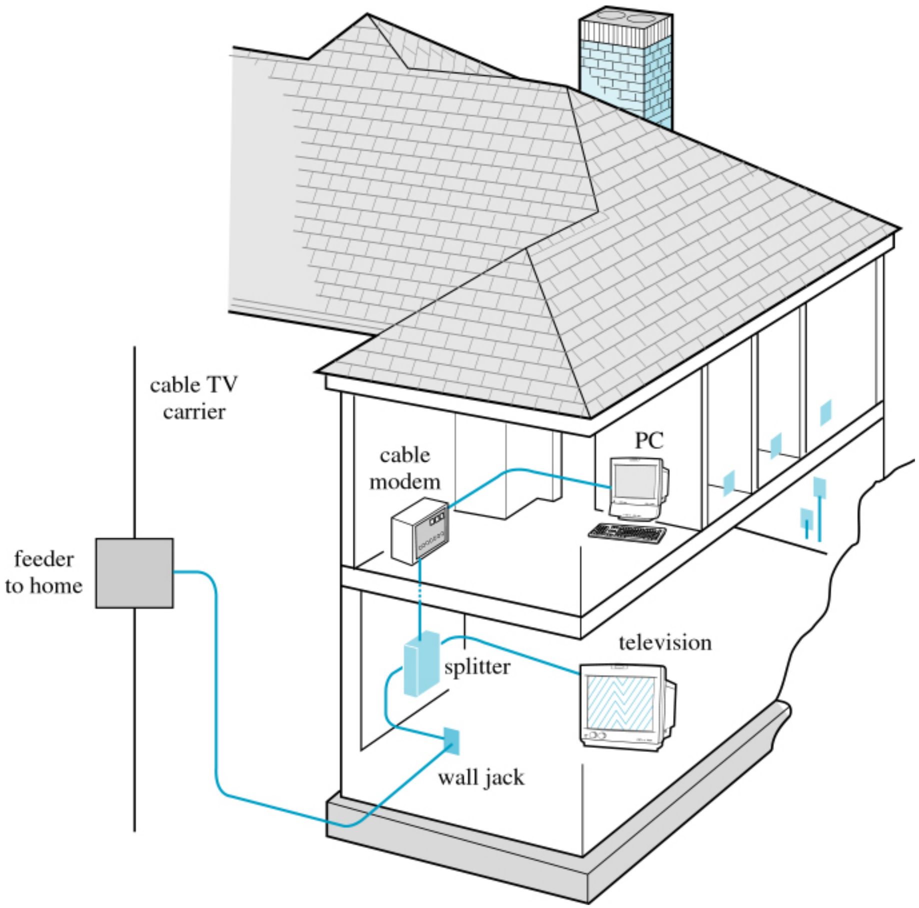
Connections Using a Modem

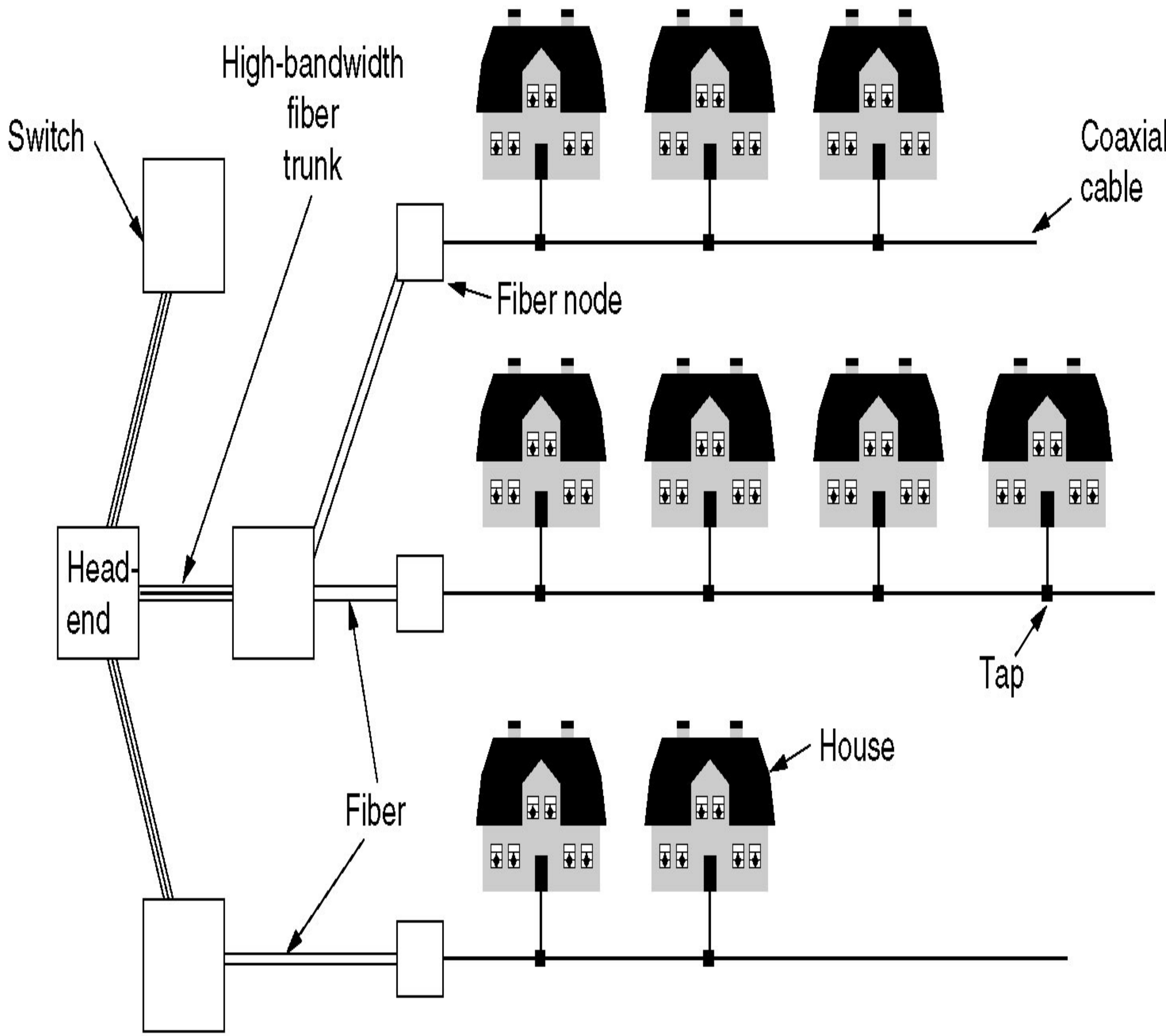


An ISP has digital equipments that communicate directly with the digital carrier. Hence, A/D conversion is required at the remote side (hence no quantization noise and consequently higher bit rate is possible)

Cable Modem

- Connection speed for Phone line (Modem) is less than 56Kbps
- Access the internet via CATV signals instead of calling an ISP
- Much higher bit rate
- No need to dial (no busy lines)
- Shared BW (security problems, load problems)

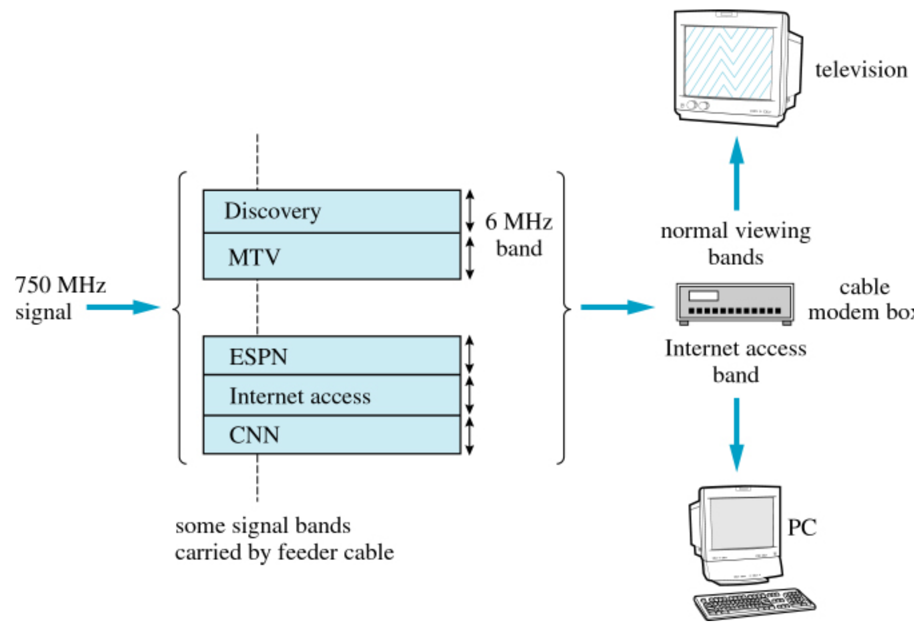



Switch
High-bandwidth fiber trunk
Coaxial cable
Fiber node
Head-end
Tap
House
Fiber

(a)

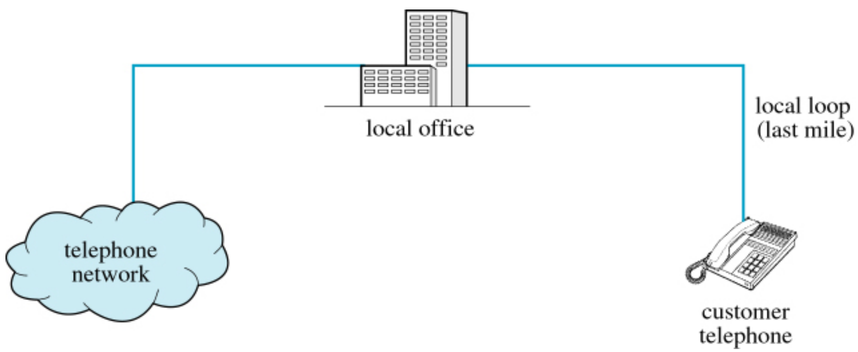
How it works

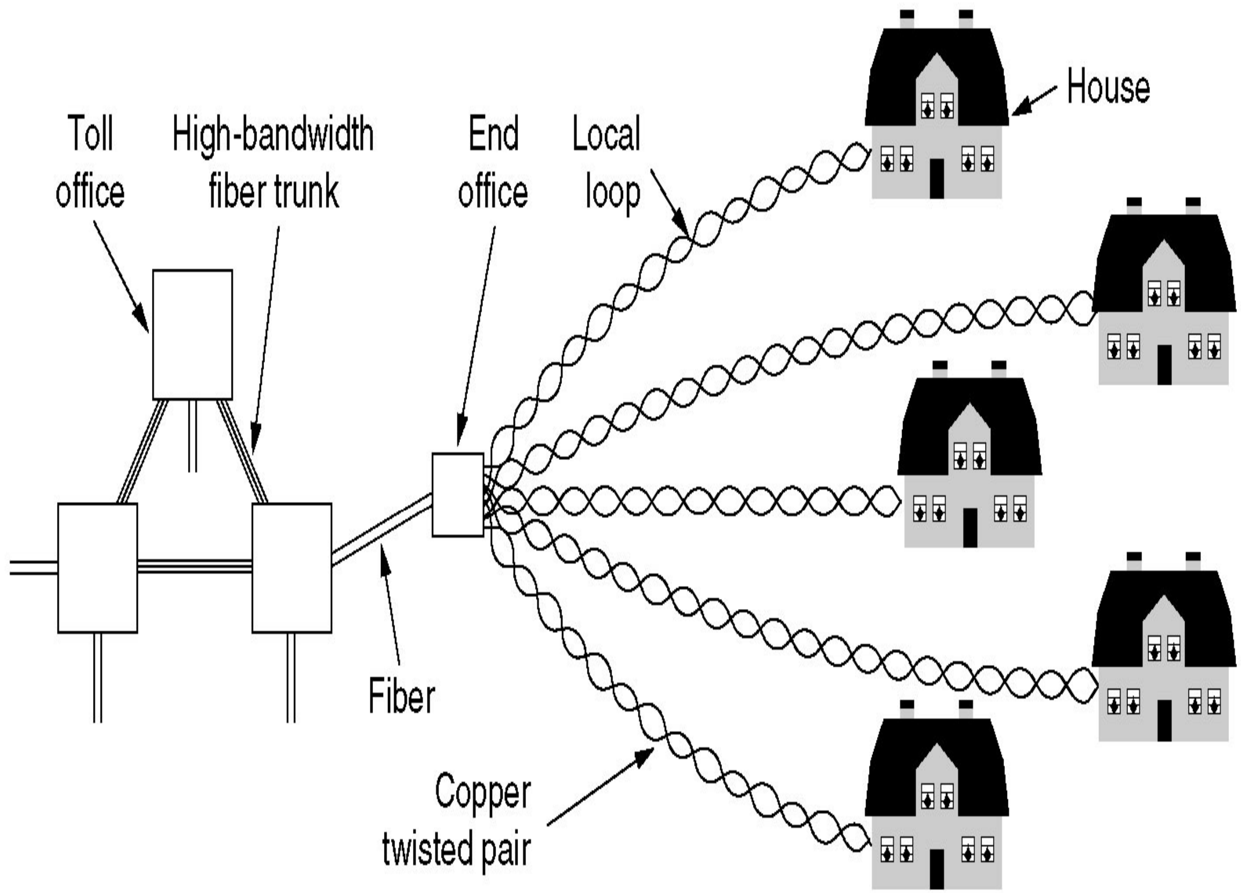
- The 750MHz band is divided into chunks of 6 MHz
- The band 42-750MHz is assigned to the information data stream
- QAM64 (Up to 36Mbps): Realistic speed: 1-10Mbsp (PC constraints)
- IEEE 802.14 standard



Digital Subscriber Line (DSL)

- CATV are not available for every one
- No need for cable (POTS (plain old telephone service) is always there)
- ADSL: About 1.5-6Mbps for down stream
- Usually not available for customers at more than 3.5 miles from local office (signal degrades with distance)



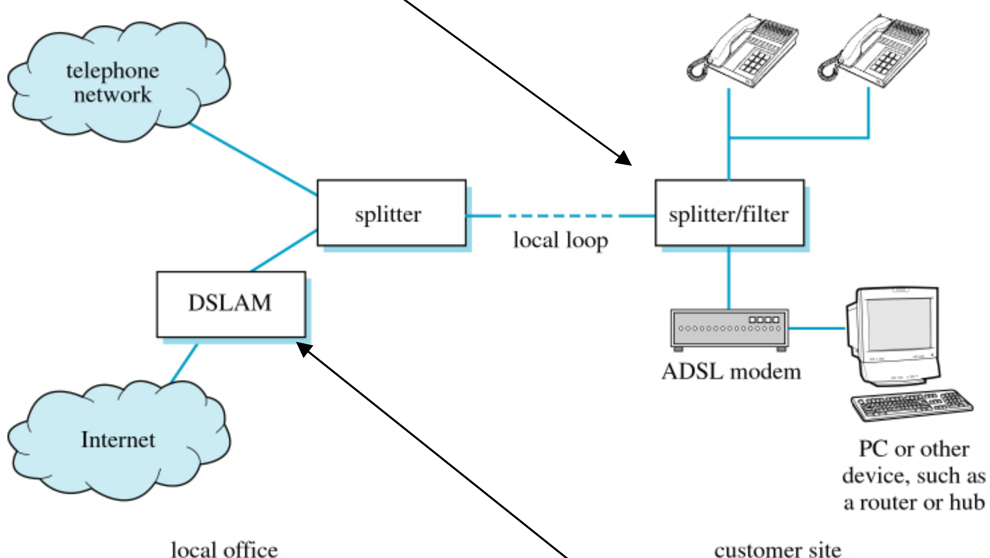


Toll office
High-bandwidth fiber trunk
End office
Local loop
House
Fiber
Copper twisted pair

(b)

How it works

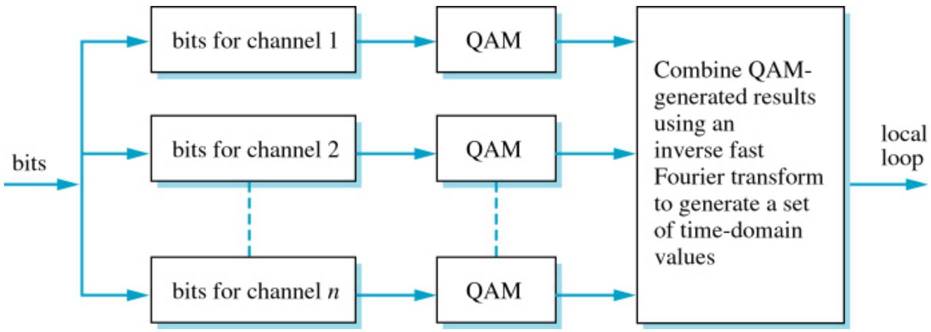
The splitter has two filters. The low frequency signal is routed to the phone set/network and the higher frequency part is routed to the PC/Internet



Access Multiplexer: interprets the signal that the user DSL modem create and route data to the internet

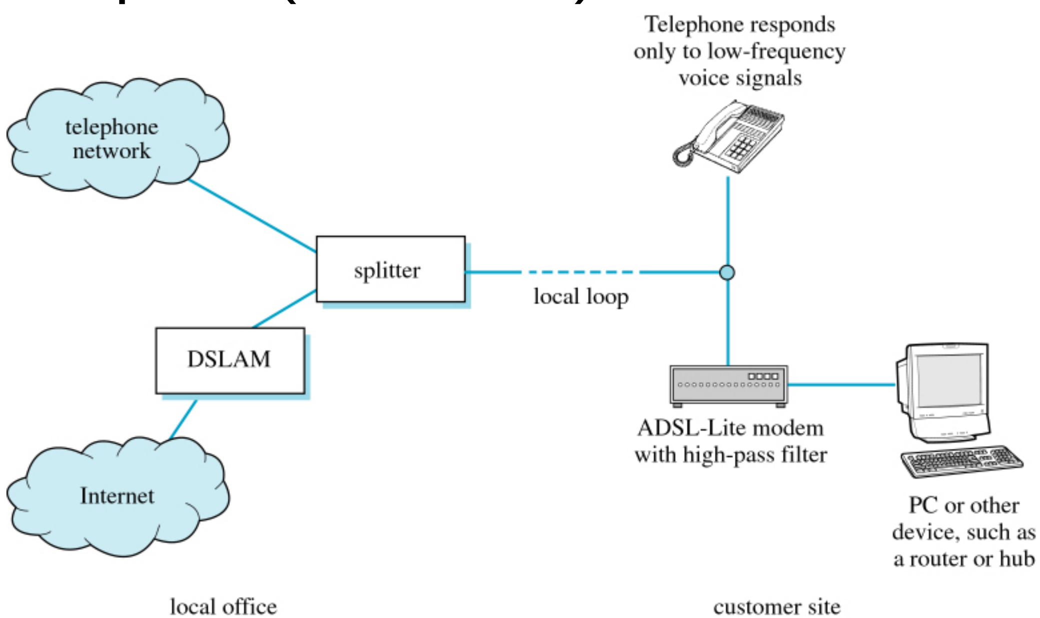
Discrete Multi-Tone (ANSI T1.413)

- The band between 0 and 1.104 MHz is divided into 256 channels
- Lowest 5 are assigned for POTS (21.5 Khz to give guard-band)



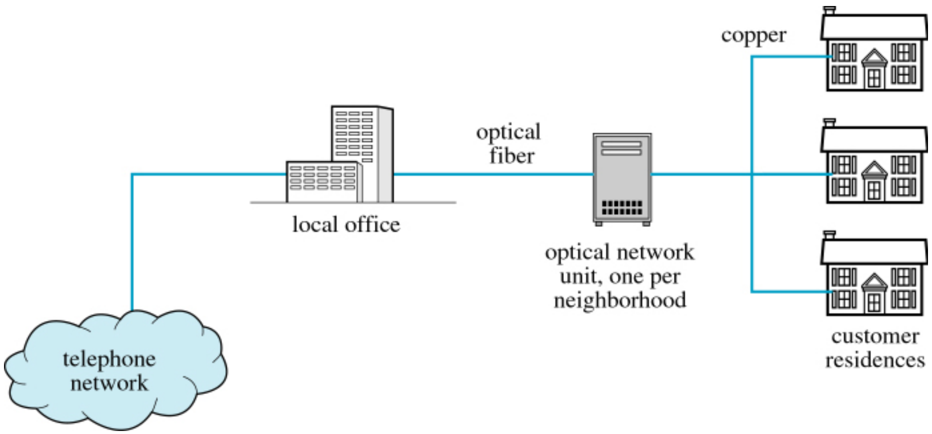
ADSL

- G.Lite (ITU G.992.2)
- Simple
  - Designed for residential customers
  - Easier to install (no splitter at the customer side)
  - Typical speed (1.5-6MHz)



Fiber/Copper Hybrid Local loop

- In order to extend the range and capacity of DSL



Other DSL Technologies

- SDSL (Symmetric DSL)
- HDSL (High rate DSL, no POTS service, 12,000 feet)
- RADSL (Rate Adaptive DSL)
- VDSL (Very high speed DSL: 55Mbsp, 1,000 feet)