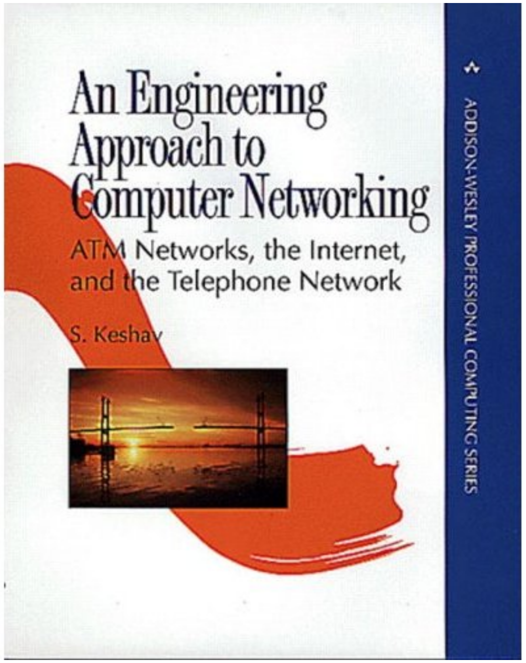
Overview on Telephone Network
- the traditional telecommunication network

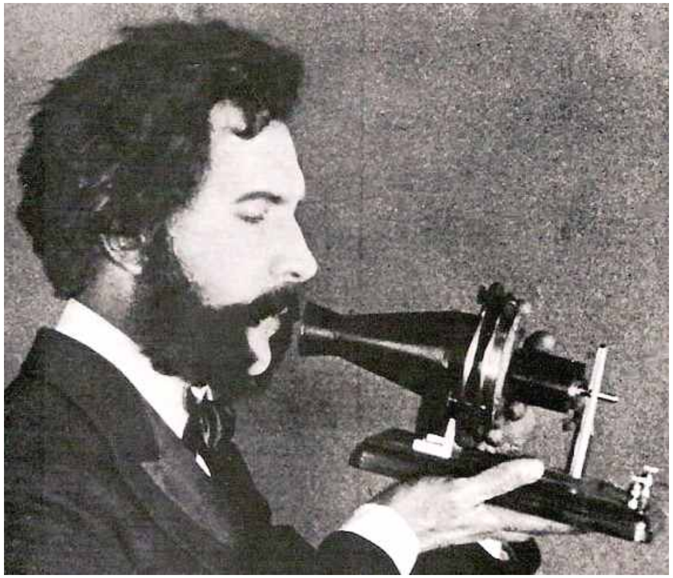


The following slides are largely based on the book: S. Keshav, “An Engineering Approach to Computer Networking”, Addison Wesley.

The Invention of the Telephone

- When?
  - March 10, 1876.

- Who invented the telephone?



Alexander Graham Bell speaking into prototype telephone.

The First Voice Message

Mr. Watson, come here.
I want to see you.

Speaking through the instrument to his assistant, Thomas A. Watson, in the next room, Bell said these famous first words.

Telephone Exchange: 1878

- (1878) Bell set up the first telephone exchange over a manual switchboard

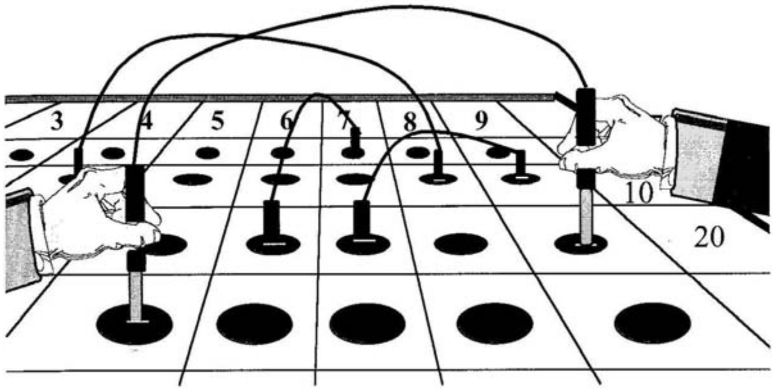
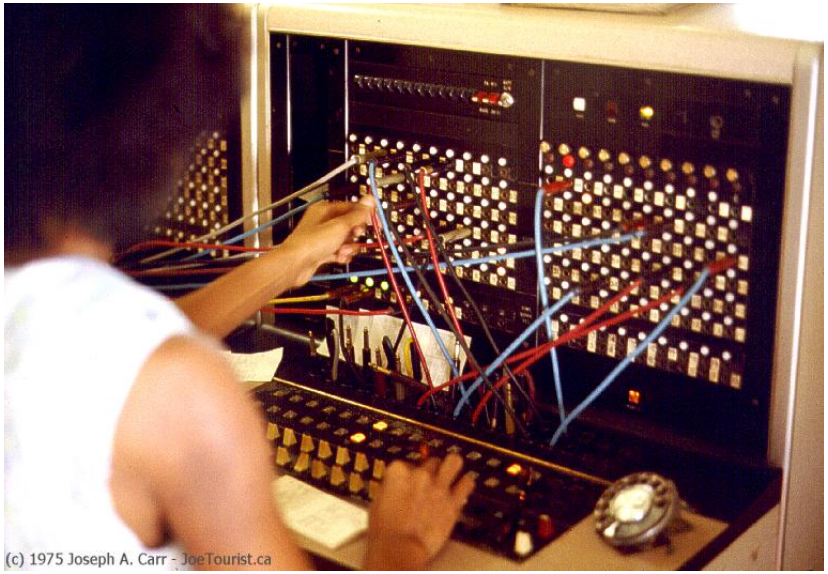


Figure 1.13. A manual switchboard.



A telephone operator manually connected calls with patch cables at a telephone switchboard.

From Analog to Digital (1960's)

- 1962 The first digital T-carrier system was introduced into commercial service by AT&T.
- 1960s Telephone network was the world's dominant communication network.
- Circuit Switching is used:
  - When there is a call, a path from source to destination is set up.
  - The bandwidth of the path is reserved for the calling parties for the whole of the call.

Example: Circuit Switching

- For host *A* to send messages to *B*, the network must reserve one circuit on each of two links.
- Each link can have more than 1 circuit. *How?*

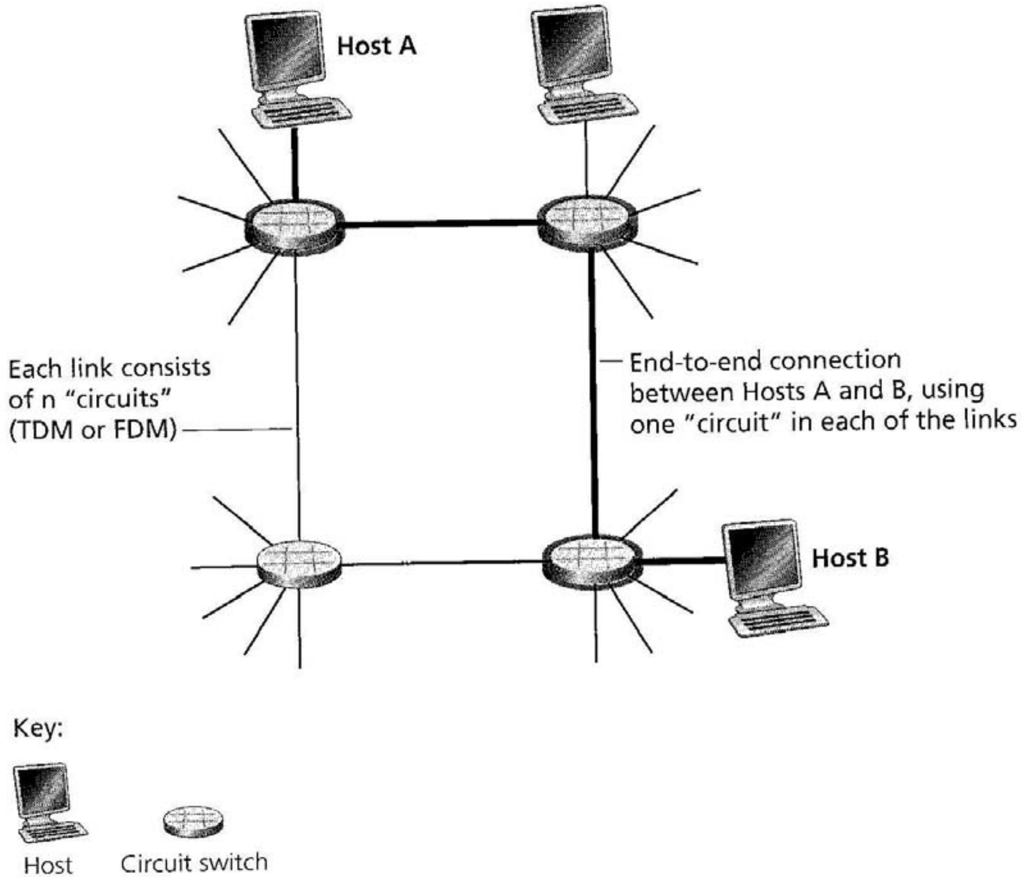
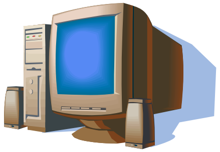


Figure 1.5 ♦ A simple circuit-switched network consisting of four switches and four links

Why Circuit Switching?

- A telephone call typically lasts for a long time
  - it justifies the cost of setting up a circuit before transmission.
- Voice signal is of constant bit rate (e.g. 64 kbps for PCM).
  - Circuit switching simplifies the allocation of bandwidth.

Computer Networks

- 1960s Computers  became important.
- How to connect computers together?
- Is circuit switching appropriate?
  - Data traffic are typically bursty.
  - Typical Scenario:
    - Sending a command to a remote computer
    - A period of inactivity
    - Sending another command
  - Assigning a dedicated channel wastes bandwidth.
- Packet switching was invented for this reason.

Learning objectives

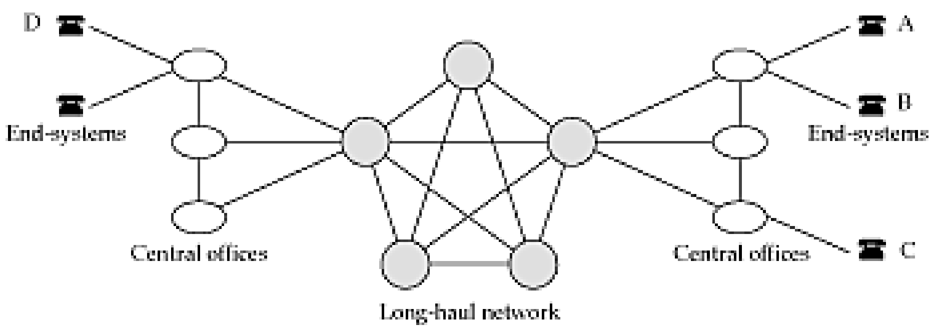
Know

- the simplified view of a typical telephone network
- how to route a call in telephone core network
- the essential issue/problem for routing a call
- features of telephone network routing
- the fundamental concepts for "transmission"
- why switching is needed instead of having a link for each pair of users
- the job of a signaling network
- how the state transition diagram can help a switch controller to decide what action to take according to the incoming signal

Concepts

- Single basic service: two-way voice
  - low end-to-end delay
  - guarantee that an accepted call will run to completion
- Endpoints connected by a *circuit*
  - like an electrical circuit
  - signals flow both ways (*full duplex*)
  - associated with bandwidth and buffer *resources*

The big picture



- Fully connected core
  - simple routing
  - telephone number is a hint about how to route a call
  - hierarchically allocated telephone number space

The basic elements

1. Routing
2. Switching
3. Transmission
4. Signaling

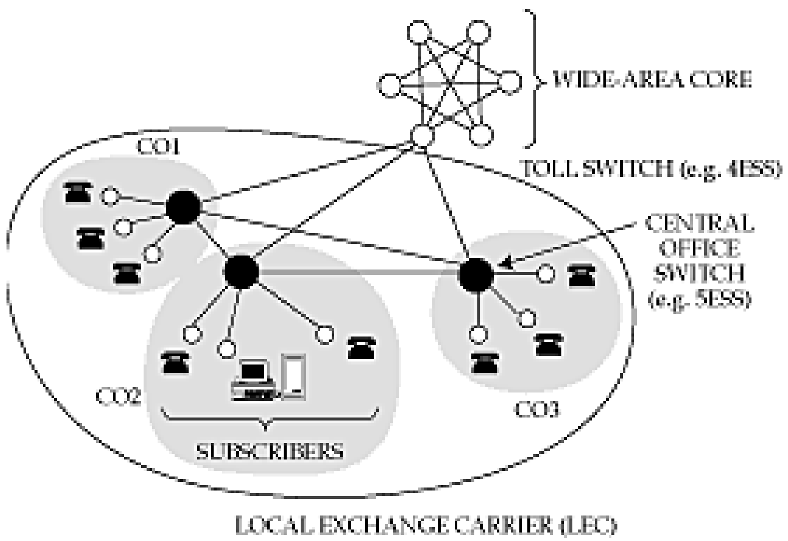
Two Key Network Functions

- *routing:* determine route taken by data from source to destination
  - *routing algorithms*
- *switching:* move data from switch's input to appropriate switch's output

analogy: **Traveling**

- *routing:* process of planning trip from source to destination
- *switching:* process of getting through single interchange

1. Routing: Telephone network topology



- 3-level hierarchy, with a fully-connected core
- AT&T: 135 core switches with nearly 5 million circuits
- Local Exchange Carriers (LEC) may connect to multiple cores

Routing algorithm

- If endpoints are within same Central Office (CO), directly connect
- If call is between COs in same LEC, use one-hop (or the shortest) path between COs
- Otherwise send call to one of the cores
- Only major decision is at core/toll switch
  - one-hop or two-hop path to the destination toll switch
  - (why don't we use paths with more than two hops?)
- Essence of problem/issue
  - which two-hop path to use if one-hop direct path is full?

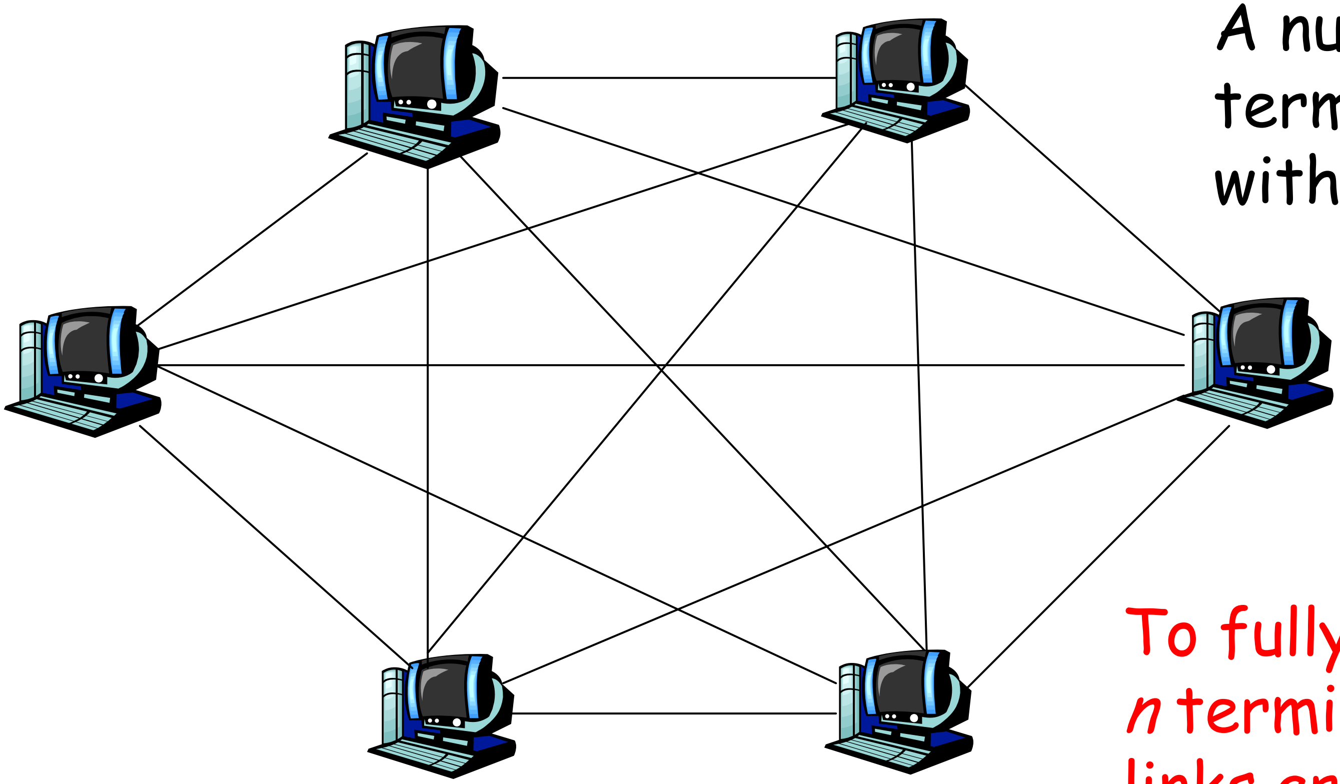
Features of telephone network routing

- Stable load
  - can predict network load throughout the day
  - can choose optimal routes in advance
- Extremely reliable switches
  - downtime is less than a few minutes per year
  - can assume that a chosen route is available
  - can’t do this in the Internet
- Single organization controls entire core
  - can collect global statistics and implement global changes
- Very highly connected network
- Connections require resources (but all need the same)

2. Switching: motivation

- Problem:
  - each user can potentially call any other user
  - can’t have direct lines! (Why?)

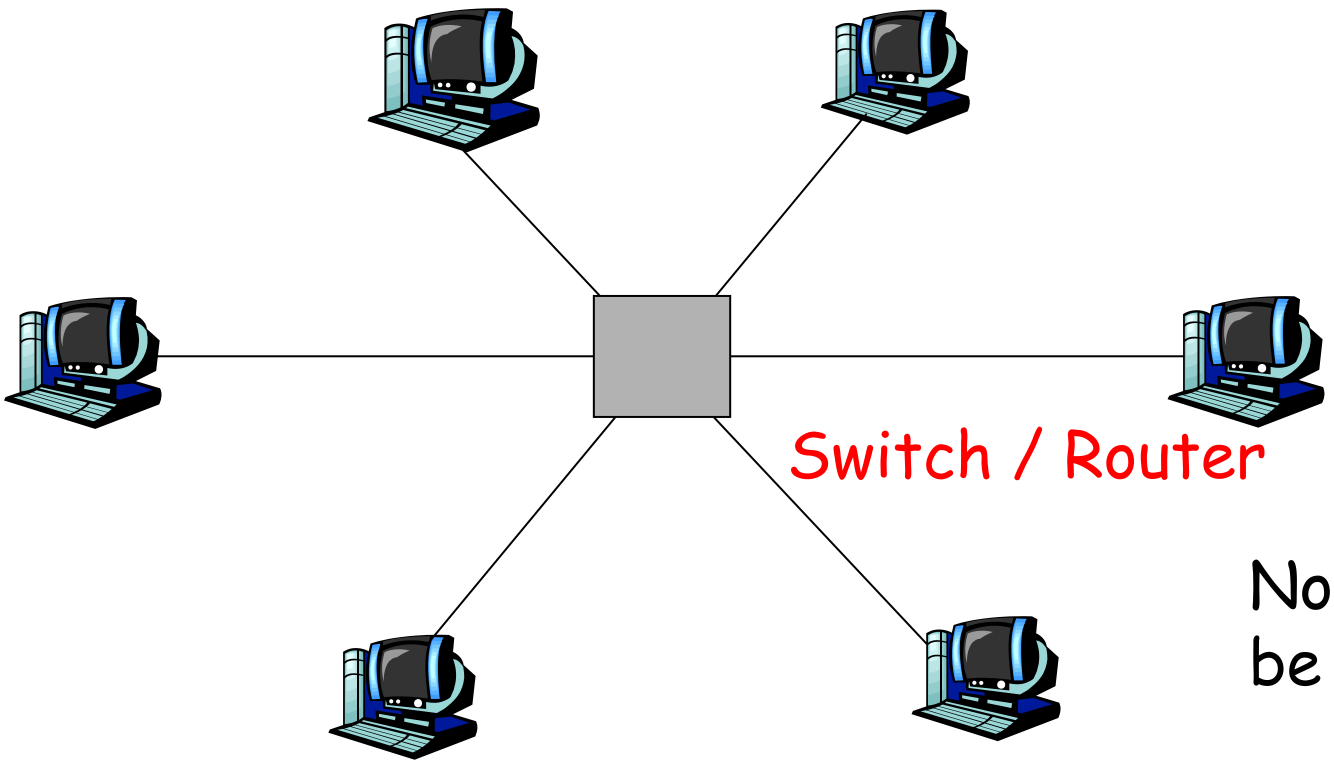
Why switching is needed instead of having a link for each pair of users?



A number of terminals connected with each other

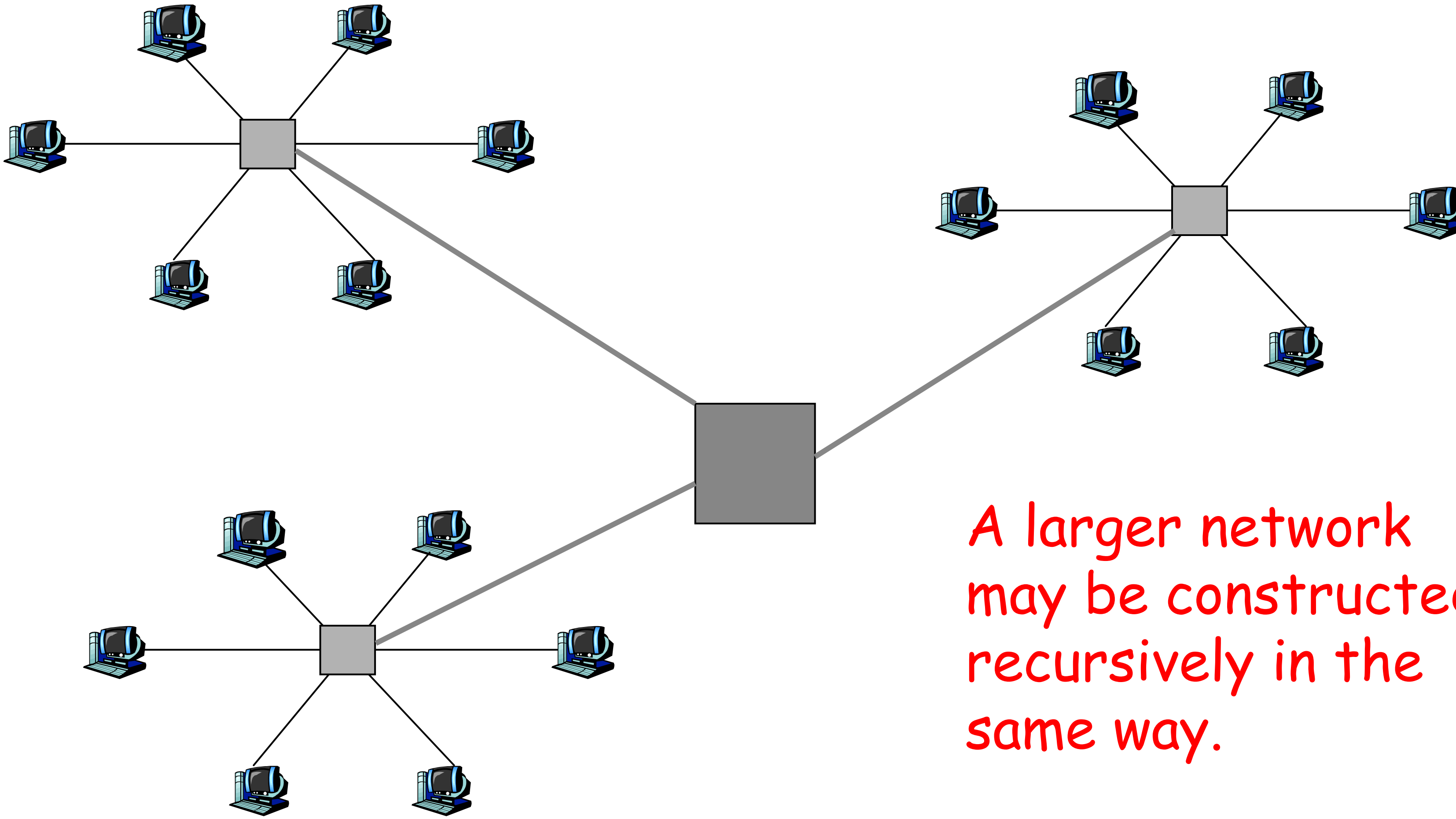
To fully connect *n* terminals, how many links are needed?

Star Topology



No. of links can be reduced

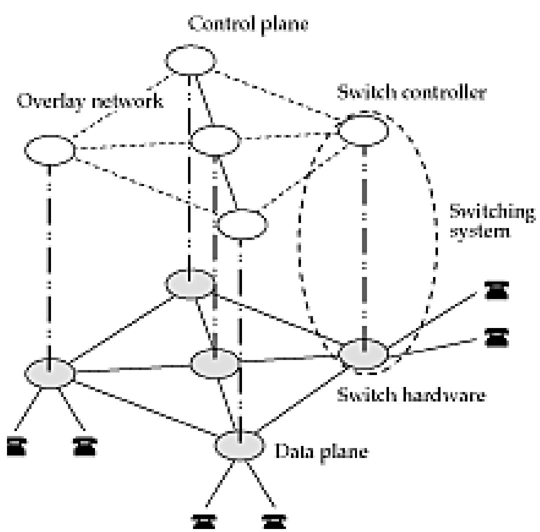
A 2-Tier Network



A larger network may be constructed recursively in the same way.

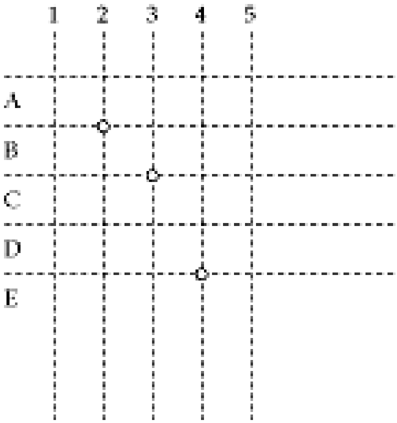
Switching

- Switches establish temporary *circuits*
- Switching systems come in two parts: switch and switch controller



Switching: what does a switch do?

- Transfers data from an input to an appropriate output
- Some ways to switch:
  - *space division*



Switching

- Another way to switch
  - *time division (time slot interchange* or *TSI)*

- To build larger switches we combine space and time division switching elements

3. Transmission

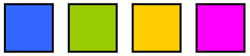
- Link characteristics
  - **information carrying capacity (bandwidth)**
    - information sent as *symbols*
    - 1 symbol >= 1 bit
  - **propagation delay**
    - time for electromagnetic signal to reach other end
    - light travels at 0.7c in fiber ~8 microseconds/mile
    - NY to SF => 20 ms; NY to London => 27 ms

Transmission: Multiplexing

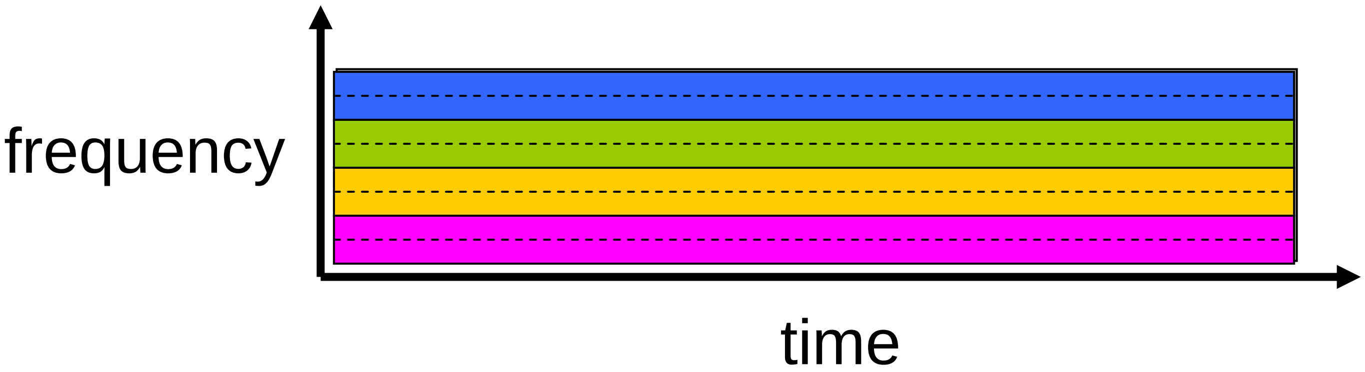
- What is multiplexing?
  - Enabling a number of lower bit rate connections to share a single higher bit rate transmission line.
- Frequency-Division Multiplexing (FDM)
  - Applies to both analog and digital signals
  - e.g. 4 kHz for each analog voice signal
- Time-Division Multiplexing (TDM)
  - Applies only to digital signals
  - e.g. digital voice using pulse-coded modulation (PCM)
    - Sampling: 8 kHz, and Quantization: 8 bits per sample => Bit Rate: 64 kbps

Multiplexing: FDM and TDM

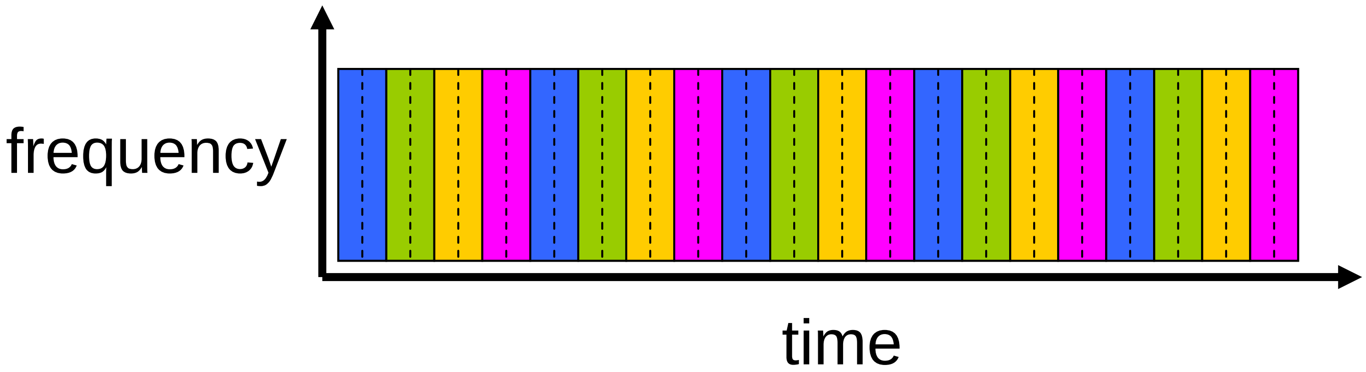
Example:

4 users 

FDM



TDM



Transmission: Multiplexing

- Multiplexed trunks can be multiplexed further
- Need a standard! (why?)
- US/Japan standard is called *Digital Signaling* hierarchy (DS)

Digital Signal Number	Number of previous level circuits	Number of voice circuits	Bandwidth
DS0		1	64 Kbps
DS1	24	24	1.544Mbps
DS2	4	96	6.312 Mbps
DS3	7	672	44.736 Mbps

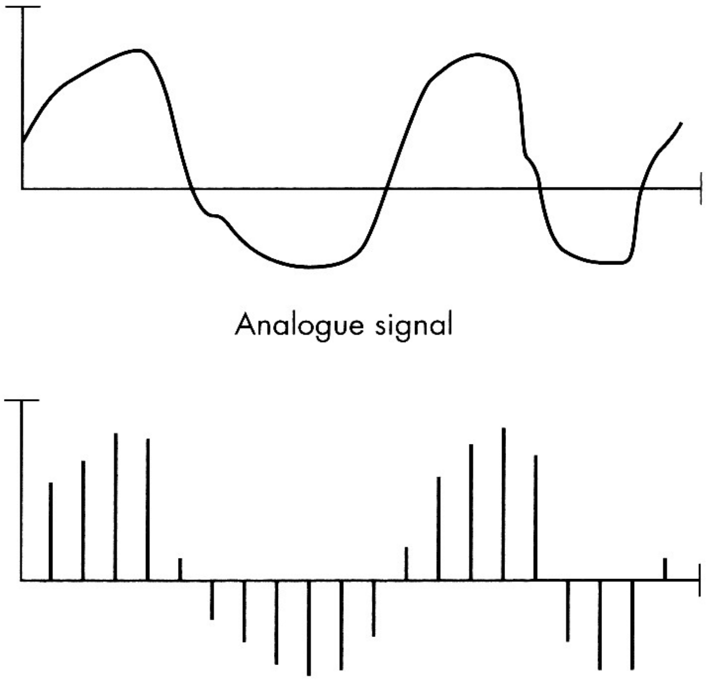
Transmission: Link technologies

- Many in use today
  - twisted pair
  - coax cable
  - terrestrial microwave
  - satellite microwave
  - optical fiber
  - ADSL (Asymmetric Digital Subscriber Lines): the most chosen broadband option in the world (more than 60% of the broadband market)
- Increasing amount of bandwidth and cost per foot
- Popular
  - fiber
  - ADSL

Transmission: Analogue to Digital Conversion

- To represent an infinite precision signal originally in an analogue form by a finite set of numbers at a fixed sample rate
- Two steps:
  - sampling
  - quantization

Transmission: Sampling

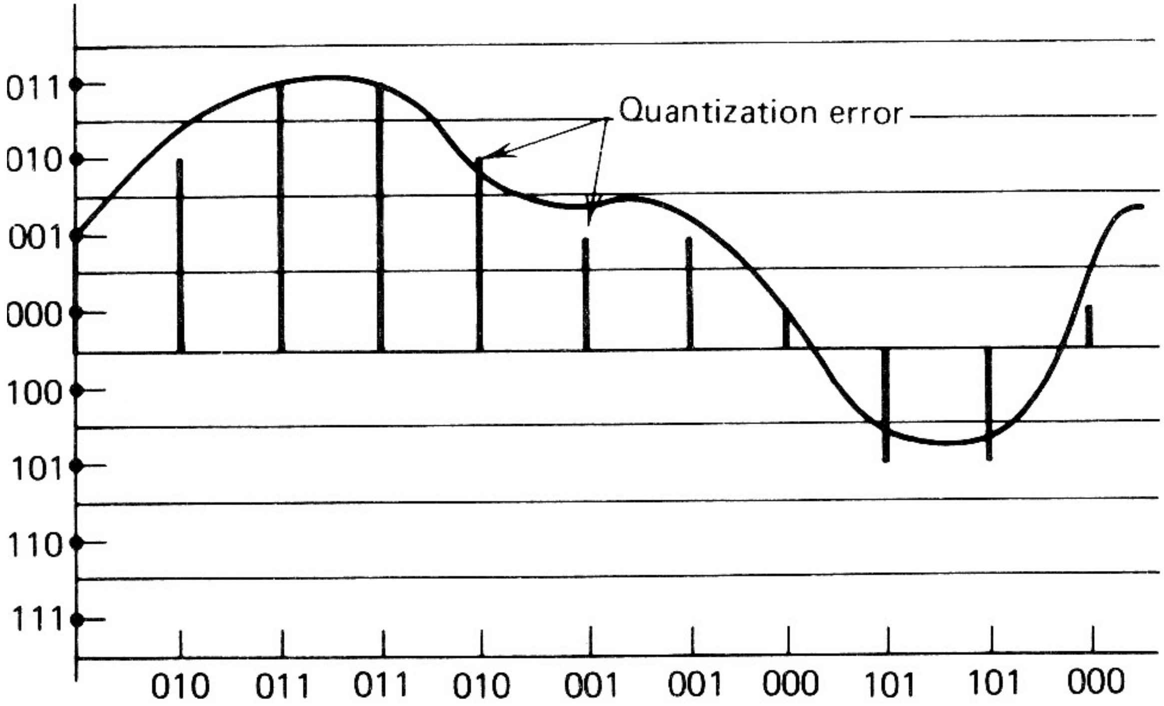


Analogue signal

Sample version of the analogue signal: PAM

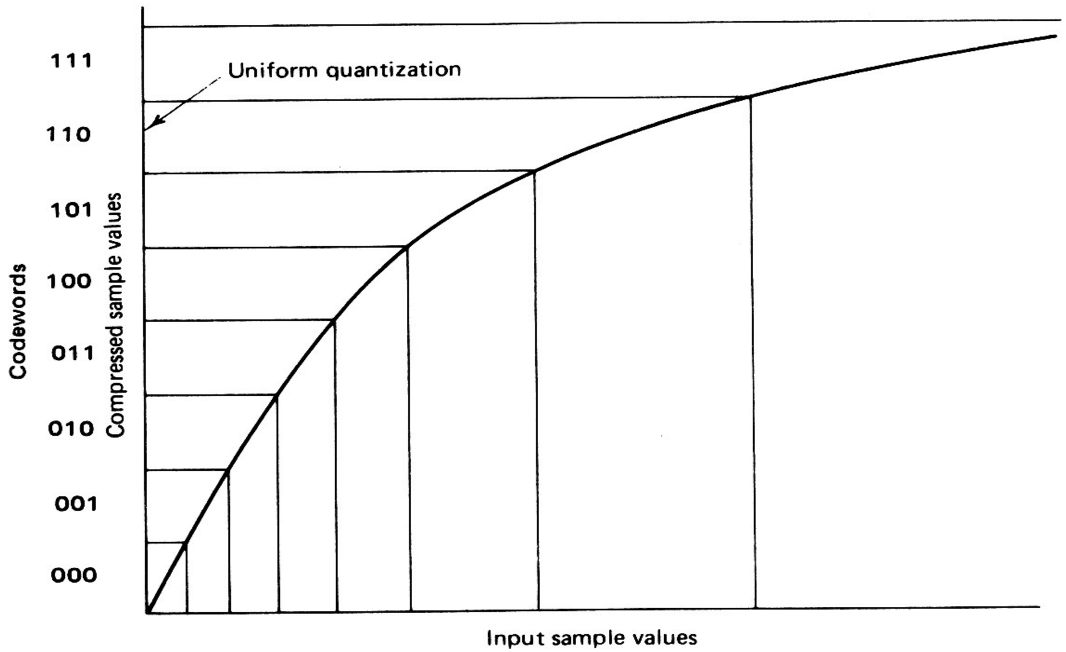
The sampling process leads to the PAM (pulse amplitude modulation) representation of the analogue signal

Transmission: Uniform Quantization



- Samples are quantized to the nearest quantization level
- PAM + quantization $\Rightarrow$ PCM (pulse code modulation)

Transmission: Non-uniform Quantization



- Signal compression + uniform quantization $\Rightarrow$ non-uniform quantization

Transmission: Ways of Compression

- μ law

$$f_\mu(x) = \operatorname{sgn}(x)\frac{\ln(1+\mu|x|)}{\ln(1+\mu)}$$

- A law

$$f_A(x) = \operatorname{sgn}(x)\frac{A|x|}{1+\ln(A)} \qquad 0 \le |x| \le \frac{1}{A}$$

$$f_A(x) = \operatorname{sgn}(x)\frac{1+\ln|Ax|}{1+\ln(A)} \qquad \frac{1}{A} \le |x| \le 1$$

Transmission: Voice Coding

- To represent the digitized voice signal at a reduced bit rate for narrowband transmission and digital storage devices with limited capacity

Transmission: Codecs in ITU standards

- G.711
  - approved in 1965
  - PCM, μ law or A law
  - 8000 sample per second
  - each sample is encoded as an octet
  - 64 kb/s

- G.722
  - approved in 1988
  - provides a higher quality of digital encoding of 7 kHz of audio spectrum
  - support a number of rates: 48, 56 or 64 kb/s, using SB-ADPCM (subband - adaptive differential PCM)
  - good for all professional conversational voice applications, but musical applications are not recommended

- G.726
  - approved in 1990
  - rates in 16, 24, 32 or 40 kb/s, using ADPCM (adaptive differential PCM)
  - the quality at 32 kb/s is taken as a reference for toll quality
- G.728
  - approved in 1992-94
  - 16 kb/s, using LD-CELP (low delay, code-excited linear prediction)
  - quality similar to G.726

- G.723.1
  - approved in 1995
  - two modes of operation
    - 6.4 kb/s, using MP-MLQ (multipulse-maximum likelihood quantization)
    - 5.3 kb/s, using ACELP (algebraic-code-excited linear prediction)
  - has a voice activity detection, discontinuous transmission, comfort noise generation capability
  - 1.1 kb/s during silence period

- G.729
  - approved in 1995
  - 8 kb/s, using ABS CS-ACELP (analysis by sythesis, conjugate structure – ACELP)
  - there is a low-complexity version G.729A, which is sometimes used in VoIP systems

Transmission: Codecs from ETSI (Europe)

- GSM 06.10
  - Approved in 1988
  - 13 kb/s, using RPE-LTP (regular pulse excitation – long term prediction)
  - used in cellular mobile system
- GSM 06.60
  - Approved in 1996
  - 12.2 kb/s, using ACELP

Transmission: Codec from IETF

- iLBC (internet Low Bitrate Codec)
  - 13.33 kb/s, LPC and block based coding of the LPC residual signal using an adaptive codebook
  - basic quality higher than G.729A, *high robustness to packet loss*
  - computational complexity in the range of G.729A
  - royalty free codec
  - http://www.ilbcfreeware.org/

4. Signaling

- Recall that a switching system has a switch and a switch controller
- Switch controller is in the *control* plane
  - does not touch voice samples
- Manages the network
  - call routing (collect *dialstring* and forward call)
  - alarms (trigger the ring at receiver)
  - billing
  - directory lookup (for 800/888 calls)

Signaling network

- Switch controllers are special purpose computers
- Linked by their own internal computer network
  - *Common Channel Interoffice Signaling (CCIS) network*
- Messages on CCIS conform to *Signaling System 7 (SS7)* spec.

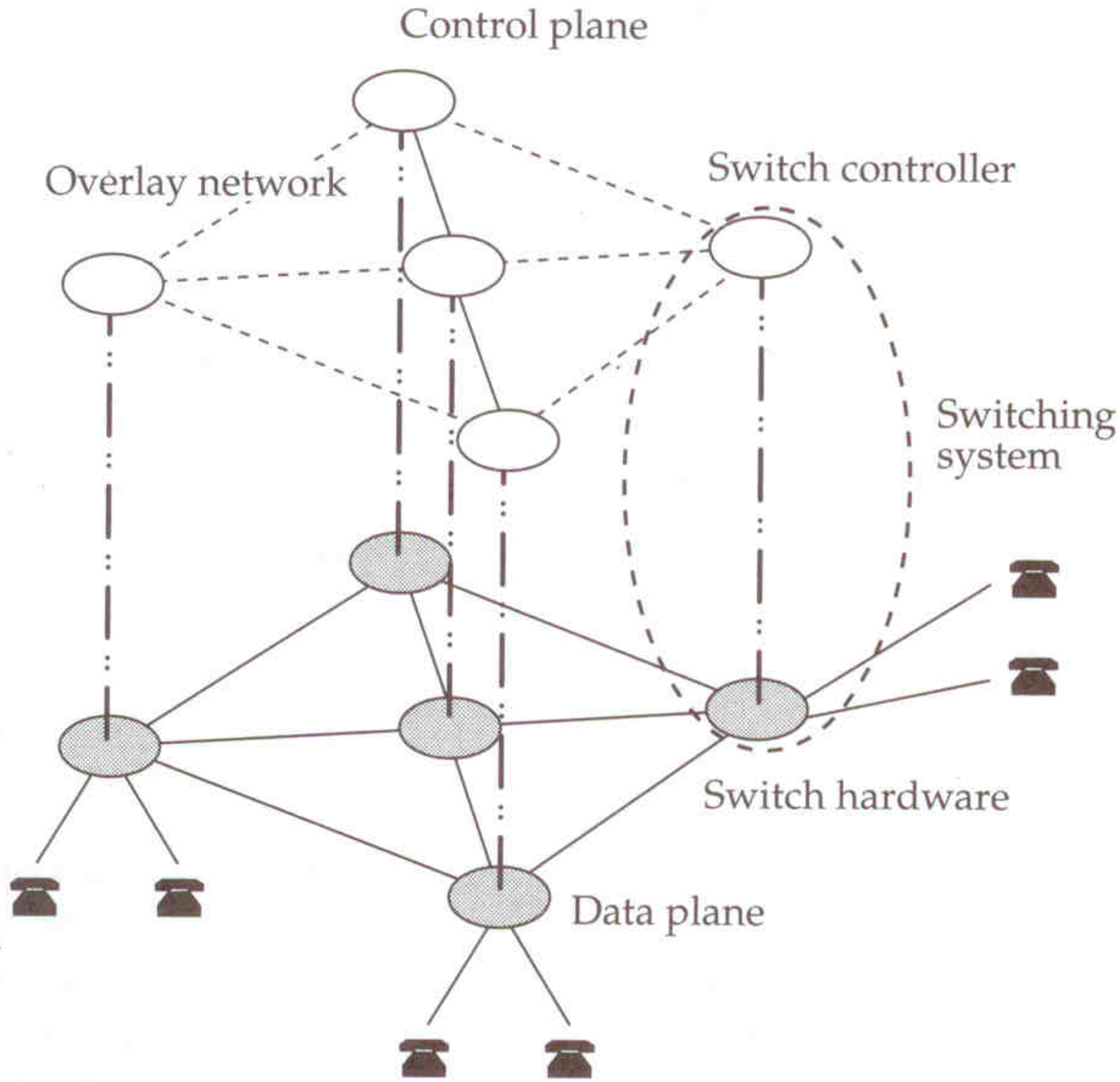
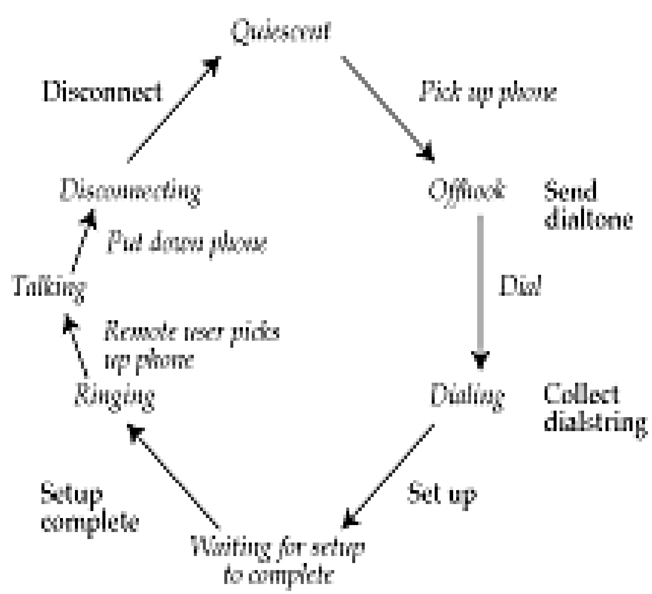


Figure 2.6: Switching fabric and switch controller. The switching fabric carries voice. The switch controllers form a logical network for setting up voice calls.

Signaling

- One of the main jobs of switch controller: keep track of *state* of every endpoint and take action according to the incoming signal
- Key is *state transition diagram*



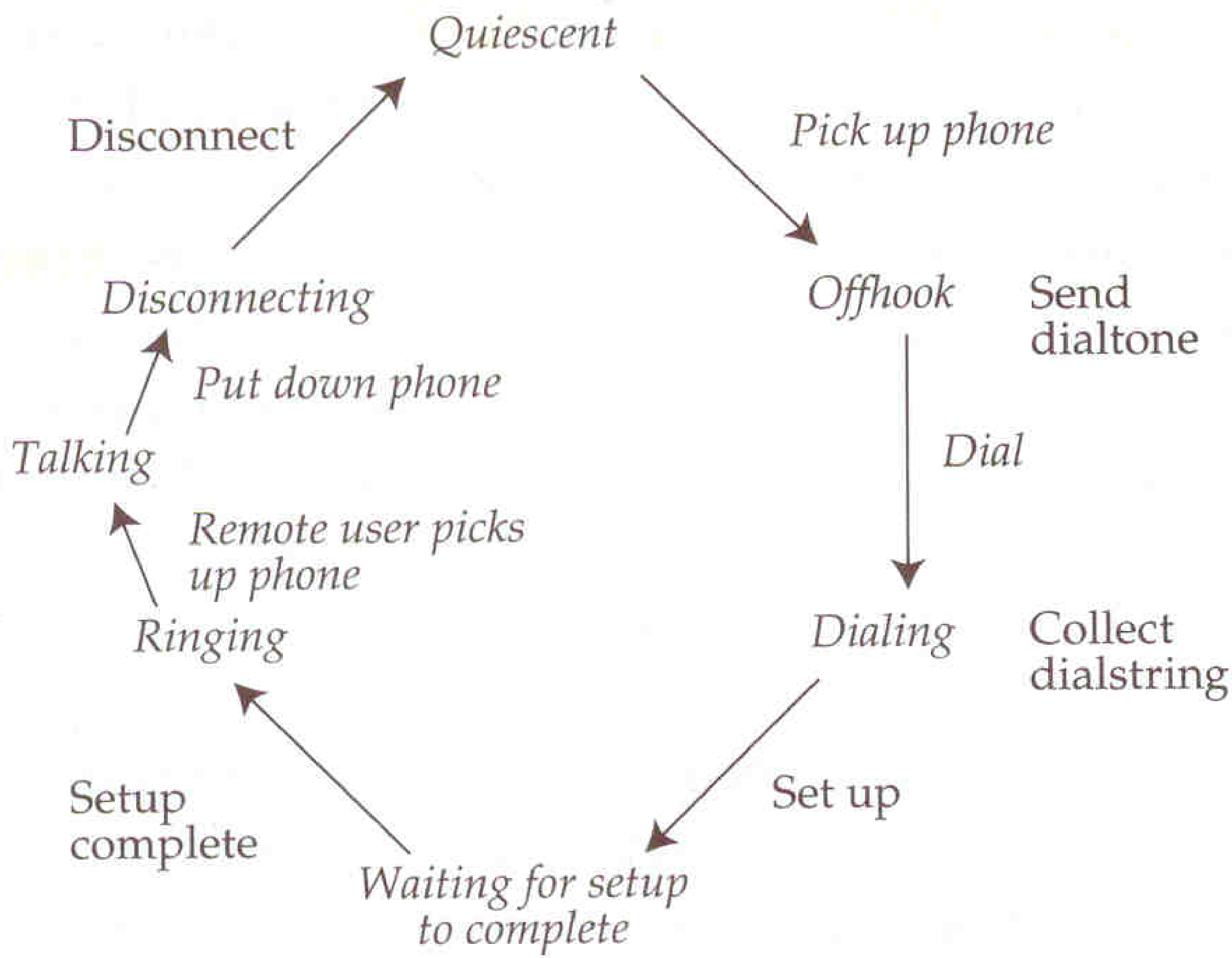


Figure 2.9: Simplified state diagram of a call at an originating switch. Actions by a user (in italics) or a switch controller (in regular font) cause the call to change state.

Q & A