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.

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

Figure 1.13. A manual 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

Switch / Router

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*

![Diagram of switch with labeled inputs and outputs]
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 $\geq 1$ bit
  - propagation delay
    - time for electromagnetic signal to reach other end
    - light travels at 0.7c in fiber $\sim 8$ microseconds/mile
    - NY to SF $\Rightarrow 20$ ms; NY to London $\Rightarrow 27$ 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)

<table>
<thead>
<tr>
<th>Digital Signal Number</th>
<th>Number of previous level circuits</th>
<th>Number of voice circuits</th>
<th>Bandwidth</th>
</tr>
</thead>
<tbody>
<tr>
<td>DS0</td>
<td>1</td>
<td>1</td>
<td>64 Kbps</td>
</tr>
<tr>
<td>DS1</td>
<td>24</td>
<td>24</td>
<td>1.544Mbps</td>
</tr>
<tr>
<td>DS2</td>
<td>4</td>
<td>96</td>
<td>6.312 Mbps</td>
</tr>
<tr>
<td>DS3</td>
<td>7</td>
<td>672</td>
<td>44.736 Mbps</td>
</tr>
</tbody>
</table>
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

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

- \( \mu \) law

\[
f_{\mu}(x) = \text{sgn}(x) \frac{\ln(1 + \mu|x|)}{\ln(1 + \mu)}
\]

- A law

\[
f_A(x) = \text{sgn}(x) \frac{A|x|}{1 + \ln(A)} \quad 0 \leq |x| \leq \frac{1}{A}
\]

\[
f_A(x) = \text{sgn}(x) \frac{1 + \ln|Ax|}{1 + \ln(A)} \quad \frac{1}{A} \leq |x| \leq 1
\]
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 synthesis, 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/](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