2 Encoding and Transmission

Chapter Objectives

Various technologies are required in order to transmit data. These technologies include converting data into signals which can be easily transmitted, and securing the timing between the parties involved in the communication.
This chapter will provide an overview of the meanings, the mechanisms and characteristics of transmission technologies.

① Understanding the modulation and encoding techniques for converting data into transmittable signals.
② Understanding the mechanisms of error handling and synchronous control that are necessary to ensure correct transmission.
③ Understanding multiplexing methods and compression and decompression methods used to ensure efficient use of communication lines.
④ Understanding the types of lines used for transmission and the mechanisms of switching systems.

Introduction

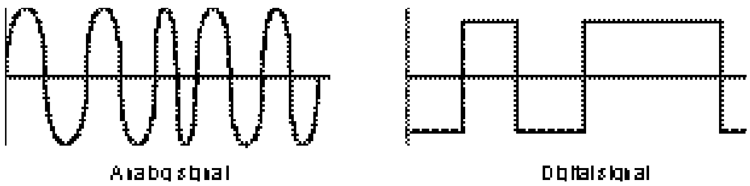
A physical communication line is necessary to transmit data from the sender to the recipient in a network. The type of communication line determines the kind of signals that can flow along the line. Consequently, it is necessary to have a mechanism that converts the data to the transmittable signals in accordance with the physical communication lines.

2.1 Modulation and Encoding

As explained in the foreword to this chapter, the techniques for data conversion are called "modulation" and "encoding." These two methods are used to transform the data into signals that can be transmitted. There are two types of convertible signals:

- Analog signals: Signals with a continuous waveform, such as audio and radio waves.
- Digital signals: Signals made up of discontinuous (discreet) pulses, and used inside computers.

Figure 2-1-1 Analog and digital signals



2.1.1 Communication Lines

A communication line is the physical transmission channel actually used for transmission of signals. These lines are broadly divided into analog lines and digital lines in accordance with the kind of signals that they can carry.

(1) Analog line

Analog lines are communication lines for transmission of analog signals. Analog signals are waveform signals, and audio signals are a typical analog signal type. Public telephone networks designed for transmission of audio signals represent the most widely used analog lines.

(2) Digital line

Digital lines are communication lines for transmission of digital signals. Digital signals are the kind of signals that are used inside computers. Digital lines for transmitting this kind of signals are lines designed for data communications. ISDN lines (explained later) are representative of digital lines.

2.1.2 Modulation Technique

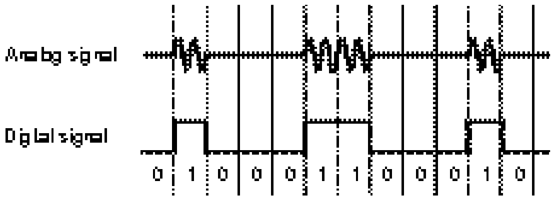
When transmitting data using an analog line, the computer's digital signals must be converted to analog signals using a MODEM (modulator/demodulator (explained later)). This is called "modulation" (the opposite is called "demodulation.")
Three methods are typically used for modulation in a MODEM:

- Amplitude modulation
- Frequency modulation
- Phase modulation

(1) Amplitude modulation (AM)

Amplitude modulation is a method in which the analog signal output is turned ON and OFF in accordance with ON (1) or OFF (0) state of the digital signal. This method is susceptible to noise; but it is the simplest modulation method, and uses narrow frequency band for effective utilization of transmission bandwidth.

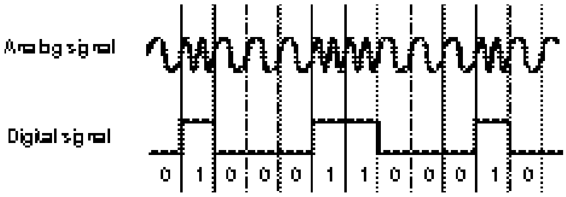
Figure 2-1-2
AM method



(2) Frequency modulation (FM)

Frequency modulation is a method which modulates the ON (1) and OFF (0) states of digital signals into two frequencies in different bands.
The drawback of this technique is that the required frequency band is wide but the method ranks as the second simplest method following the amplitude modulation method. It is also resistant to noise, etc.

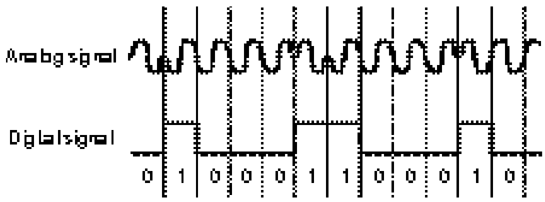
Figure 2-1-3
FM method



(3) Phase modulation (PM)

Phase modulation is a method in which the phase of the carrier is shifted to represent the ON (1) or OFF (0) states of the digital signal.
The simplest method is the 180-degree shifting method in which the phase is inverted when the digital signal is ON (1) and the carrier is output as it is prior to modulation when the signal is OFF (0).
This method is resistant to noise and allows much information to be sent simultaneously.

Figure 2-1-4
PM method



2.1.3 Encoding Technique

(1) PCM

When transmitting data using a digital line, it is necessary to convert analog signals, such as audio, to digital signals. This is called "encoding." PCM (Pulse Code Modulation) is a technique used for encoding.

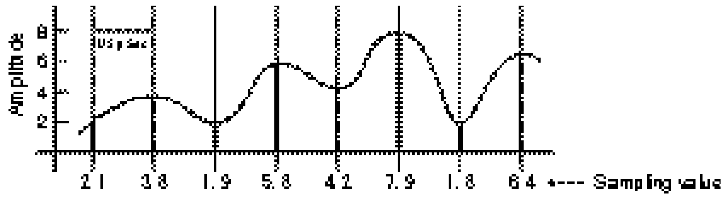
(2) Encoding procedures

The procedures involved in encoding (digitizing) analog signals, like audio signals, and sending these to another party are:
Sampling → Quantization → Encoding
On the receiver side, this process is reversed to obtain analog signals.

① Sampling

The sampling theorem (Shannon's theorem) is an important part of sampling. This theorem states "if the highest frequency of the target analog signal is "f," the recipient can restore the original analog signal if the signal is sampled at a frequency of 2f or higher for transmission."

Figure 2-1-5
Sampling

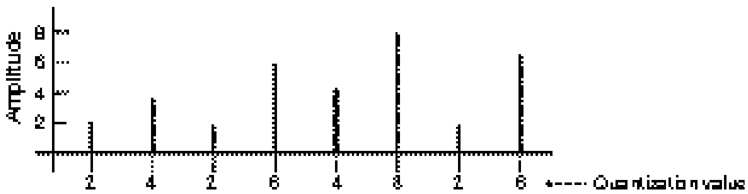


Example 300 - 4,000 Hz audio signal
As the highest frequency is 4,000 Hz, it is enough to sample the signal at 8,000 Hz according to Shannon's theorem. In other words, if 8,000 oscillations are performed per second, this audio signal will oscillate at the frequency of 125 μ (micron) second.

② Quantization

Quantization rounds the value of a measured signal to a finite number by rounding down or rounding up.

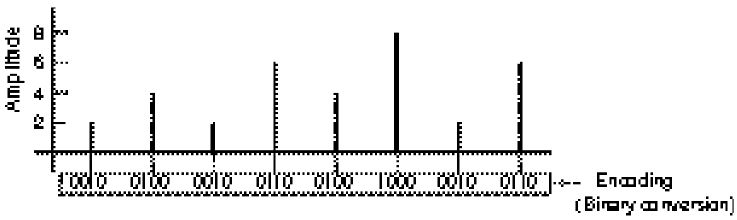
Figure 2-1-6
Quantization



③ Encoding

Encoding encodes the integral numbers obtained by quantization.

Figure 2-1-7
Encoding



Example Transmission speed when a signal sampled at 8,000 Hz is transmitted using 8-bit codes
As 8 bits must be sent every 125 μ sec, i.e., an 8-bit code must be sent 8,000 times per second, the transmission speed becomes
8 bits × 8,000/sec = 64,000 bps

(3) ADPCM (Adaptive Differential PCM)

ADPCM is a method that employs the PCM technique for audio compression.
ADPCM samples audio waves in the same manner as PCM, but it compresses encoding data by changing the quantization width in accordance with the differences in samples. When using the conventional PCM method, the line transmission capacity must be 64 kbps to enable transmission of audio data. Since this can be accomplished with 32-kbps lines with the ADPCM, this method has been adapted for use in PHS (Personal Handyphone System).

2.2 Transmission Technology

Many transmission technologies are employed to ensure reliable and correct transmission.
Some of these are:

- Conversion of analog signals and digital signals when exchanging data between computers using a communication line. → "Modulation, demodulation"
- Transmission accuracy → "Bit error detection"
- Timing control for data exchange → "Synchronization"
- Techniques for effective and economical use of communication lines → "Multiplexing," "Compression, decompression"

Modulation and demodulation have already been explained, and the following explains other transmission technologies.

2.2.1 Error Control

In data transmission it is necessary to establish countermeasures to prevent bit errors caused by electromagnetic induction, etc.
Two representative error control methods are:

- Parity check
- CRC

One error-correcting system is the family of codes called:

- Hamming code

(1) Parity check

The parity check technique is a method for bit error detection in which an additional bit for detection (called the parity bit) is appended to the bit string to be transmitted. Upon reception, the receiver side references the bit string and the parity bit (Figure 2-2-1).
There are two methods for appending the parity bit.

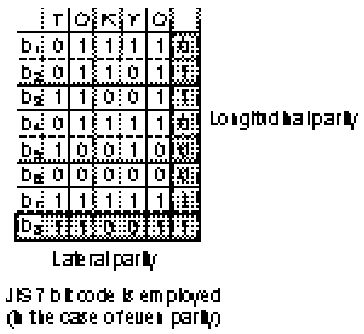
- Odd parity: 1 or 0 is appended to make the number of 1s in each set of bits odd.
- Even parity: 1 or 0 is appended to make the number of 1s in each set of bits even.

The two check methods are:

- Lateral parity check: Lateral inspection of the bit strings making up the characters.
- Longitudinal parity check: Longitudinal inspection of the bit strings making up the data block.

Normally, both methods are used in combination.

Figure 2-2-1
Parity check techniques



(2) CRC (Cyclic Redundancy Check)

The CRC is a transmission method that judges the data strings using a polynomial expression, and appends a check data (CRC code), which is a remainder calculated using an arithmetic operation called "modulo," to the data.

Figure 2-2-2 shows an example of CRC calculation.

This method is suitable for detecting burst (continuous) errors.

Figure 2-2-2 CRC calculation method (CRC-ITU-TS)

① Transmission data characters 'TY' → '01010100 01011001'

② Polynomial expression of ① (K) = $0 \cdot X^{15} + 1 \cdot X^{14} + 0 \cdot X^{13} \cdots\cdots + 0 \cdot X^{1} + 1 \cdot X^{0}$

$= X^{14}+X^{12}+X^{10}+X^{6}+X^{4}+X^{3}+1$

③ Generating polynomial G = $X^{16}+X^{12}+X^{5}+1$ (decided in advance)

④ ② is multiplied by the highest order of ③ (X^{16})

$K' = X^{30}+X^{28}+X^{26}+X^{22}+X^{20}+X^{19}+X^{16}$

⑤ The first 16 bits of K' are inversed..

$K' = X^{31}+X^{29}+X^{27}+X^{25}+X^{24}+X^{23}+X^{21}+X^{18}+X^{17}$

⑥ ⑤ is divided by ③ to find the remainder.

$$
\begin{array}{l}
\qquad\qquad\qquad\qquad X^{15}+X^{13}+X^{8}+X^{7}+X^{5}+X^{3} \\
X^{16}+X^{12}+X^{5}+1 \;\big)\; X^{31}+X^{29}+X^{27}+X^{25}+X^{24}+X^{23}+X^{21}+X^{18}+X^{17} \\
\qquad\qquad\qquad\qquad X^{31}+X^{27}+X^{20}+X^{15} \\
\qquad\qquad\qquad\qquad X^{29}+X^{25}+X^{24}+X^{23}+X^{21}+X^{20}+X^{18}+X^{17}+X^{15} \\
\qquad\qquad\qquad\qquad X^{29}+X^{25}+X^{18}+X^{13} \\
\qquad\qquad\qquad\qquad X^{24}+X^{23}+X^{21}+X^{20}+X^{17}+X^{15}+X^{13} \\
\qquad\qquad\qquad\qquad X^{24}+X^{20}+X^{13}+X^{8} \\
\qquad\qquad\qquad\qquad X^{23}+X^{21}+X^{17}+X^{15}+X^{8} \\
\qquad\qquad\qquad\qquad X^{23}+X^{19}+X^{12}+X^{7} \\
\qquad\qquad\qquad\qquad X^{21}+X^{19}+X^{17}+X^{15}+X^{12}+X^{8}+X^{7} \\
\qquad\qquad\qquad\qquad X^{21}+X^{17}+X^{10}+X^{5} \\
\qquad\qquad\qquad\qquad X^{19}+X^{15}+X^{12}+X^{10}+X^{8}+X^{7}+X^{5} \\
\qquad\qquad\qquad\qquad X^{19}+X^{15}+X^{8}+X^{3} \\
\qquad\qquad\qquad\qquad X^{12}+X^{10}+X^{7}+X^{5}+X^{3}
\end{array}
$$

⑦ Finding the remainder.

$R = X^{12}+X^{10}+X^{7}+X^{5}+X^{3}$ = 0001010010101000

⑧ On the sender side, ⑦ is appended to ① for transmission.

⑨ On the receiver side, the same calculation is performed. If the result of the calculation matches the remainder added on the sender side, it signifies correct data reception.

(3) Hamming code

Hamming code is a technique in which a redundancy bit, called the Hamming code, is appended for error detection and correction. Using the hamming distance (the bit number that differs in the information bits of the same bit length), the following detection/correction becomes possible.

- If the hamming distance is m+ 1 or longer, m bit error can be detected.
- If the hamming distance is 2n+ 1 or longer, n bit error can be corrected.

Assuming that the transmission data is $(b_4, b_3, b_2, b_1) = (0110)$, the procedure of the error detection of the Hamming code technique becomes as follows:

1. Transmission bits are grouped, and each group is calculated using the modulo 2 operation. The calculated result becomes the check bit (Hamming code) for the respective group.
   $S_1 = b_4 + b_3 + b_2 = 0 + 1 + 1 = 0 \ldots c_1$
   $S_2 = b_4 + b_3 + b_1 = 0 + 1 + 0 = 1 \ldots c_2$
   $S_3 = b_4 + b_2 + b_1 = 0 + 1 + 0 = 1 \ldots c_3$
2. The transmission bit string including the Hamming code is made.
   Transmission bit string $= (b_4, b_3, b_2, c_1, b_1, c_2, c_3)$
   $= (0110011)$
3. On the receiver side, the received bit string is disassembled.
   Received bit string $= (d_7, d_6, d_5, d_4, d_3, d_2, d_1)$
   $= (b_4, b_3, b_2, c_1, b_1, c_2, c_3)$
4. Each group bit (b) includes the Hamming code (c) and is calculated using modulo 2.
   The calculated result is converted to binary notation to identify the error bit.
   - In the case of the received bit string (0100011)

     $s_1 + c_1 = 0 + 1 + 0 + 0 = 1$
     $s_2 + c_2 = 0 + 1 + 0 + 1 = 0$ $\quad (101)_2 = 5 \ldots d_5$ is wrong
     $s_3 + c_3 = 0 + 0 + 0 + 1 = 1$

(4) Bit error rate

The bit error rate is one indicator showing the transmission error rate for transmitted data, and it shows the percentage of errors in the total of transmitted bits.

$$\text{Bit error rate} = \frac{\text{No. of error bits}}{\text{Total number of transmitted bits}}$$

Example A message is transmitted using a line with a bit error rate of 1/500,000. When the transmitted message consists of 100 characters (1 character equals 8 bits), it can be calculated how many messages can be transmitted on an average before a 1-bit error may occur.

① No. of bits in one message
= 100 characters/message × 8 bits/character
= 800 bits/message

② Bit error rate = 1/500,000
→ On an average, a 1-bit error will occur for every 500,000 bits transmitted.

③ Average number of messages before a 1-bit error will occur
= No. of bits before error occurs ÷ No. of characters per message
= 500,000 bits ÷ 800 bits/message
= 625 messages

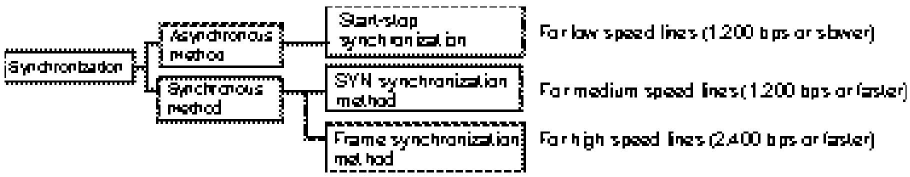
2.2.2 Synchronous Control

When playing catch ball, the thrower yells out and throws the ball after obtaining acknowledgment from the catcher. The one to catch the ball is helped to accomplish this, as he/she has been notified that the ball is to be thrown.

The same principle applies to data transmission. Transmitting the data while synchronizing the timing of the sender and receiver ensures reliable transfer of the data. This is called "synchronization."

Figure 2-2-3 shows the methods available for synchronization.

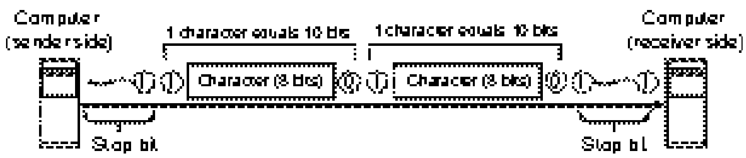
Figure 2-2-3
Types of synchronization



(1) Start-stop synchronization (Asynchronous)

Start-stop synchronization is asynchronous transmission that relies on a start bit (value "0," 1 bit) and a stop bit (value "1," 1 bit, 1.5 bit, 2 bits) being appended to the beginning and the end of each character of the data. When no data is transmitted, a stop bit is sent constantly.

Figure 2-2-4 Start-stop synchronization (example in which the stop bit is 1 bit)



Synchronization is easily achievable using the start-stop synchronization method but since at least 10 bits are required to send one character, the transmission efficiency is poor. Accordingly, this method is used for data transmission at relatively slow speeds (1,200 bps or lower).

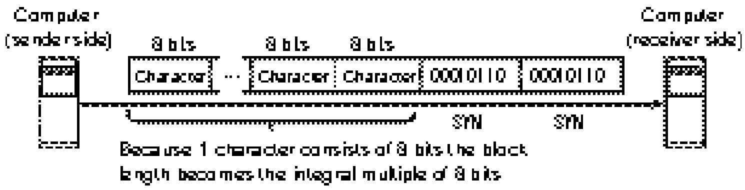
(2) Synchronous method

The synchronous method transmits data after appending a code for synchronizing the character strings of the data. The method is divided into SYN synchronization and Frame synchronization.

① SYN synchronization

The SYN synchronization method is also called the "character synchronization method" as it relies on sending a number of character codes, called SYN, before transmitting data. After synchronization between the sender and the receiver is accomplished with these codes, the data is sent consecutively. The receiver recognizes the SYN code as character data separated by a number of bits (8 bits) for one character.

Figure 2-2-5 SYN synchronization method

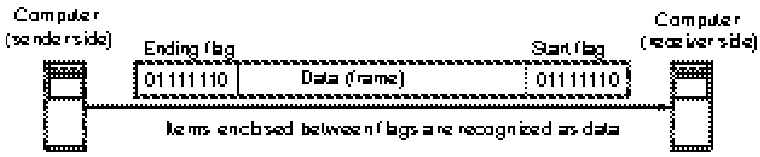


Compared with the start-stop synchronization method SYN synchronization allows data to be sent consecutively which enables efficient data transmission, making this method suitable mainly for transmission at rates of 1,200 bps or higher. However, because there is no code for block ending, the method has the limitation that the block length must be an integral multiple of the bits used for one character.

② Frame synchronization

Frame synchronization accomplishes synchronization by treating the part (frame) surrounded by the flag patterns (bit pattern "01111110") as one unit. This method is also called the "flag synchronization method" because it relies on the flag patterns (flag sequences).

Figure 2-2-6 Frame synchronization method



The sender sends flag patterns incessantly when there is no data for transmission, and when a send request is issued, data is sent following the flag pattern. Conversely, the receiver recognizes the data when bit patterns other than flag patterns are sent, and continues to receive the data until a flag pattern is sent.
Since there are no restrictions on the length of data, this synchronization method is suitable for sending large data loads at relatively high speed.

2.2.3 Multiplexing Methods

Fundamentally, if you have to transmit to "n" number of parties, "n" number of lines are required. However, this is uneconomical. Multiplexing is a technology that was developed to enable communication with multiple parties using just one communication line. In other words, "multiplexing" is a technique in which multiple communications are overlapping on one communication line. Some of the multiplexing methods are:

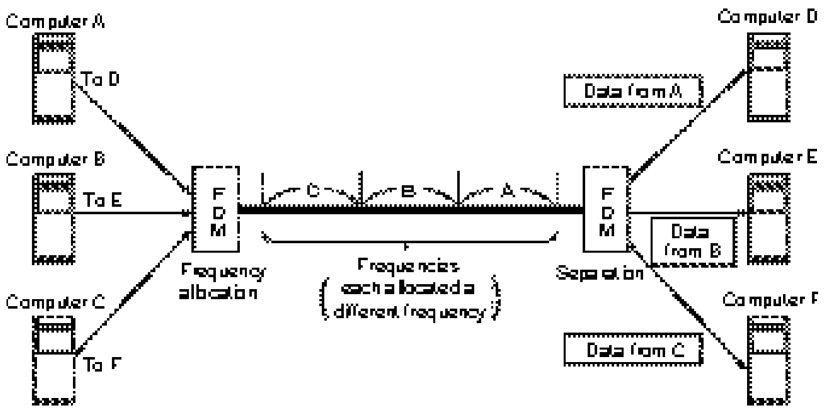
- Frequency division multiplexing (FDM) for multiplexing analog lines
- Time division multiplexing (TDM) for multiplexing digital lines

Other methods include code division multiplexing (CDM) used in mobile communications, and wavelength division multiplexing (WDM) used for transmission with optical fiber cables.

(1) Frequency division multiplexing (FDM)

The FDM (frequency division multiplexing) method transmits using one high-speed analog line by allotting different frequencies to each of several low-speed analog lines. The receiver separates the communication lines for each of the different frequencies and receives data from each of these.

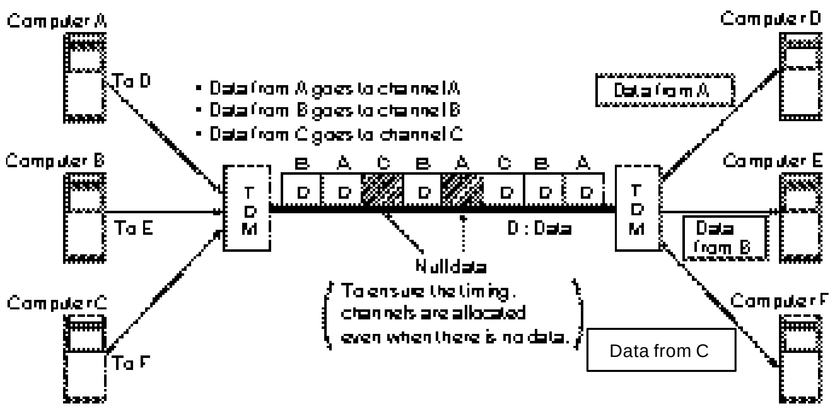
Figure 2-2-7 FDM



(2) Time division multiplexing (TDM)

The TDM (time division multiplexing) method transmits by combining multiple low-speed digital lines into one high-speed digital line. To ensure that the signals of the multiple digital lines are not overlapped, time switch is employed so that each signal is allotted its own fixed time (time slot) during which it is transmitted. Data is transmitted by repeating this process with regularity.
TDM is employed in most multiplexing equipment for digital data.

Figure 2-2-8 TDM



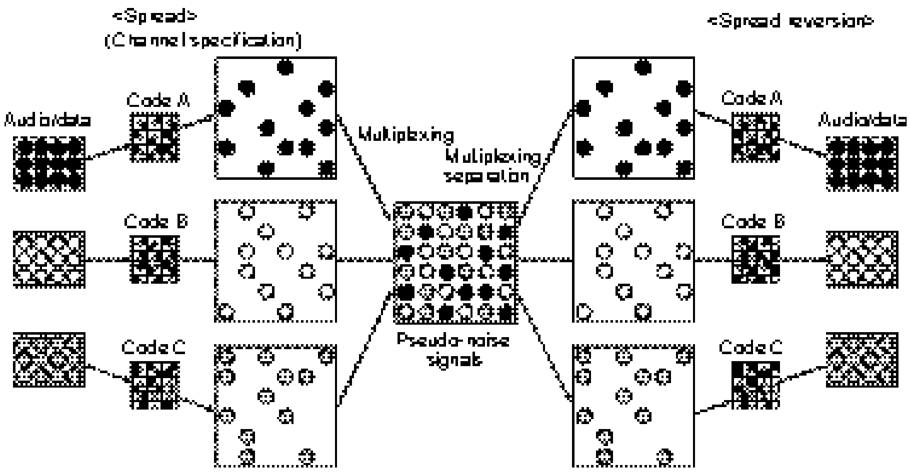
In addition to supporting satellite lines and ISDN, communication systems supporting ATM (explained later) such as B-ISDN have been appearing recently.

(3) Code division multiplexing (CDM)

The CDM (code division multiplexing) method is a multiplexing technology used in mobile communication systems, such as cellular phones. Even though all users use the same frequency, an individual code is allocated to each user to allow communications to each other.

As shown in Figure 2-2-9, inherent PN (Pseudo-Noise) codes are applied to the audio/data of multiple users, and then the system spreads all signals across the same broad frequency spectrum.
The receiver side uses the same PN codes to receive the original audio/data separated out of the pseudo-noise signals of the broad frequency spectrum.
Compared with the FDM or TDM method, each bandwidth can accommodate many channels for use. One of the characteristics of this method is the superior confidentiality obtained because demodulation is impossible without using the same codes as those used at the time of transmission.
Not only does the CDM method allow effective use of frequency bandwidth but it also results in reduced costs for land stations, while it enables high-speed data communication (14.4 kbps or higher). Although research is still being pursued, the commercial deployment of this method have started recently.

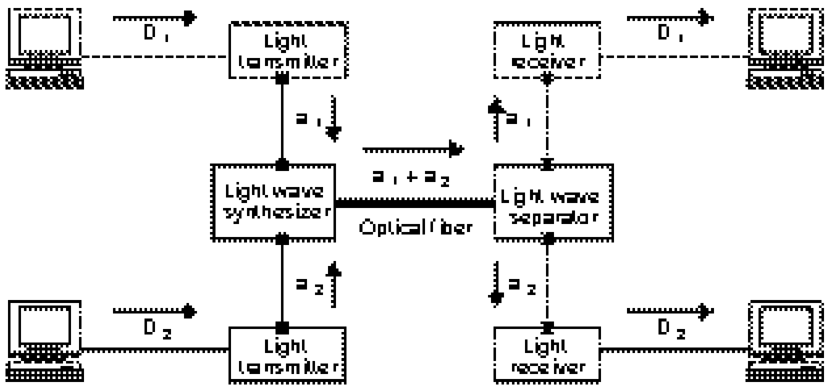
Figure 2-2-9 CDM



(4) Wavelength division multiplexing (WDM)

WDM (wavelength division multiplexing) is a multiplexing communication method used for optical fiber cables (cables that utilize light to transmit data). This method relies on altering the wavelengths of light to allow multiple signals to be transmitted simultaneously on the same fiber cable.
For example, as Figure 2-2-10 shows, when multiple signals (D_1, D_2) are transmitted, each of the signals is converted into separate signals (a_1, a_2) having a different wavelength by light transmitters, and these signals are then combined to a composite wave by a light wave synthesizer. On the receiver side, the light signal transmitted via the optical fiber cable is separate into two signals by a light wave separator, and then sent to the respective destination terminals.

Figure 2-2-10 WDM



At present, because the wavelengths that can be used effectively with optical fiber cables are limited, a method that separates into 4 wavelengths by using 2 cables for upstream and 2 cables for downstream is commonly used.

2.2.4 Compression and Decompression Methods

Previously, the only type of data using in data transmission was simple character data but these days a variety of data, including still images and video, is flowing along the lines. This has resulted in increasing data sizes and increased traffic together with increase in communication costs. When transmitting audio signals digitally, these must be transmitted at a speed of 64 kbps. Consequently, it is extremely important to compress the data to within a range where the original data is not damaged.

Compression of digital data is applied to a variety of data types, such as audio, still images and video (TV pictures), and is especially efficient and beneficial for the large information content and for items demanding high-speed transmission. In the case of TV images, for example, a moving image can be created by sending 30 frames per second, but if these are simply digitalized as they are, at transmission speed of 100 Mbps or more is required to reproduce the same quality. However, detailed analysis of images reveals that the background and other characteristics do not change very frequently. This means that the data that is required to be sent as information is only what is at the front of the image and the parts that have changed from the previous image. The information contents can be reduced considerably by only sending these parts (interframe prediction). Further efficient compression can be accomplished by employing methods (motion compensation) that predict the current position and shape of an object by the movement and shape of the object in frames that preceded the current one by several frames.

For mobile telephone systems in which the available frequency ranges are limited, audio signals can be compressed to 11.2 kbps. By further application of the half rate method, it is possible to compress the signals to 5.6 kbps.

Data compression and decompression methods are explained in the following.

(1) Huffman coding

Huffman coding is compression method developed by D.A. Huffman that replaces frequently occurring characters and data strings with shorter code.

Let us look at an example in which the symbol string R={vuxzvvyyzuvyzvzuyvuu} is encoded. The five types of symbols x, y, z, u, and v occur in the symbol string. In this state, 3 bits are necessary to represent each character using the normal method as shown in Figure 2-2-11. This means that 60 bits are required to represent 20 characters.

Figure 2-2-11 Normal representation method

Character	Bit string
x	000
y	001
z	010
u	011
v	100

Symbol string R

v	u	x	z	v	v	y	y	z	u
100	011	000	010	100	100	001	001	010	011

v	y	z	v	z	u	y	v	u	u
100	001	010	100	010	011	001	100	011	011

Huffman coding allocates specific codes based on the probability of "frequency of occurrence" (a value found by dividing the total number of symbols by the number of times each symbol appears in the symbol string).

In general, in symbol strings formed by M type symbols $\{a_1, a_2, \ldots, a_M\}$, the probability (frequency of occurrence) with which a_i appears is represented as P (a_i). Figure 2-2-12 shows the result when the probability of frequency of occurrences of all the symbols in the symbol string R has been calculated.

Figure 2-2-12
Frequency of occurrences of all the symbols in the symbol string R

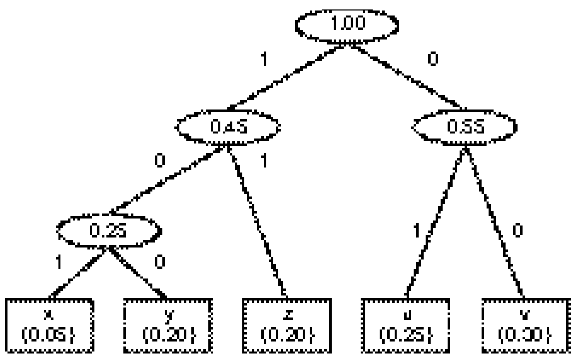
Character	No. of times appearing	Frequency of occurrence
x	1	0.05
y	4	0.20
z	4	0.20
u	5	0.25
v	6	0.30

The Huffman coding works in the way that symbols that do not appear often (have low frequency of occurrence) are allocated a code with long bit length and those appearing frequently (having high frequency of occurrence) are given a code with short bit length.
The procedures in Huffman coding are:

1. Arrangement of each symbol in descending order according to frequency of occurrence. It plays no role which symbol is placed first in case of symbols having identical frequency of occurrence.
2. The symbol with the smallest frequency of occurrence and the symbol with the next-smallest frequency of occurrence become leaf nodes, and a new node is established. This node is given the total frequency of occurrence of the two symbols combined. The branch from this node in the direction of the symbol with lowest frequency of occurrence is labeled 1, and the other branch is labeled 0.
3. Regarding the node created in Step 2 as a new code, Step 2 is repeated until no further new nodes can be created.
4. The sequence of labels granted the branches leading to each symbol from the root node becomes the Huffman code of that symbol.

Figure 2-2-13 shows the Huffman code of the symbol string R and reveals that 45 bits can represent the data of 20 characters.
Huffman coding is still used for compression of this kind of character data. At the present, Huffman coding is also used in JPEG, MPEG and other compression methods (explained later).

Figure 2-2-13 Huffman coding representation method



Character	Bit string
x	101
y	100
z	11
u	01
v	00

Symbol string R

v	u	x	z	v	v	y	y	z	u
00	01	101	11	00	00	100	100	11	01
v	y	z	v	z	u	y	v	u	u
00	100	11	00	11	01	100	00	01	01

(2) JPEG (Joint Photographic coding Expert Group)

JPEG is a worldwide standard for compression and decompression of still images using color/gray scale digitalization, normally relying on an irreversible compression method (DCT: Discrete Cosine Transform) (a reversible method also exists).
This method offers a very high compression ratio (from 1/8 to about 1/100), making JPEG the most commonly used method for distributing full-color still images on the Internet.
JPEG comprises two types of data compression.

- Reversible compression: After decoding of the encoded data, these are completely restored to their original form.
- Irreversible compression: After decoding of the encoded data, these are not restored completely to their original form, but visual observation will show almost no difference.

In addition to JPEG there is another method for compression of still images called LZW, which is used for GIF (Graphics Interchange Format) images. However, the JPEG method is technically superior.

(3) MPEG (Motion Pictures Coding Expert Group)

MPEG it is a set of standards for audio and video compression and decompression and is named after the standardization committee jointly established by ISO (International Organization for Standardization) and IEC (International Electrotechnical Commission).
MPEG enables high compression with very high quality, but since it takes time for restoring the compression, the playback component is normally in the form of a piece of hardware.
Standardization of MPEG encoding is progressing with the division into the four types called MPEG 1, MPEG 2, MPEG 4 and MPEG 7.

① MPEG 1

MPEG 1 was standardized by ITU-T in 1992. Using this standard makes it possible to compress images with a quality like video to 1.5 Mbps.

② MPEG 2

MPEG 2 was standardized by ITU-T in 1994. Using this standard makes it possible to compress television images to about 3 to 6 Mbps, and detailed images, like high-definition television images, to about 10 to 20 Mbps.

③ MPEG 4

With transfer rates ranging from a few kbps to dozens of kbps, MPEG 4 envisioned to be used for mobile communications.

④ MPEG 7

MPEG 7 is under development and is envisioned for use as a high-speed search engine for multimedia information.

(4) Facsimile coding

Facsimile refers to equipment and techniques for transmitting data in the form of documents, drawings, etc. The international facsimile standard for use with analog lines is G3, and G4 is the standard for use on high-speed lines like ISDN lines.
In facsimile, data such as documents or drawings, etc. are captured as an image by scanning, etc., and then encoded by the CODEC method. At this point, the data amount will be very large if the image is encoded as it is. Compression is therefore commonly employed.
MH, MR, MMR, run-length, etc. are some of the techniques used in facsimile encoding.

① MH (Modified Huffman)

MH is facsimile compression encoding method standardized by ITU-T. This compression method builds on the thoughts behind the Huffman coding, and relies on a succession of white and black signals. Each scanned line is processed separately, making it a "one-dimensional encoding method."

② MR (Modified READ)

MR is standardized by ITU-T, and is one of the facsimile encoding methods that yield a higher compression ratio than that obtained with MH. This is a two-dimensional encoding method that also relies on the correlation between scanned lines in the vertical direction, making it more efficient than the one-dimensional encoding method.

③ MMR (Modified Modified READ)

MMR is a compression encoding method that includes partial modification in order to make it more efficient than the MR method.

④ Run-length

The run-length encoding method represents data in which the same elements are occurring consecutively by the elements and the number of times the elements are repeated. Using this method, data like "xxxxxyyyyxxxxxxx," for example, is represented as "05x04y07x."

2.3 Transmission Methods and Communication Lines

A physical network is required in order to transmit data. The following explains the types and characteristics of the networks actually in use.

2.3.1 Classes of Transmission Channel

Channels making up networks can be classified as follows.

- Physical category
- Category classified by communication mode
- Category classified by transmission method

(1) Physical channels

① Two-wire channel

The minimum requirement for one communication line is that it must have one channel for sending the electric signals and one channel for the returning electric signals. A communication line made up of these two channels is called a "two-wire channel."

② Four-wire channel

A communication line made up of four channels (two channels each made up of two lines) is called a "four-wire channel."

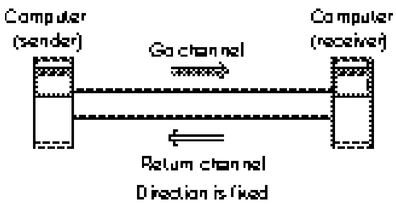
(2) Communication mode

Depending on the data flow direction, communication modes are divided into the following three types.

① One-way mode

In the one-way mode, data only flows in a single direction. Imagine television and radio broadcasting, they are one-way transmission. One-way communication uses a two-wire channel.

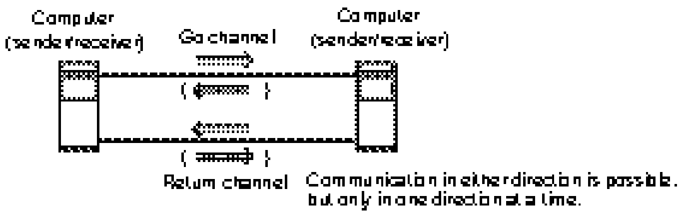
Figure 2-3-1
One-way mode
(two-wire channel)



② Half-duplex mode

Half-duplex allows two-way communication, but only in one direction at a time (Figure 2-3-2). This technique does not allow signals to pass in both direction concurrently, and is used in interactive systems, etc. Half-duplex communication also uses a two-wire channel.

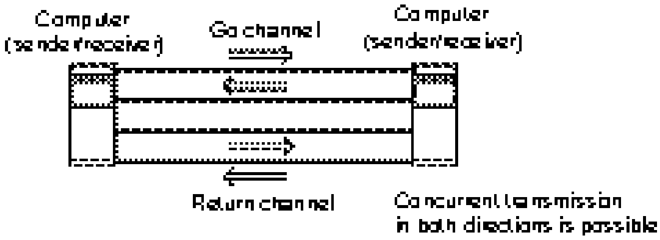
Figure 2-3-2
Half-duplex mode
(two-wire channel)



③ Full-duplex mode

This mode allows concurrent transmission in both directions and can be used with both two-wire channel and four-wire channel.

Figure 2-3-3 Full-duplex mode (four-wire channel)



(3) Transmission methods

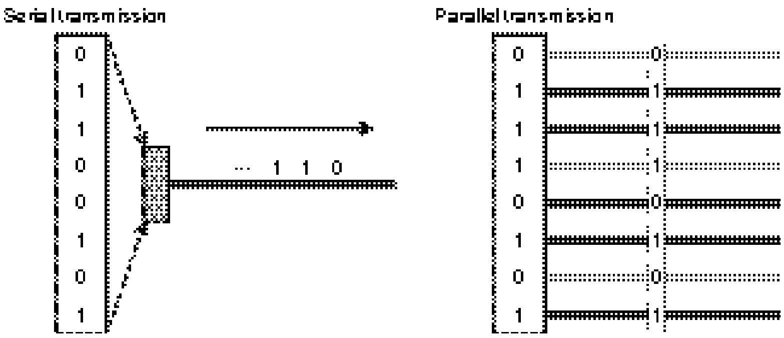
① Serial transmission

Serial transmission is transmission in which data is transmitted one bit at a time. The transmission technique is extremely simple, and low cost, but the transmission speed is slow.

② Parallel transmission

In parallel transmission, several bits are transmitted concurrently. This method is expensive but the transmission speed is high and the technique is used when large amounts of data are sent as a batch.

Figure 2-3-4 Serial transmission and parallel transmission



2.3.2 Types of Communication Lines

The following types of lines are used for transmission of data:

- Leased lines
- Switched telephone network

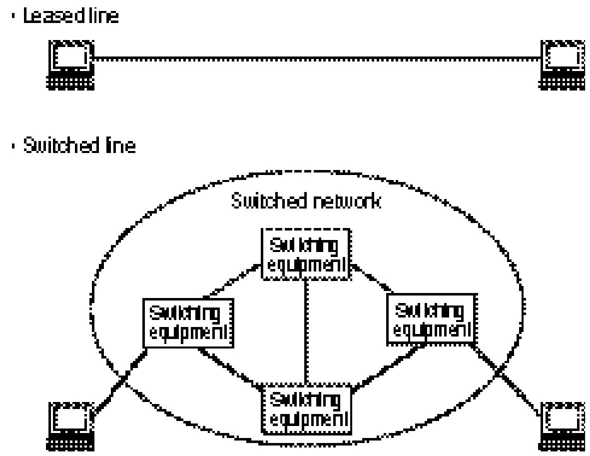
(1) Leased lines

Leased lines are dedicated lines wired directly between the communicating parties, and a flat fee is charged for this arrangement. You hold the right to use the leased line and this arrangement is suitable when large amounts of data have to be transmitted.

(2) Switched network

In switched networks, the communicating parties are not specified. When switched telephone networks are used, the other party must first be dialed to secure transmission channel. Representative examples of switched networks are public telephone networks and ISDN (explained later).

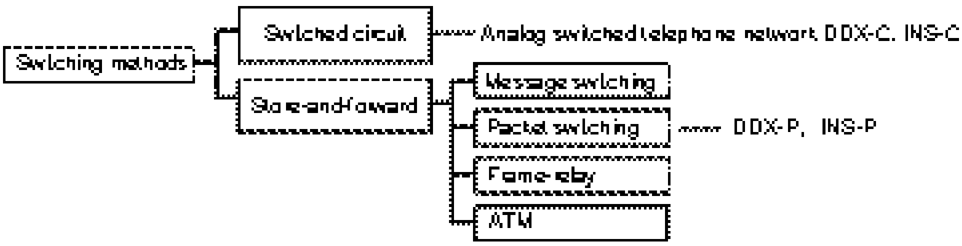
Figure 2-3-5 Leased lines and switched networks



2.3.3 Switching Methods

There are two switching methods available for use with switched networks: switched circuit and store-and-forward.

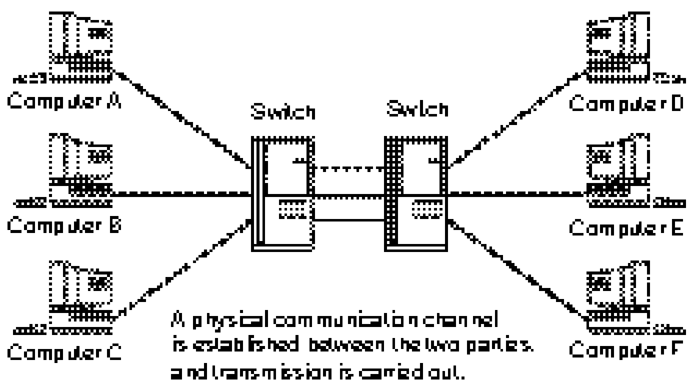
Figure 2-3-6 Switching methods



(1) Switched circuit

A switched circuit has the same structure as a public telephone networks. Each time a request for data transmission is issued, a physical communication channel is established and data transmission is carried out. Because the sender and recipient are physically connected, this method is applicable to relatively large data transmission, but it is restricted by the factor that the transmission rate must be the same in both directions. Analog switched telephone networks employ the circuit switching method.

Figure 2-3-7 Switching circuit



There are two switched circuits for digital data exchange.

① DDX-C (Digital Data eXchange-C)

DDX-C is a circuit switching service for digital transmission at 200 - 48,000 bps. Currently, the trend is towards use of INS-C and public telephone networks, and new initiatives using this method are not under consideration. (For details, see Section 3.6.2, Telecommunications Services and WAN.)

② INS-C (Information Network System-C)

INS-C is a circuit switching service using ISDN, and is offered for use on both of the basic interface (INS net 64; 2 B + D), and the primary rate interface (INS net 1500; 23B + D or 24B). (For details, see Section 3.6.3, Telecommunications Services and ISDN.)

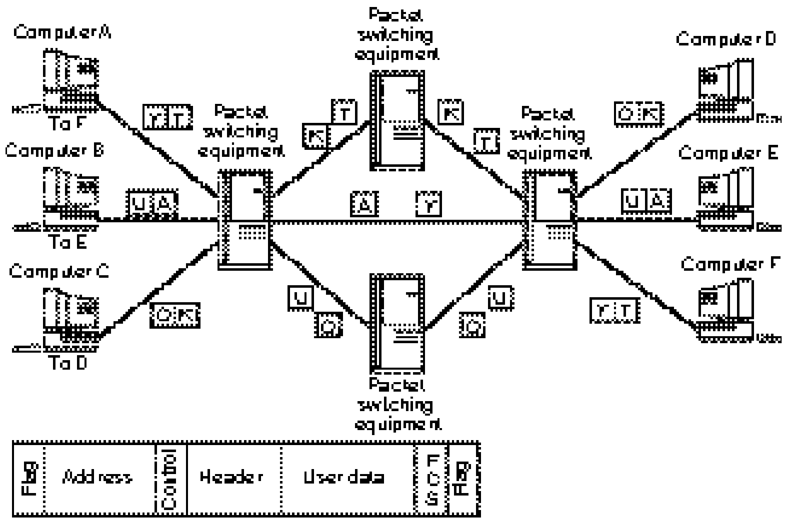
(2) Store-and-forward

Store-and-forward is a message-passing technique in which data is exchanged by means of addresses appended to the data units (packets) without the establishment of a physical communication channel to the recipient as in the case of circuit switching. X.25 is commonly used terminal interface for this technique.

① Packet switching

Figure 2-3-8 illustrates packet switching and its formats.

Figure 2-3-8 Packet switching and formats



<Characteristics>

- Data is divided into units, called packets, having a uniform length. An address and information (header) indicating the serial number of the packet, etc. is appended to the packet.
- The packets are stored in the switching equipment, which then sequentially forwards the packets taking traffic condition of the line into consideration. It is of no importance even if the transmission speeds of the recipient and the receiver are different. However, differences in transmission speeds can lead to "transmission delays."
- The PAD (Packet Assembly and Disassembly) interface is necessary to disassemble data into packets and later assemble the data again. This function is already installed if the terminal type is PT (Packet mode Terminal), but if the terminal type is NPT (Non-Packet mode Terminal), the function has to be performed by the switching equipment.
- Highly reliable communication is possible, because transmission confirmation and error control are performed at packet unit level, but transmission speed suffers from these characteristics.
- Circuit switching systems only require the same number of lines as the number of terminals. In packet switching, a packet is sent to the recipient via multiple circuits, so it is sufficient with only one trunk line between switching equipment. In packet switching, multiple logical lines are established on the same physical circuit enabling simultaneous communications with multiple terminals. This is called

"packet multiplexing."

The following two examples are typical packet switching services.

a. DDX-P, DDX-TP

Packet switching services employing digital data exchange (DDX) comprise DDX-P (Type 1 packet switching service) and DDX-TP (Type 2 packet switching service; DDX-P service using public telephone networks).

b. INS-P

INS-P is a packet switching services using ISDN, and it is available with both the basic interface (INS net 64; 2 B + D), and primary rate interface (INS net 1500; 23B + D or 24B). INS-P also allows packet transmission using the D channel. (For details, see Section 3.6.3 Telecommunications Services and ISDN.)

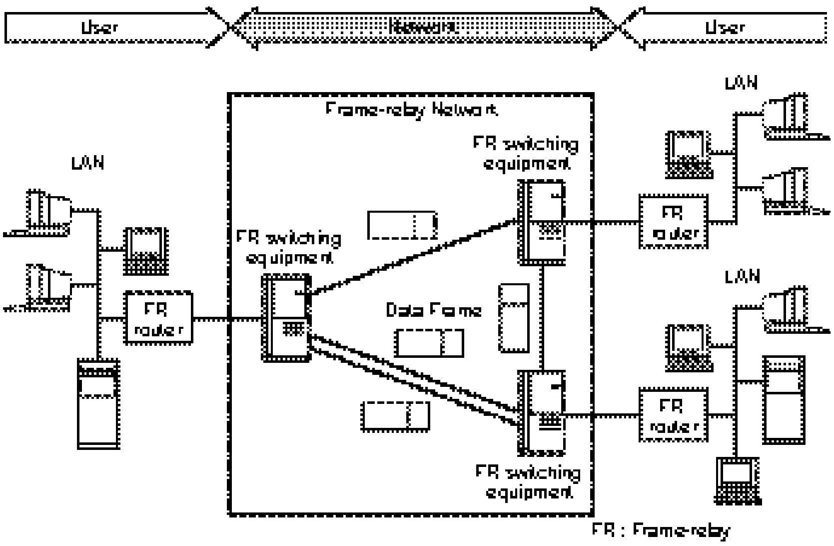
② Message switching

Message switching system is a technique in which all the data, such as files and images, etc., are transmitted as one message unit. The differences in data length cause problems in term of efficiency and transmission time, and it is rarely used these days.

③ Frame-relay

Briefly said, the frame-relay is a "high-speed version of packet switching." This transmission technology enables high-speed transmission and is used in WAN (Wide Area Network).
The frame-relay has inherited the X.25 packet switching protocol, and realized throughput enhancement up to about 1.5 Mbps by the employment of new techniques.
Figure 2-3-9 shows the network structure of the frame-relay system.

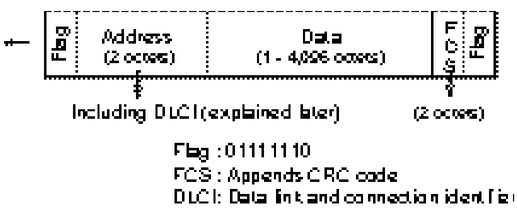
Figure 2-3-9 Frame-relay network



Basically, the frame-relay system transmits data by relay via FR (frame-relay) switch in the same manner as the packet switching system.
<Characteristics>

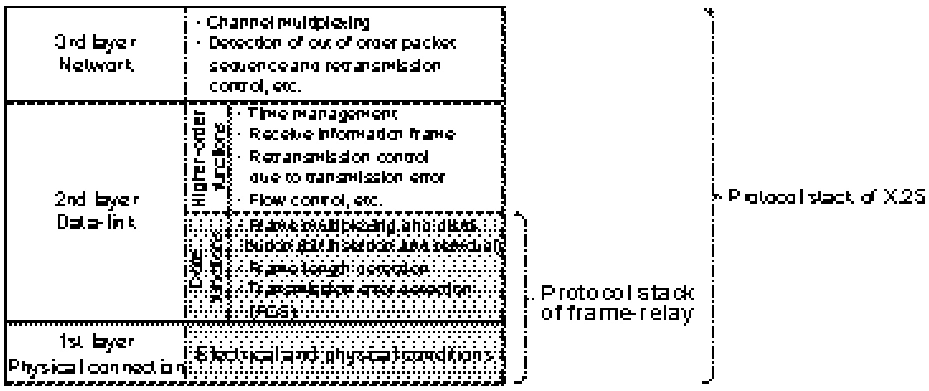
- Employs variable length frames
  Variable length frames are used for the message format that consists of flag, address field, data field, and FCS.

Figure 2-3-10 Message format



- High-speed transmission is possible at about 1.5 to 2 Mbps.
- Simplification of the X.25 protocol
  This protocol simplifies the ITU-T X.25 recommendation (omission of the resending control by means of packet units), and comprises only the basic controls, such as transmission error detection by FCS. This simplification makes high-speed transmission possible.

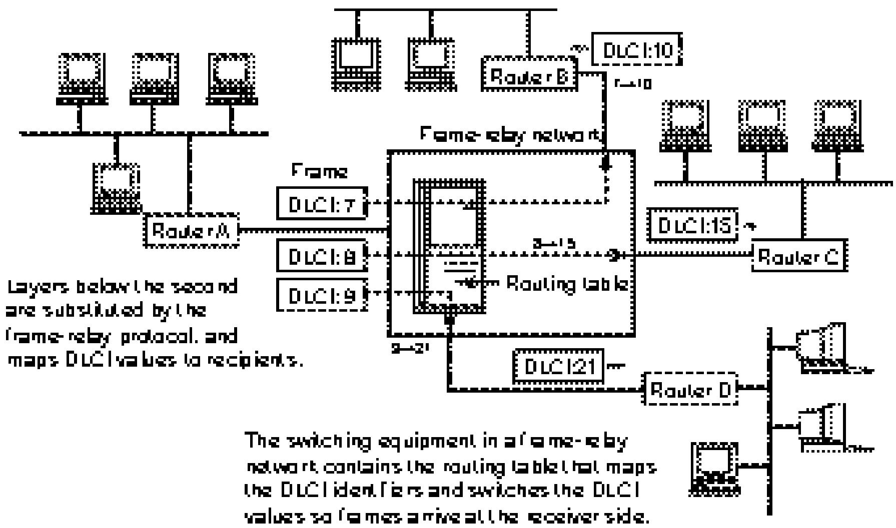
Figure 2-3-11 Frame-relay protocol



In frame-relay, only the core part of the second layer (data-link) of the OSI hierarchical structuring is defined. As frame-relay relies on the higher levels of protocols existed in other network systems, it is highly compatible with existing products.
Packet switching is based on the X.25 protocol, and the word "switching" is applied strictly to control each packet transmission. The word "relay" is used in connection with frame-relay, because this technique sends packets using the "bucket-relay" from the sender to the receiver via frame-relay switching equipment without confirming the transmission.

- Frame multiplexing
  Even though frame-multiplexing has the same characteristics as packet multiplexing, the frame's address field contains the DLCI (Data Link Connection Identifier). The destination can be identified by this DLCI.
  Consequently, simultaneous transmission of frames to multiple destinations physically using the same circuit is enabled by consecutively sending frames with different DLCI identifiers.

Figure 2-3-12 Frame multiplexing



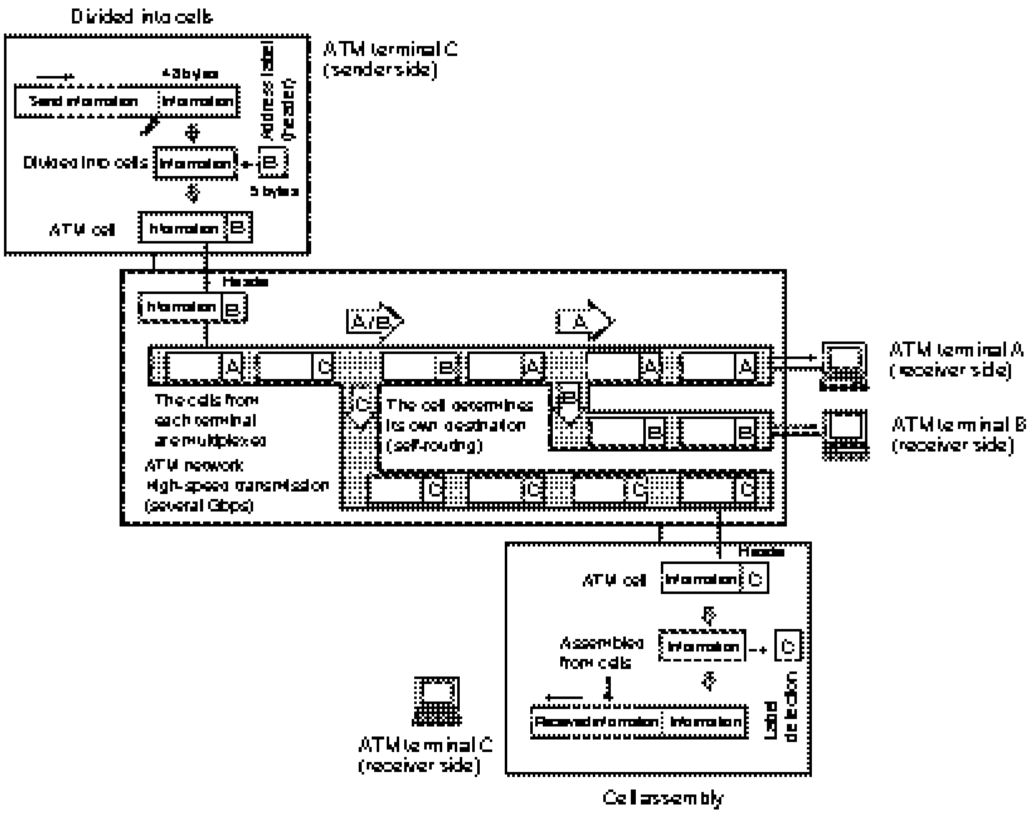
- CIR (Committed Information Rate)
  CIR denotes the information transmission rate guaranteed by the frame-relay network and is a newly established standard for frame-relay. The guaranteed rate differs in the speed under normal circumstances or congestive conditions (when the traffic on the network is excessive).

During congestions, the data load is controlled on the terminal side by using the guaranteed CIR value as the criterion.

④ ATM (Asynchronous Transfer Mode)

The ATM offers a much higher transmission rate (several megabits to several gigabits) than that of the frame-relay, and it is probably the technique that communications will come to rely on in the multimedia era. Research in order to commercialize this technique is under way in many countries.
B-ISDN (Broadband-ISDN) is closely related to ATM and enables data transmission at superfast speeds (156 Mbps and 622 Mbps, etc.). In the multimedia era, B-ISDN is likely to become an extremely effective means of communication for transmission of video that requires high quality images.
The two new communication methods STM (Synchronous Transfer Mode) and ATM are used with B-ISDN, but basically the efforts aim at integrated networks employing ATM.
A LAN technique incorporating ATM technologies and called "ATM-LAN" is also receiving much attention.

Figure 2-3-13 ATM image illustration

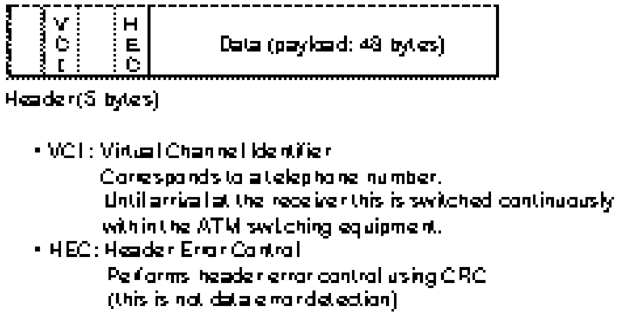


<Characteristics>

- Cell unit transmission

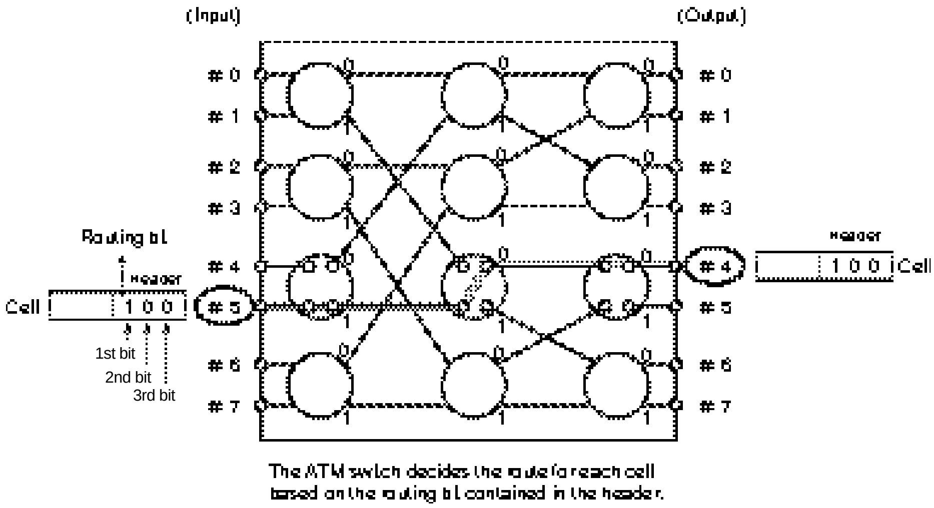
  ATM transmits data in cell units. This method is called cell-relay. ATM is one type of several cell-relay techniques.
  A cell consists of small units of data, image or other information, each unit having the size of 48 bytes (octet). A header (5 bytes) indicating destination address, etc. is appended as the head of the cell.
  The header includes a 1-byte header error detection code (CRC code).

Figure 2-3-14
Cell



- Hardware switching
  ATM uses ATM switching hardware, which enables continuous transmission at extremely fast transmission rate.

Figure 2-3-15 Switch principles



ATM sends data in cell units but since the communication line is decided instantaneously by means of a hardware switch, the ATM is placed midway between "packet switching" and "circuit switching."

- ATM protocol
  As mentioned earlier, the frame-relay enables higher transfer rate by simplification of the X.25 protocol, and the ATM simplifies even further than the frame-relay in order to realize high-speed transmission.

Figure 2-3-16
ATM protocol

	Upper layer	
Layer 2	AAL (ATM adaption layer)	• Cell disassembly and assembly, etc.
Layer 1 (Physical layer)	ATM layer	• Cell and header generation, extraction • Cell multiplexing/separation, etc.
	Physical layer	• Cell synchronization • HEC generation/verification • Cell speed adjustment • Physical media dependence

• It is apparent that functionalities are concentrated in layer 1 to an even higher degree than in the frame-relay.

- Congestion control
  In advance, cells are arranged in priority order (included in the header) in accordance with their respective importance, and when congestion occurs, cells with high priority are not affected. Additionally, the technique is perfected by establishing congestion bypasses to maintain the best possible high-speed transmission.
- Allows transmission of all kinds of data
  ATM is independent of data types and forms and allows transmission of any kind of data.
- Applicable fields
  Due to its superfast characteristics and flexibility, ATM is expected not only to find employment in a variety of fields such as LAN and WAN but also in broadcasting and VOD (Video On Demand).

Exercises

Q1 **In order to transmit digital data using analog communication lines, the operation called "modulation" is required. Which of the following modulation techniques is the simplest to implement though it is susceptible to noise and fluctuations in signal levels?**

A. Phase modulation B. Frequency modulation C. Amplitude modulation
D. Quadrature amplitude modulation E. Code multiplex modulation

Q2 **Which modulation technique is used for transmitting audio via digital networks?**

A. Phase modulation B. Frequency modulation
C. Amplitude modulation D. Pulse code modulation

Q3 **Which is the correct description of the parity check used to counter transmission errors in communication lines?**

A. 1-bit errors can be detected.
B. 1-bit errors can be compensated and 2-bit errors can be detected.
C. In the case of even parity 1-bit errors can be detected, and 1-bit errors cannot be detected in case of odd parity.
D. In the case of odd parity, odd figure bit errors can be detected, and even figure bit errors can be detected in case of even parity.

Q4 **A parity bit should be appended to a 7-bit character code so that the number of "1"s contained in the 8 bits, including the parity bit, becomes an even figure. The parity bit is placed at the higher-order position in the 7-bit character code. In this case, which of the following is the hexadecimal notation code representing 4F with the parity bit added to the character code?**

A. 4F B. 9F C. CF D. F4

Q5 **Which is the error detection technique that adds a remainder, found by a certain generating polynomial expression, to the bit string on the sender side, and detects errors by whether or not the remainder is the same on the receiver side by dividing the received string using the same polynomial expression?**

A. CRC B. Longitudinal parity check
C. Lateral parity check D. Hamming code

Q6 **In memory error control technique, which of the following employs 2-bit error detection and 1-bit error correction functions?**

A. Even parity B. Lateral parity
C. Check sum D. Hamming code

Q7 **When using a line whose bit error rate is 1/600,000, and you send data at a transmission rate of 2,400 bits/sec, in how many seconds will one bit error occur on an average?**

A. 250 B. 2,400 C. 20,000 D. 600,000

Q8 Which is the correct description of asynchronous transmission?

A. The receiver side constantly watches for the bit string used for synchronization sent from the sender side, and when this is received, it regards what follows as data from the next bit.
B. The receiver side is able to recognize where characters start by the bits that the sender side has appended at the start and ending of each character.
C. The sender side appends a bit so that "1" bits in each character becomes an even number.
D. The sender side and receiver side retains timing by constantly sending a specific bit pattern on the communication line even when there is no data to be sent.
E. Timing signals for synchronization is always flowing on the communication line, and the terminals send and receive data in sync with these timing signals.

Q9 The character T (JIS 7-unit code string 1010100) is sent using the start-stop synchronized data transmission technique that employs even parity as the character check method. Which is the correctly received bit string? The received bit string is written in order from the left beginning with the start bit (0), lower order bits to higher order bits of the characters, parity bit and stop bit (1).

A. 0001010101 B. 0001010111 C. 1001010110 D. 1001010111

Q10 What is the time required to transmit a data of 120 characters using the start-stop technique with a communication line having a transmission rate of 2,400 bit/sec? The data is an 8-bit code with no parity bit, and both the start signal and the stop signal are 1-bit length.

A. 0.05 B. 0.4 C. 0.5 D. 2 E. 200

Q11 What is the technique that combines multiple slow-speed lines into one high-speed line by time division multiplexing to convert the bit strings to be transmitted on the high-speed line?

A. CDM B. FDM C. TDM D. WDM

Q12 What is the name of the irreversible compression method for still images that has become an international standard?

A. BMP B. JPEG C. MPEG D. PCM

Q13 Which of the following adequately describes the characteristic of packet switching?

A. Delays do not occur inside the switched network.
B. Suitable for transmission of large amounts of consecutive data.
C. Is not suitable for transmission of information between equipment where transmission speeds and protocols differ.
D. Enables efficient use of communication circuits (by sharing multiple communication path).

Q14 Which is the correct description of packet switching?

A. Packet switching service is not possible with ISDN.
B. Compare to circuit switching, the latency within the network is short.
C. In order to carry out communication by packet switching, both the sender and the receiver must be packet mode terminals (PT).
D. By setting multiple logical circuits, concurrent communication with multiple parties can be performed using one physical line.

Q15 What is the adequate description of the characteristic of frame-relay?

A. DLCI (Data Link Connection Identifier) enables frame multiplexing.
B. Based on the premise of the use on a low-quality communication line with errors frequently occurring.
C. As communication method, only the SVC (Switched Virtual Circuit) technique is used.
D. When a frame error is detected, the frame-relay switching equipment resends the particular frame.