

# **Sistem Komunikasi 1**

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## **BAB 9**

# **ADC-MUX**



# ADC (ANALOG TO DIGITAL CONVERTER) / PCM (PULSE CODE MODULATION)

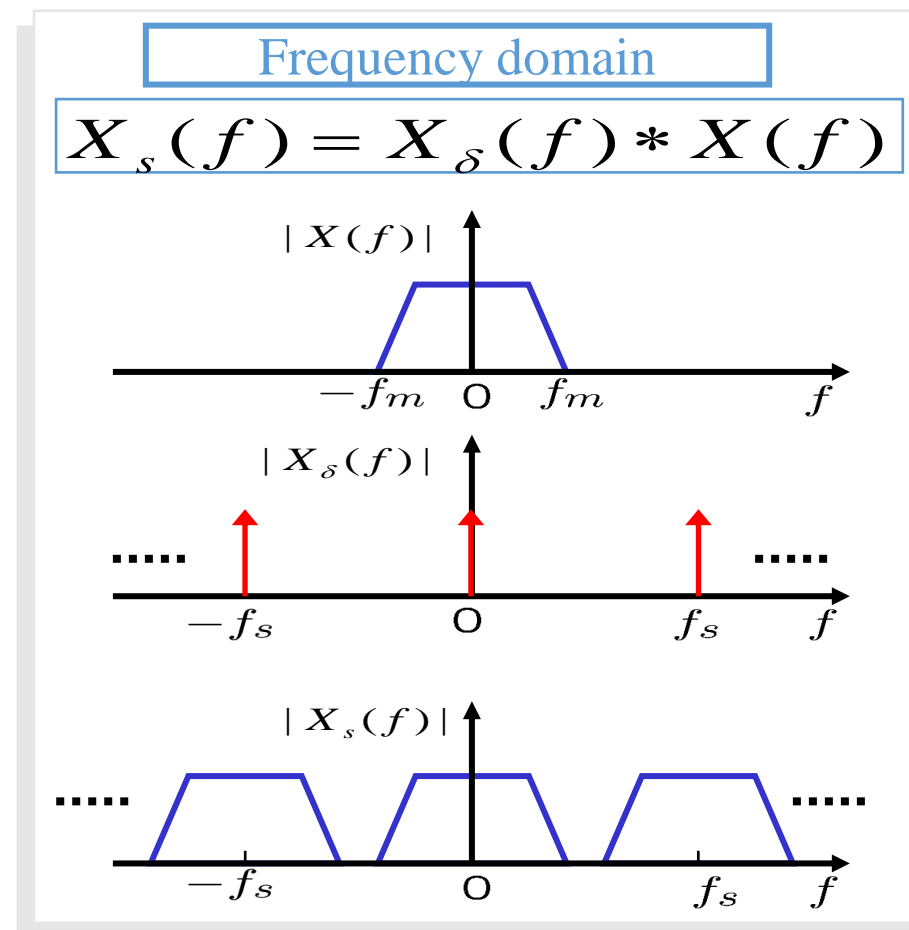
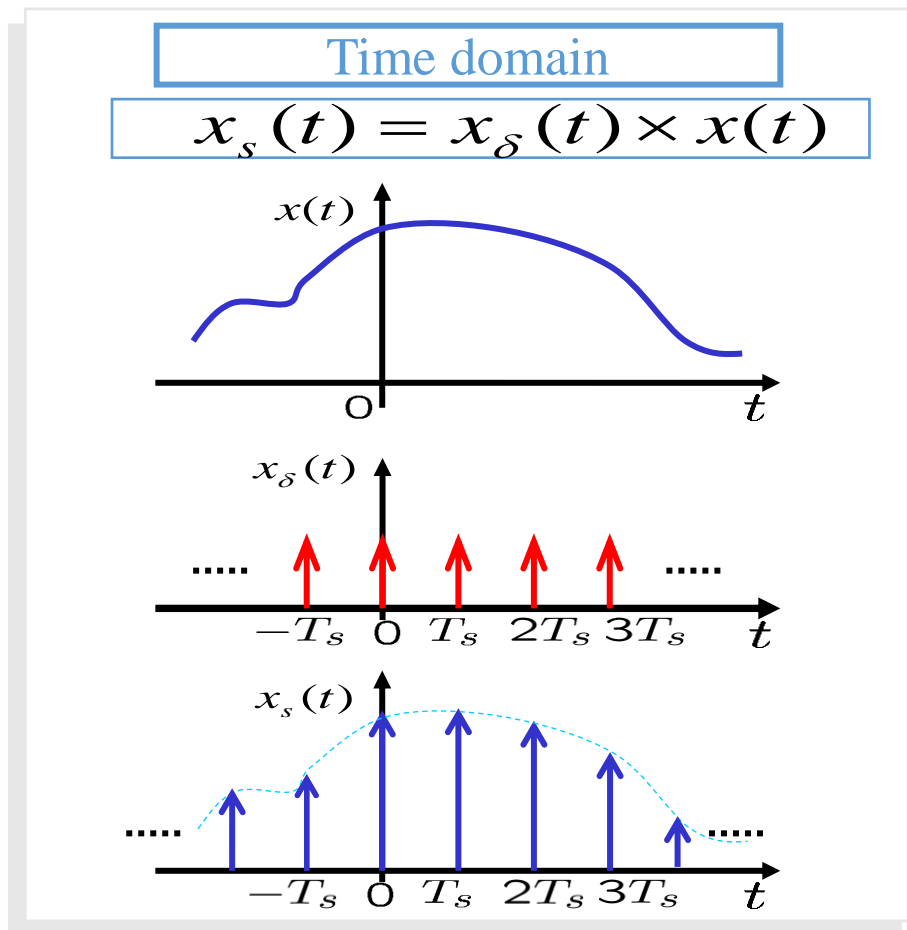
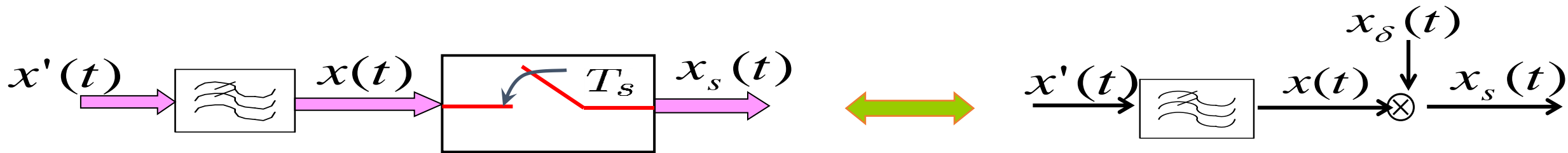
Mengubah sinyal voice analog menjadi sinyal digital

Proses yang terjadi dalam PCM :

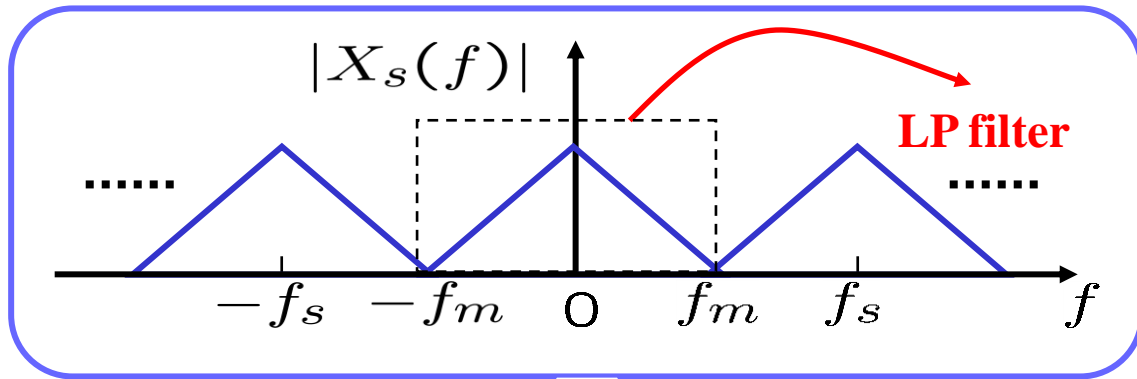
- Sampling (pencuplikan)
- Quantizing (kuantiasasi)
- Encoding (pengkodean)



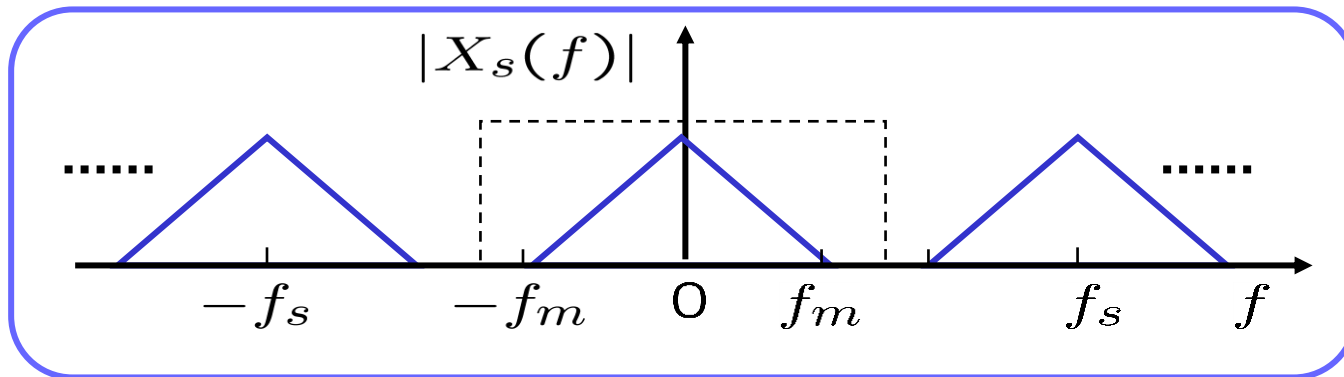
# PROSES PENCUPLIKAN (SAMPLING)



# Aliasing effect

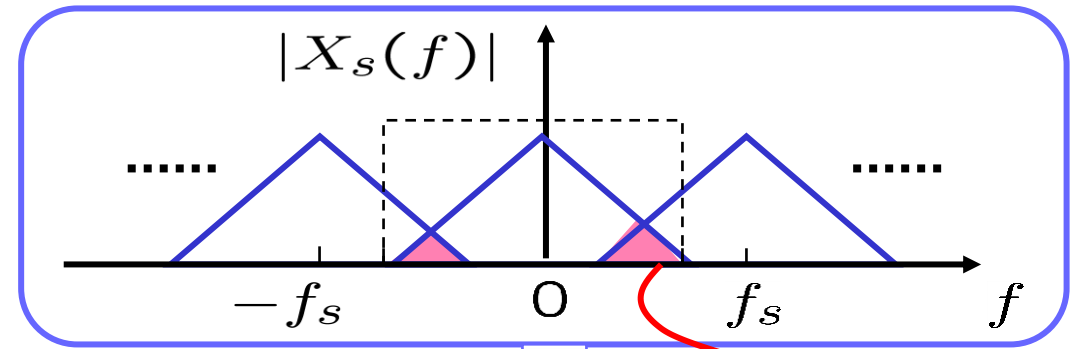


$$f_s = 2f_m$$



$$f_s > 2f_m$$

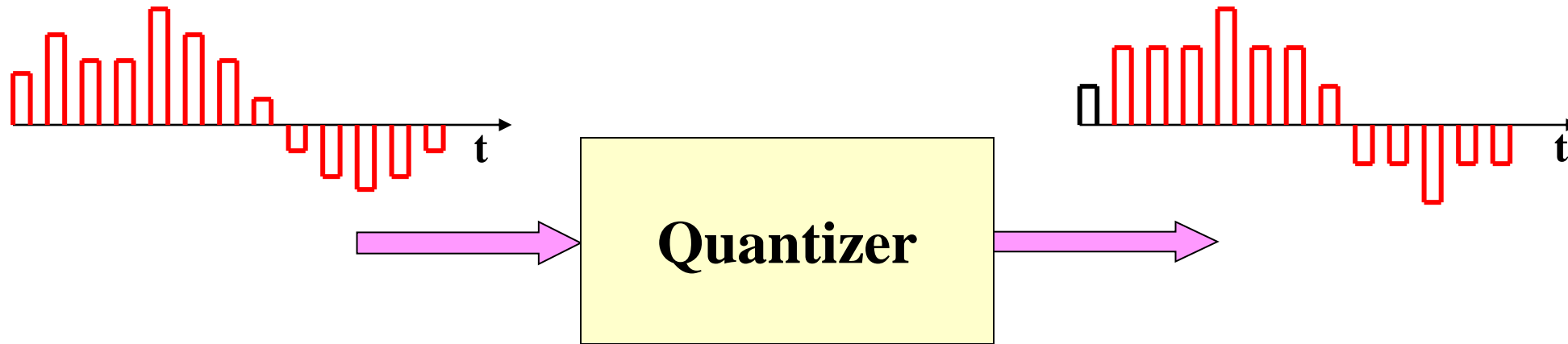
Nyquist criteria



$$f_s < 2f_m$$

aliasing

# PROSES KUANTISASI (QUANTIZATION)



**Kuantisasi** : mengubah level amplituda menjadi diskret dengan jumlah terbatas.

Jumlah level kuantisasi  $M = 2^N$

$N$  = jumlah bit pengkodean

Terdapat 2 jenis kuantiser yaitu :

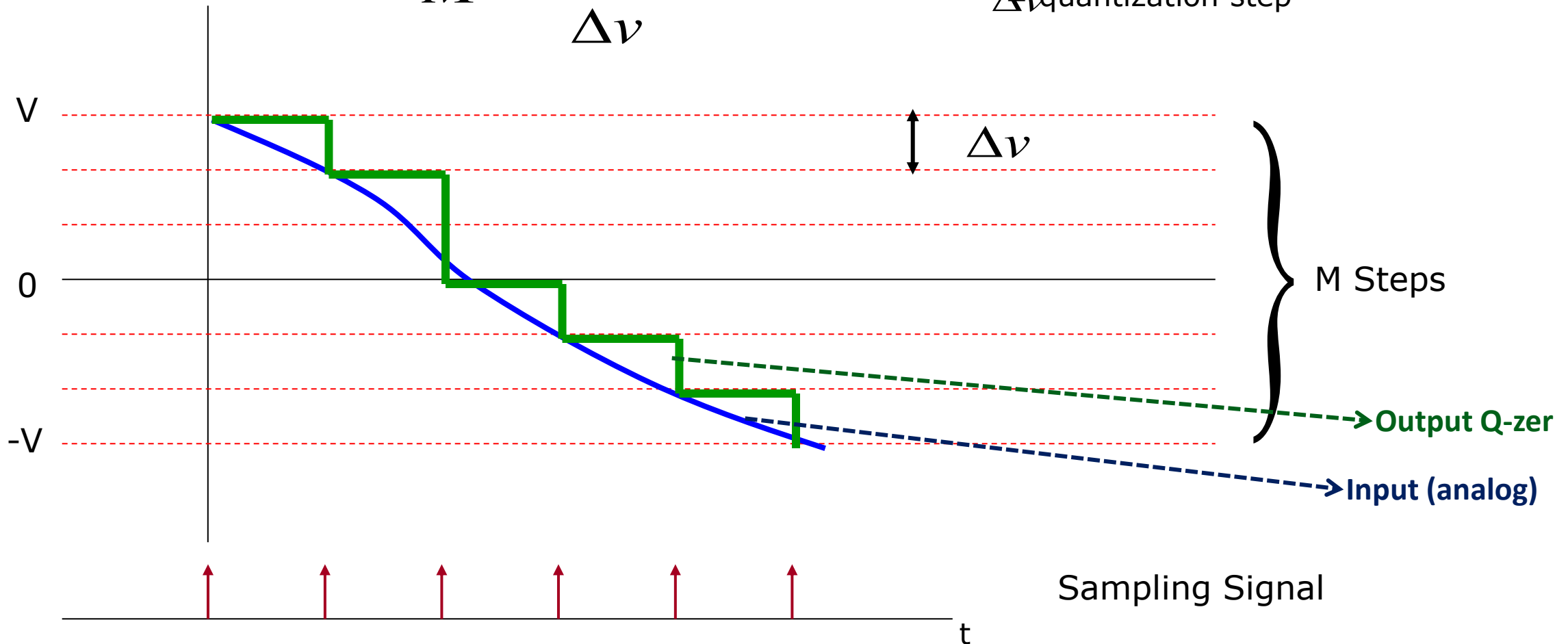
**Kuantiser Uniform** (lebar selang kuantisasi seragam)

**Kuantiser Non-Uniform** (lebar selang kuantisasi tidak seragam)

# Quantization

$$M = \frac{2V}{\Delta v}$$

Where  $M$  = no. of steps  
 $\Delta v$  = quantization step



# QUANTISER UNIFORM

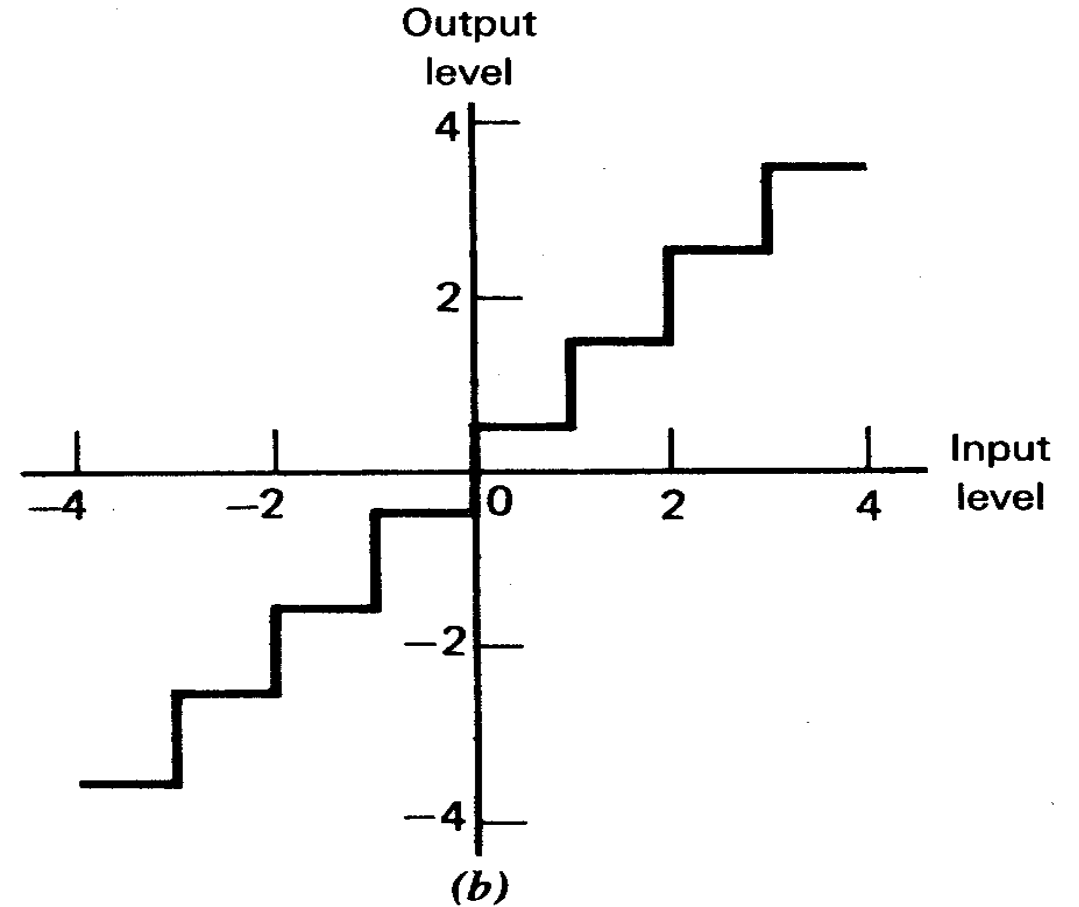
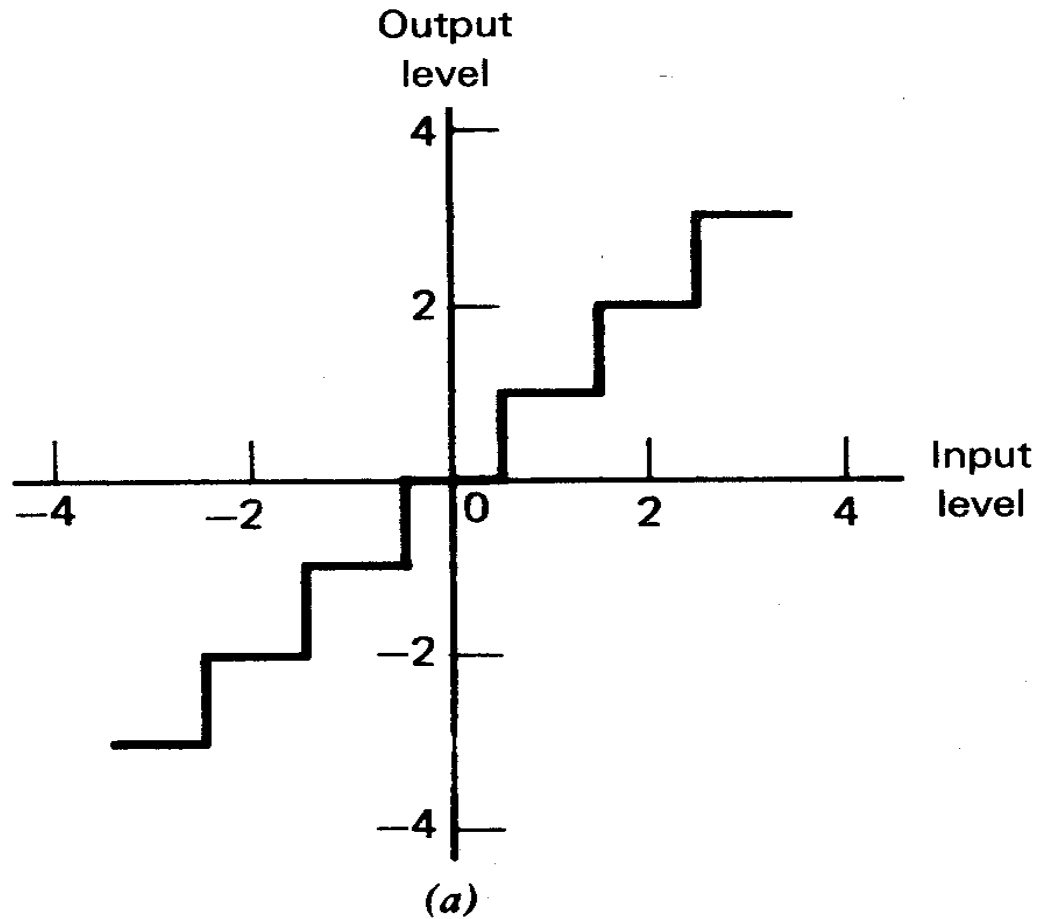
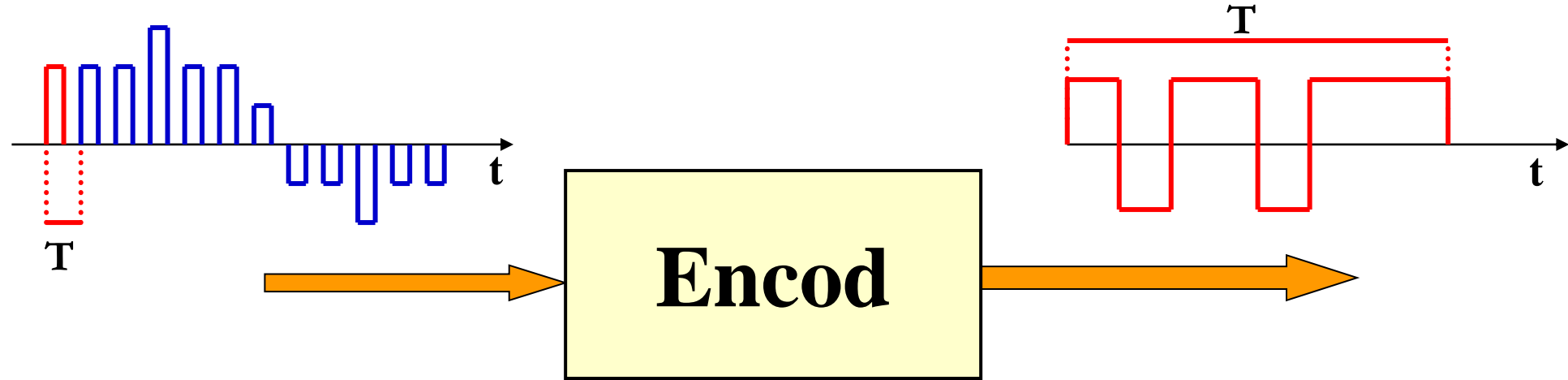


Figure 6.17 Two types of quantization: (a) midtread and (b) midrise.

# PROSES PENGKODEAN (ENCODING)

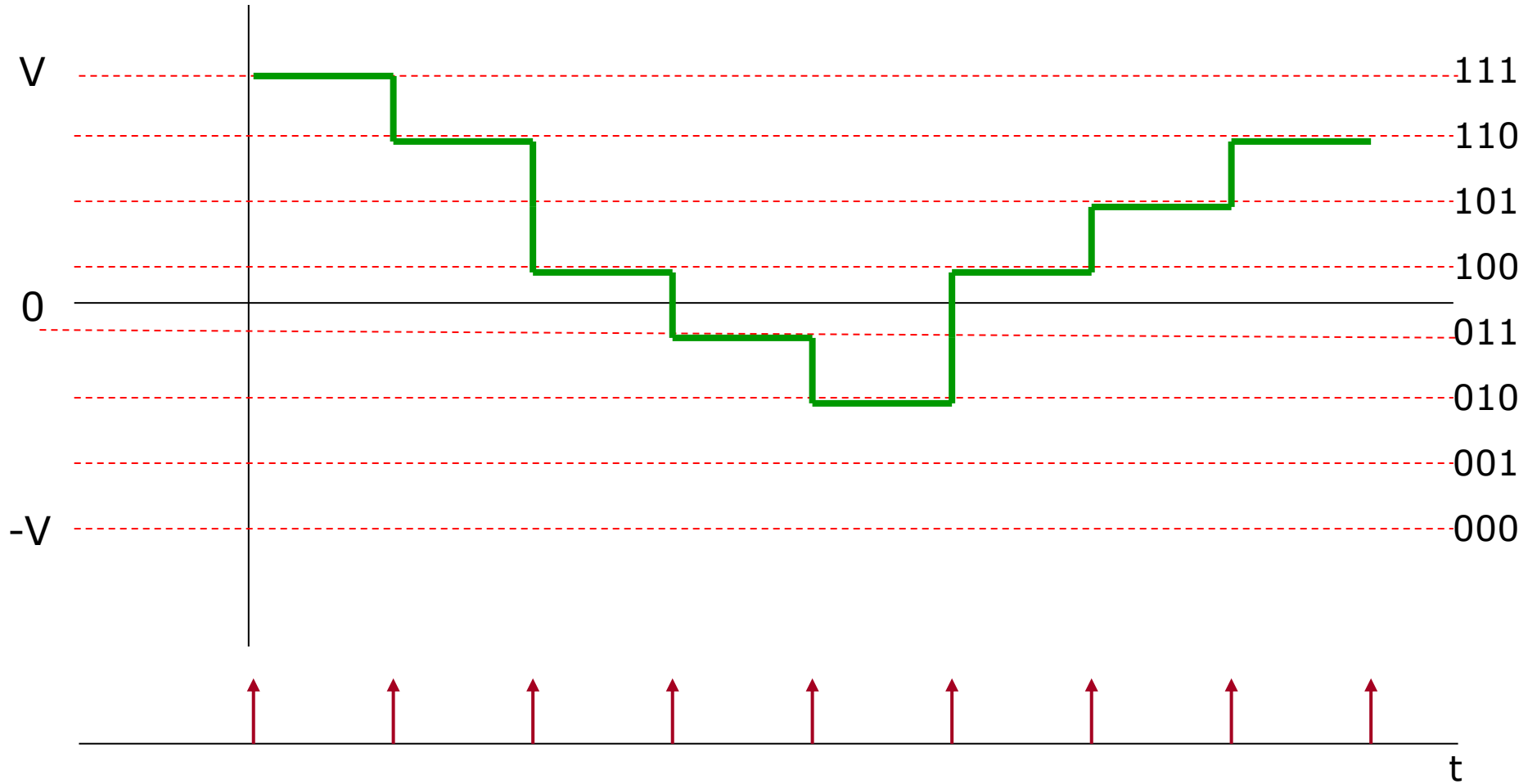


Contoh di atas menunjukkan proses encoding,  
1 simbol masukan dikodekan menjadi 8 bit

Jumlah bit untuk mengkodekan tiap simbol ditentukan oleh  
perangkat ADC (Analog to Digital Converter)



# Encoding



$$M = 2^N$$

1 1 1 1 1 0 1 0 0 0 1 1 0 1 0 1 0 0 1 0 1 1 1

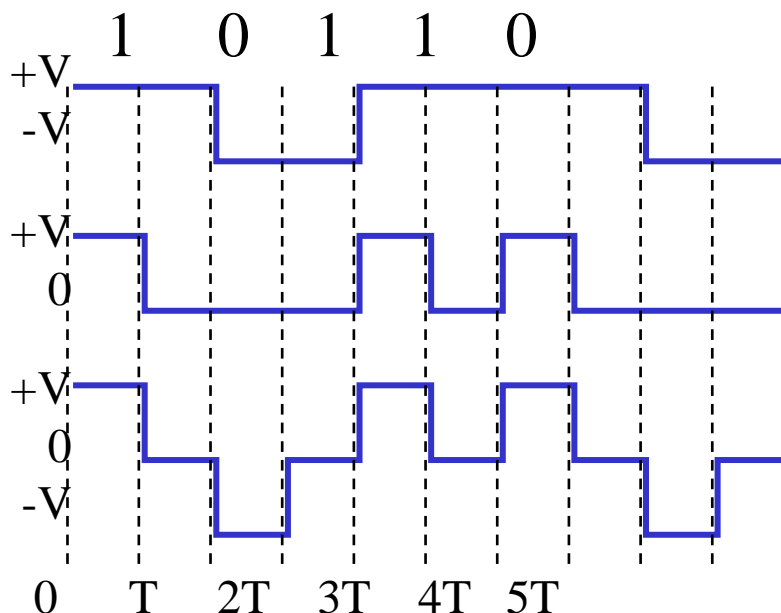


# Bentuk gelombang/sinyal PCM

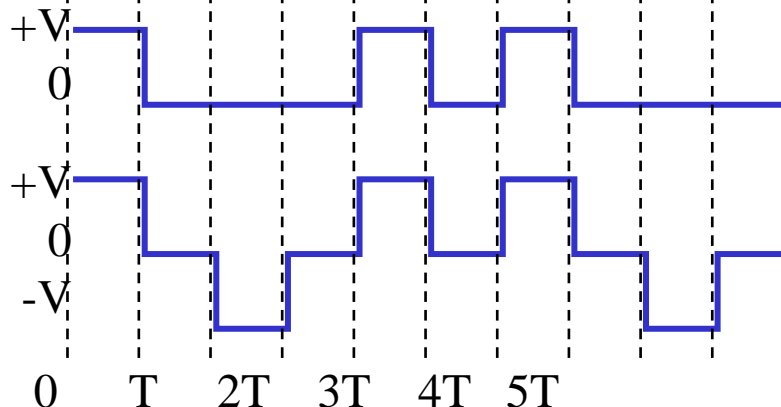
- NonReturn-to-Zero (NRZ)
- Return-to-Zero (RZ)

- Phase encoded
- Multilevel binary

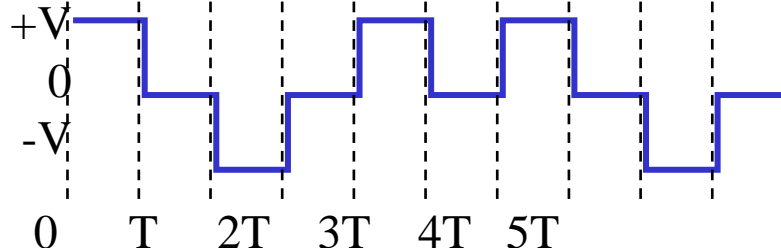
NRZ-L



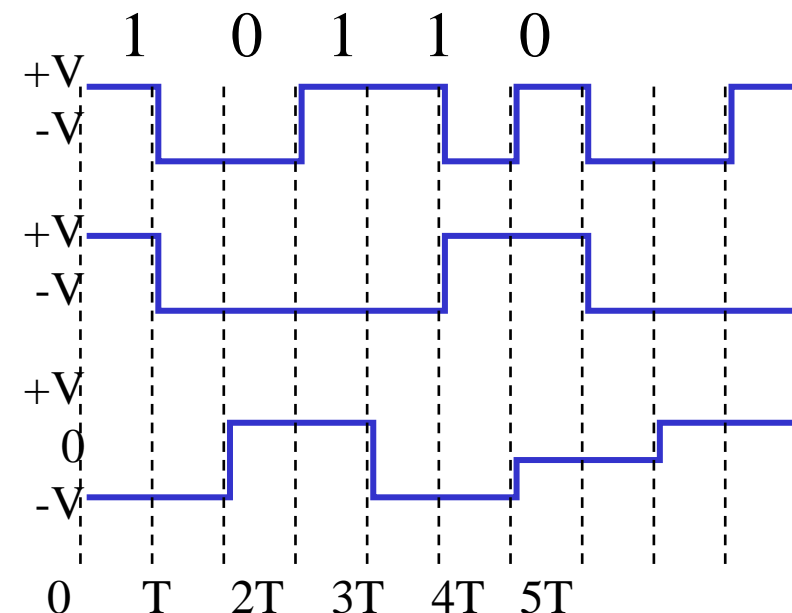
Unipolar-RZ



Bipolar-RZ



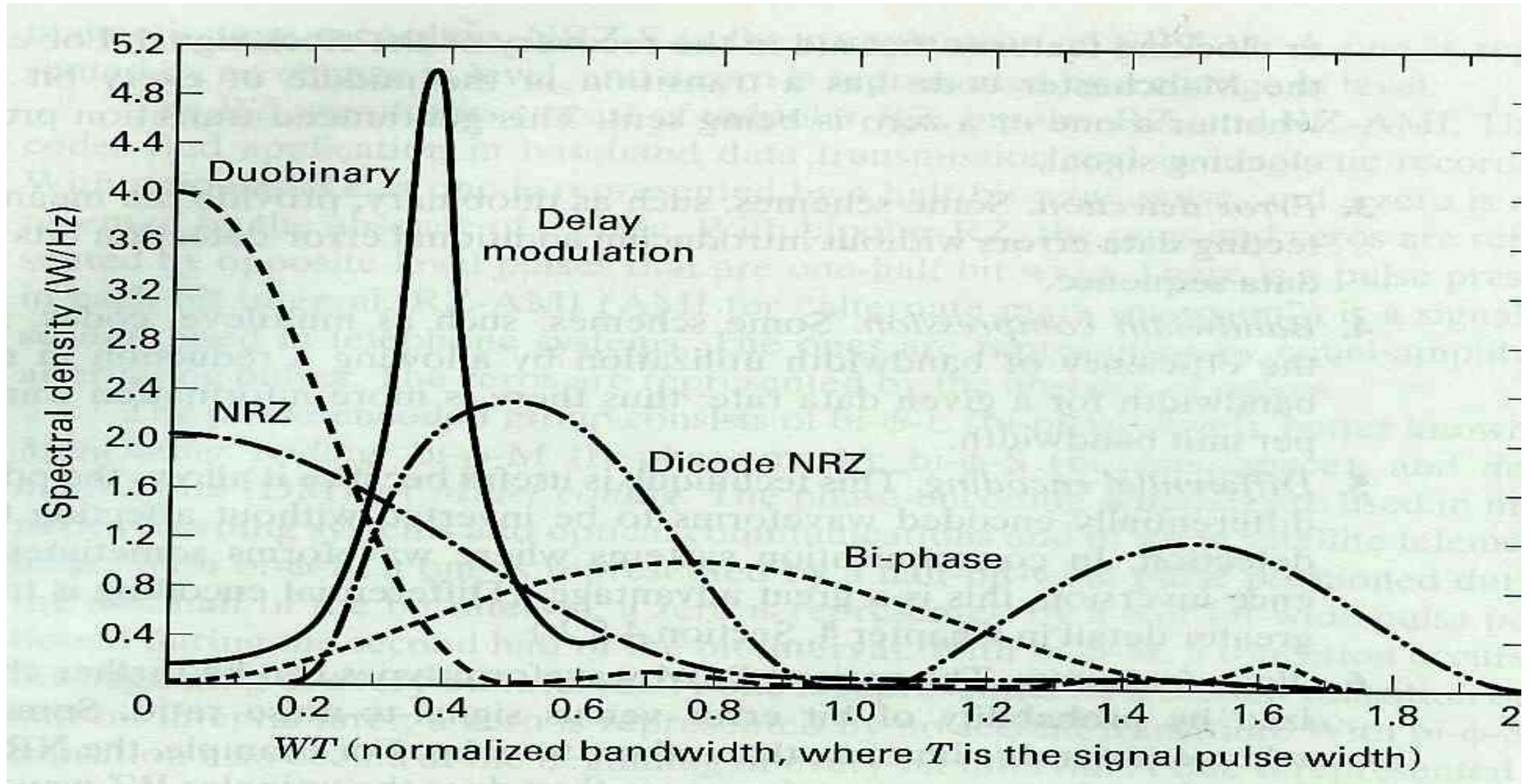
Manchester



Miller

Dicode NRZ

# Spectrum sinyal PCM



# BIT RATE KANAL VOICE

Frekuensi sampling ( $f_s$ )  $> 2 \cdot BW$   
 $> 2 \cdot$  frekuensi informasi maksimum  
(berdasarkan kriteria Nyquist)

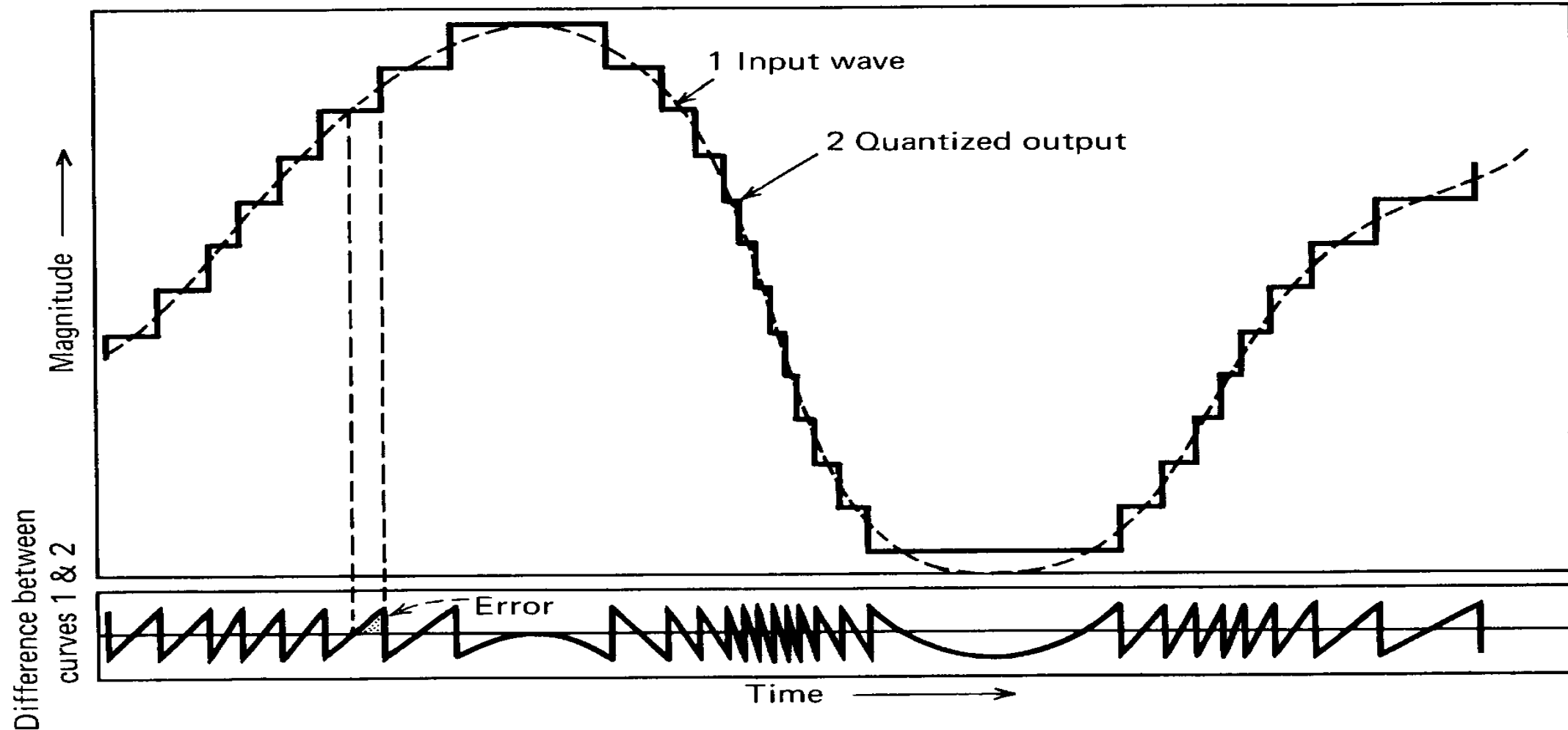
BW kanal suara  $\sim 4$  kHz (300 – 3400 Hz)

Kecepatan sampling untuk tiap kanal suara  $= 2 \times 4000 = 8000$  sample/s  
1 sample dikodekan menjadi 8 bit

Bit rate 1 kanal voice :

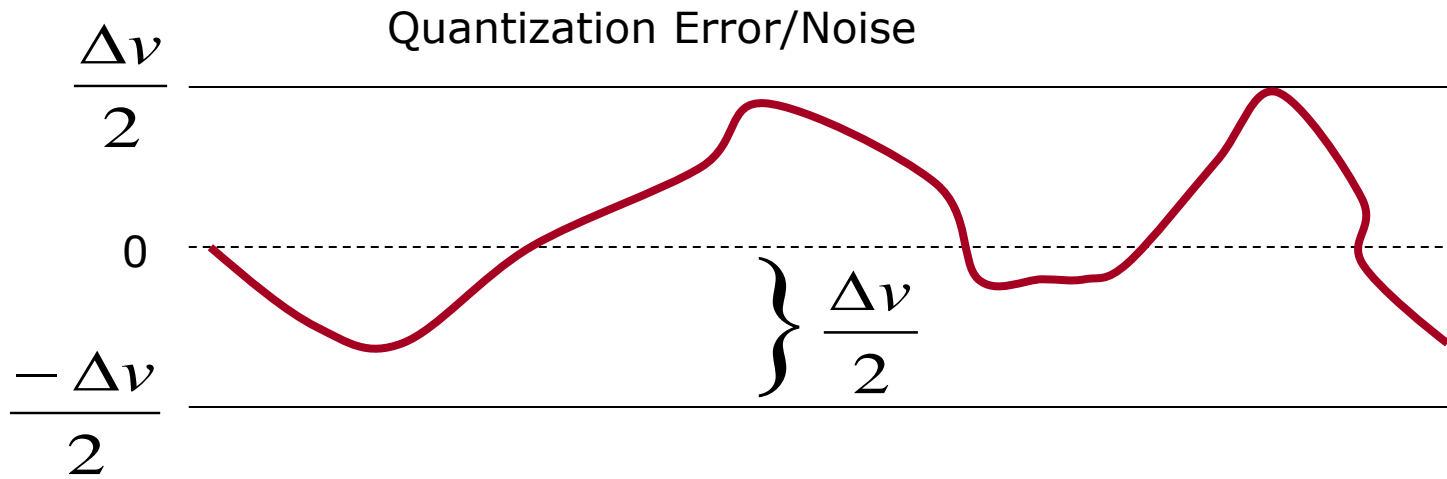
$$\begin{aligned} \text{BR} &= 8000 \text{ sample/detik} \times 8 \text{ bit/sample} \\ &= 64 \text{ kbps} \end{aligned}$$

# Quantization Error

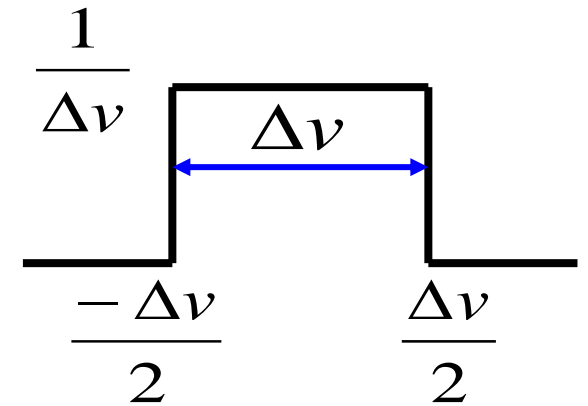


**Figure 6.18** Illustration of the quantization process. (Adapted from Bennett, 1948, with permission of AT&T.)

# Quantization Error



Uniform distribution



$$e(t) = f(t) - f_Q(t)$$

$$-\frac{\Delta v}{2} \leq e(t) \leq \frac{\Delta v}{2}$$

# Signal to Noise Ratio

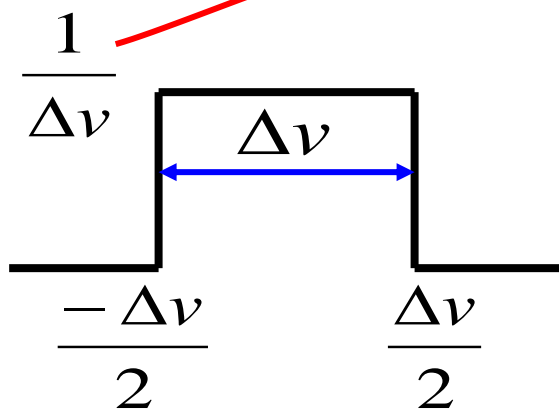
$$SNR|_Q = \frac{\text{Signal Power}}{\text{Error Signal Power}}$$

The average power

$$\bar{P} = \lim_{T \rightarrow \infty} \frac{1}{T} \int_{-\infty}^{\infty} f^2(t) dt$$

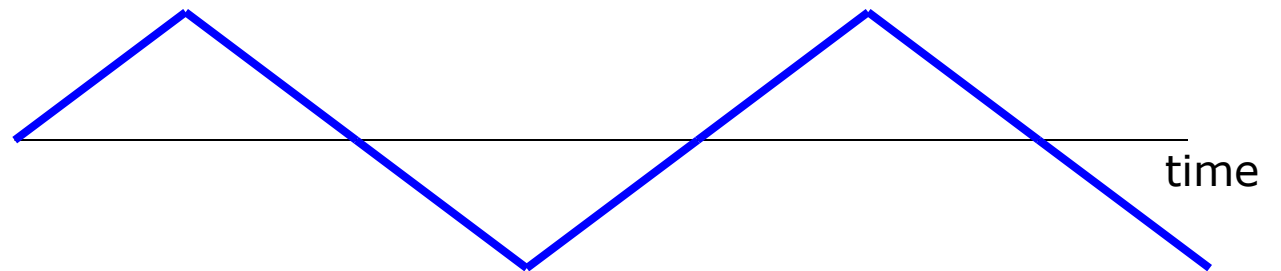
$$\overline{e^2(t)} = \frac{1}{T} \int_{-T/2}^{T/2} e^2(t) dt$$

Time average Noise



$$= \int_{-\infty}^{\infty} v^2 p(v) dv$$

Continuous Random Variable



# Signal to Noise Ratio[1]

$$\begin{aligned}\overline{e^2}(t) &= \int_{-\Delta v/2}^{\Delta v/2} v^2 \frac{1}{\Delta v} dv = \frac{1}{\Delta v} \left[ \frac{v^3}{3} \right]_{-\Delta v/2}^{\Delta v/2} \\ &= \frac{1}{3\Delta v} \left[ \frac{\Delta v^3}{8} + \frac{\Delta v^3}{8} \right] \\ &= \frac{\Delta v^2}{12}\end{aligned}$$

$$\begin{aligned}SNR|_Q &= \frac{\overline{f^2}(t)}{\overline{e^2}(t)} = \frac{12}{\Delta v^2} \overline{f^2}(t) \quad ; \Delta v = \frac{2V}{M} \\ &= \frac{12}{4V^2} M^2 \overline{f^2}(t) \\ &= \frac{3M^2 \overline{f^2}(t)}{V^2} = \frac{3M^2}{\alpha} \quad \text{where} \quad \alpha = \frac{V^2}{\overline{f^2}(t)} = \frac{PeakPower}{AvgPower}\end{aligned}$$



# Signal to Noise Ratio[2]

$$SNR|_Q = \frac{3M^2}{\alpha}$$

In dB

$$SNR|_Q = 10 \log_{10} 3 + 20 \log_{10} M - 10 \log_{10} \alpha (dB)$$
$$= 4.77 + 20 \log_{10} M - 10 \log_{10} \alpha (dB)$$

**Encoding** : each quantization level is encoded into N binary digit

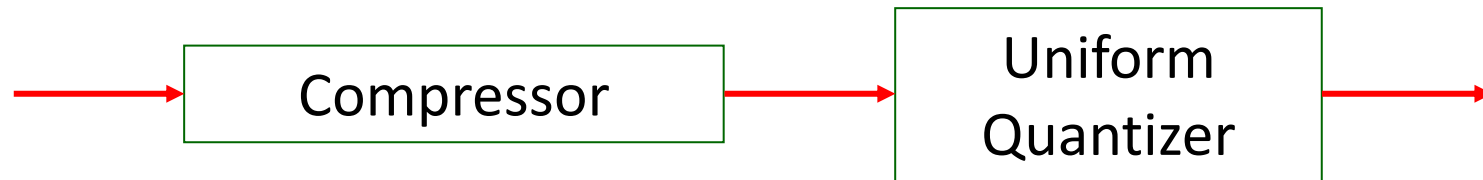
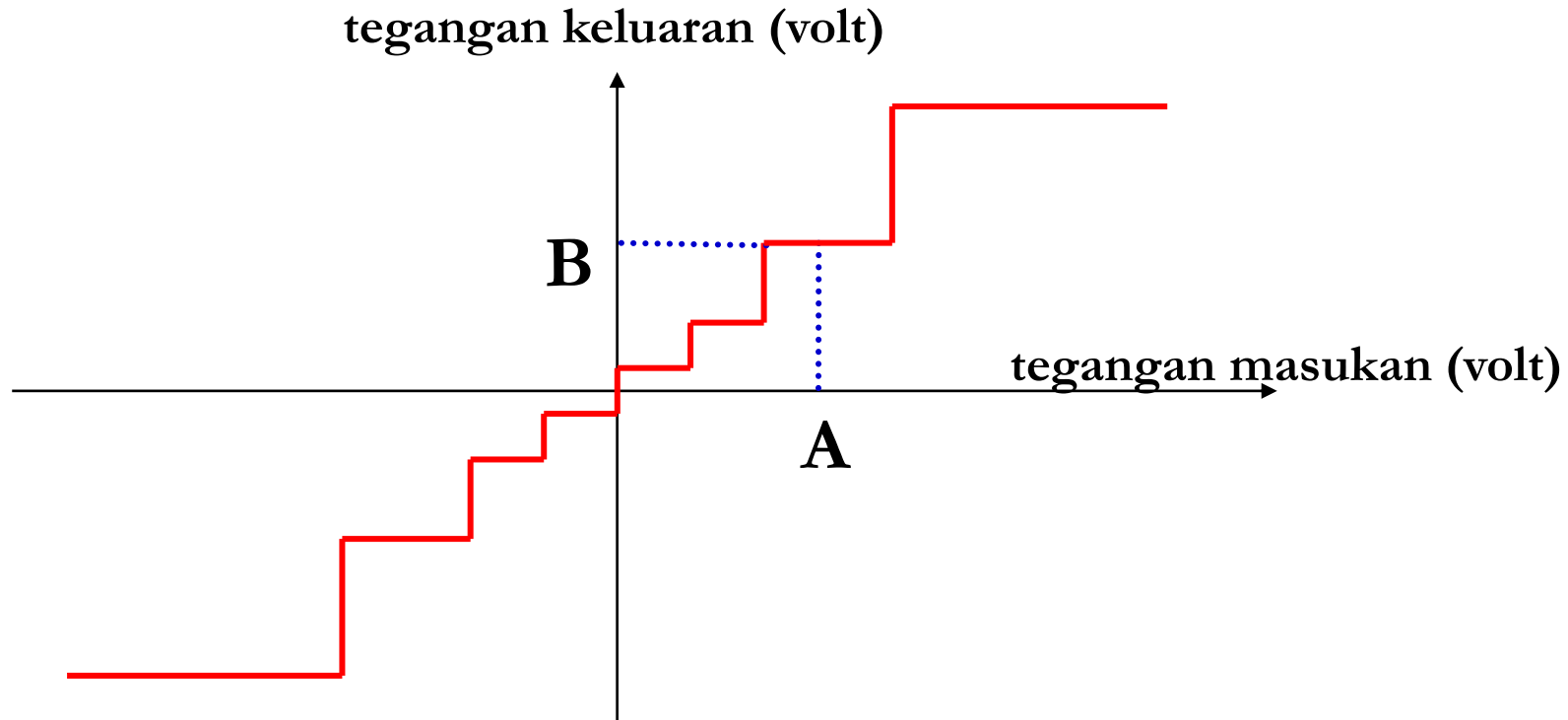
$$\therefore M = 2^N$$

$$\log_b a = \frac{\log_{10} a}{\log_{10} b}$$

$$N = \log_2 M \leftarrow \text{No.of level}$$

No.of binary digit per code word

# QUANTISER NON-UNIFORM



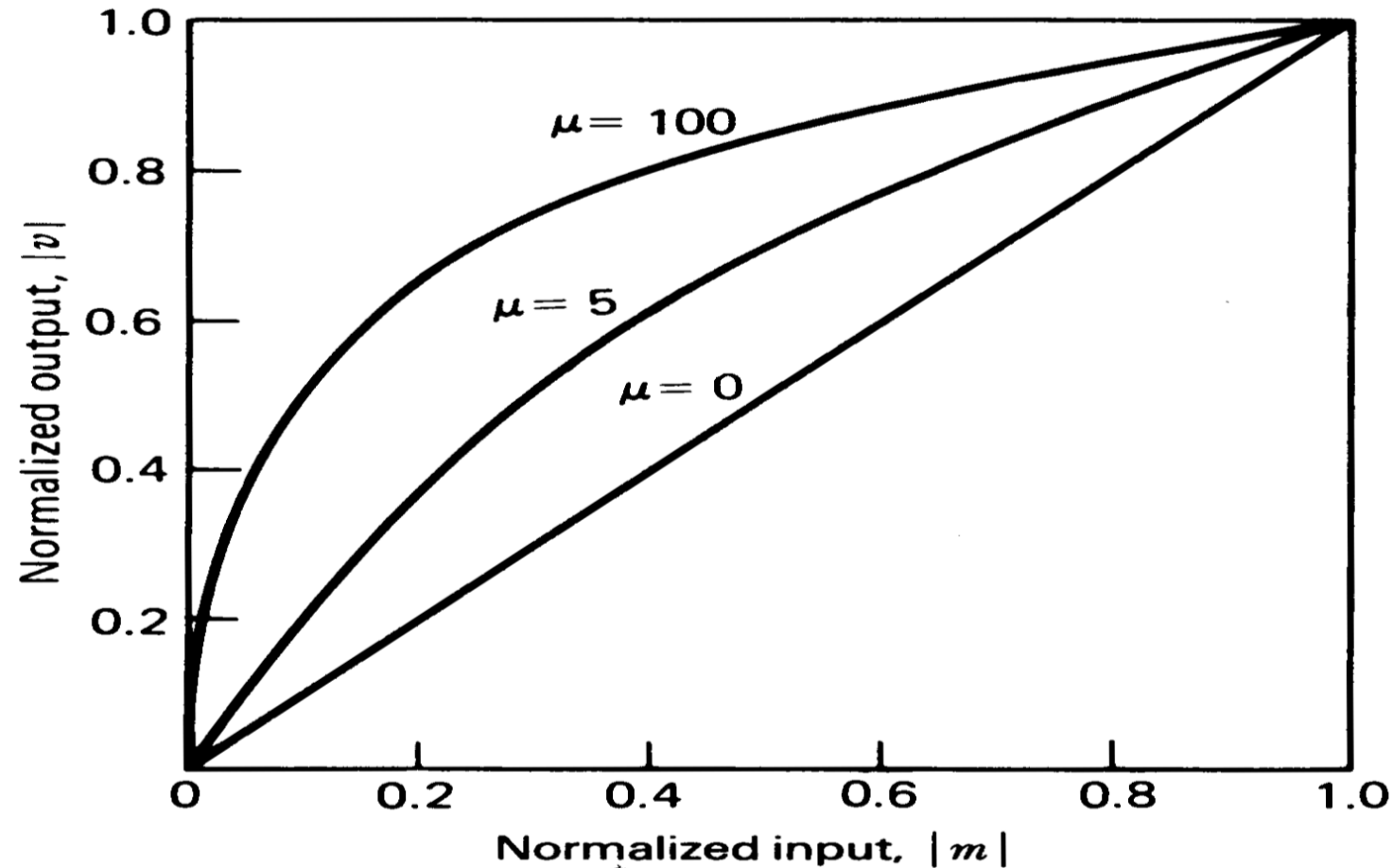
NonUniform / Nonlinear Quantizer

# QUANTISER NON-UNIFORM

□  $\mu$  - law

$$|v| = \frac{\log(1 + \mu |m|)}{\log(1 + \mu)}$$

- if  $\mu = 0 \rightarrow$  Uniform Quantizer



Standard  
Amerika Utara

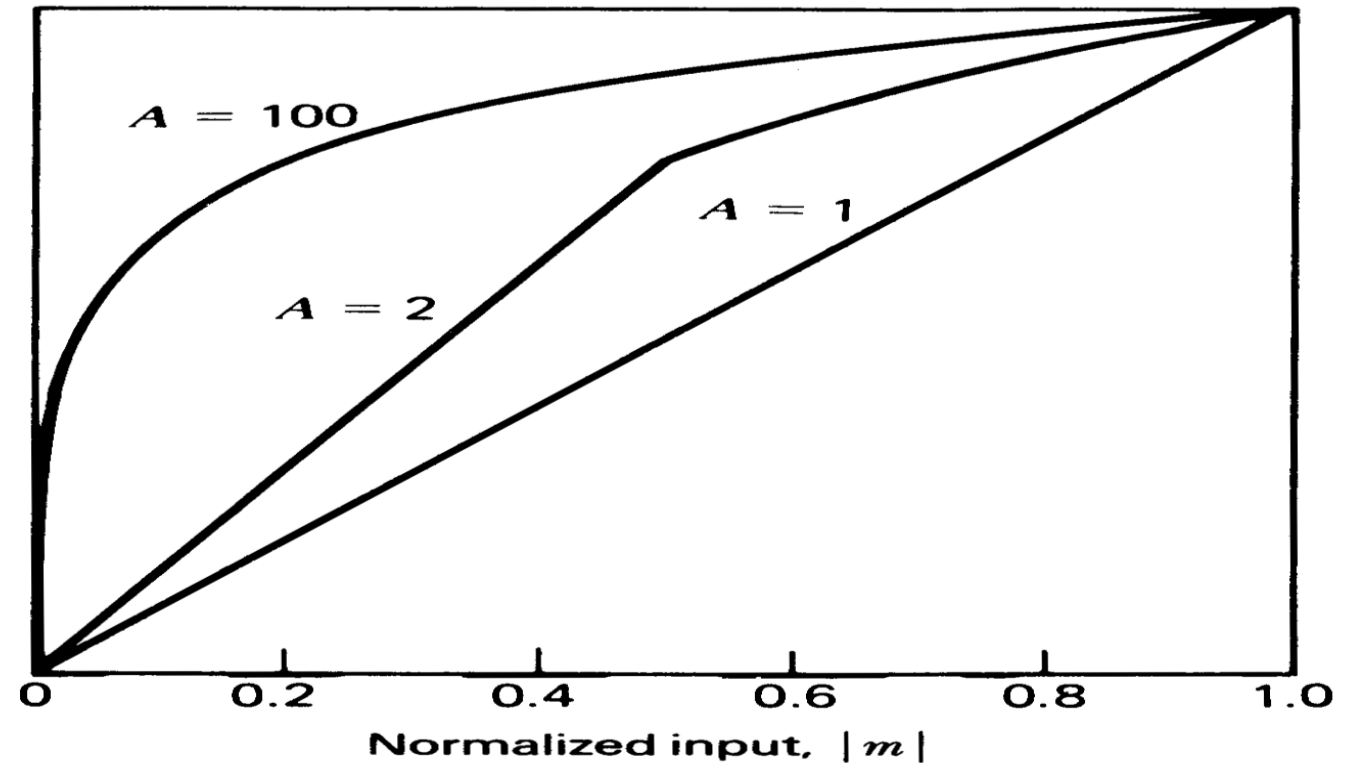
# QUANTISER NON-UNIFORM

A - law

$$|v| = \begin{cases} \frac{A|m|}{1 + \log A}, & 0 \leq |m| \leq \frac{1}{A} \\ \frac{1 + \log(A|m|)}{1 + \log A}, & \frac{1}{A} \leq |m| \leq 1 \end{cases}$$

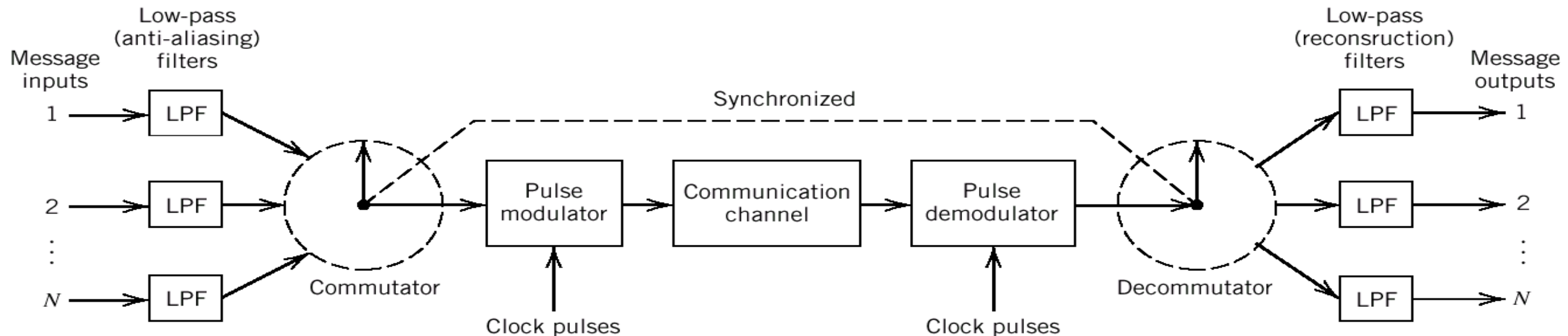
- $A = 1 \rightarrow$  Uniform Quantizer
- Practical value of  $A \Rightarrow A \cong 100$
- Reciprocal slope

**Standard Eropa  
(digunakan di Indonesia)**

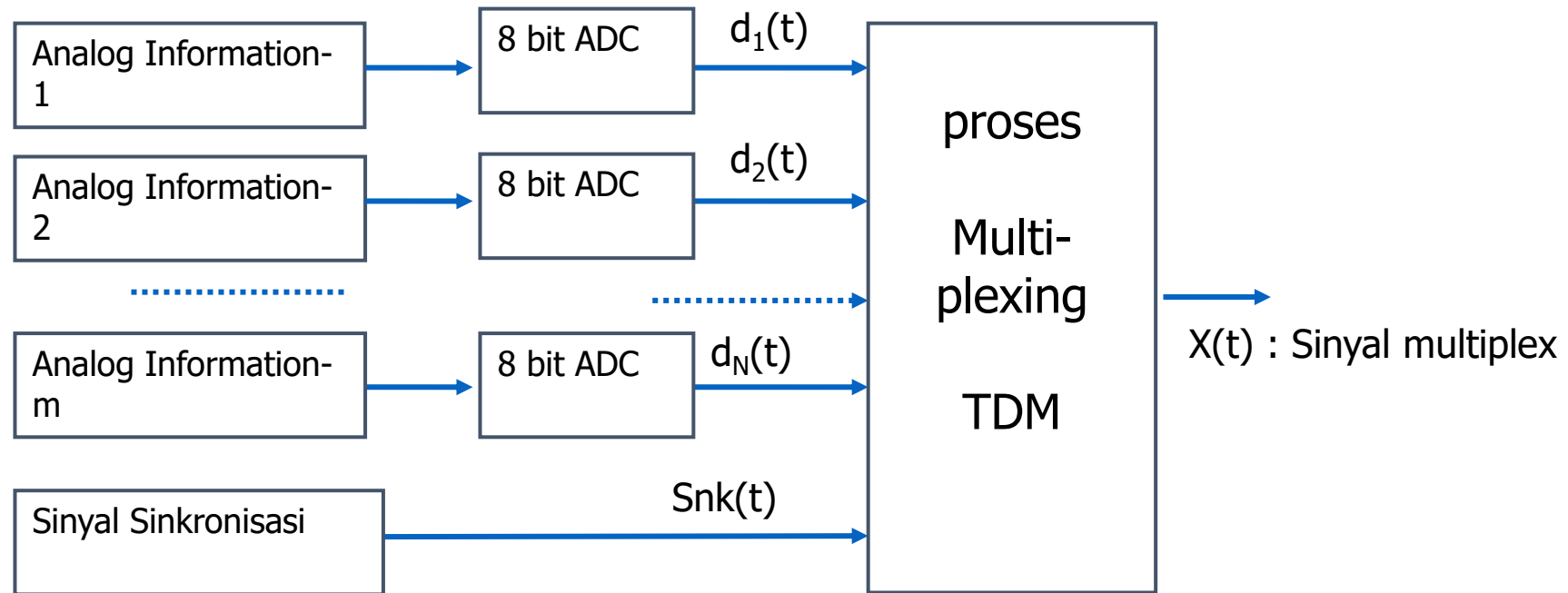


# Multiplexing TDM

- Multiplexing merupakan proses penggabungan beberapa kanal sinyal informasi kedalam satu kanal informasi dengan tujuan agar sinyal informasi dapat dikirimkan secara simultan dalam satu kanal
- Time Division Multiplexing merupakan proses multiplexing dengan cara membagi waktu menjadi slot-slot waktu yang menyatakan informasi dari tiap kanal
- TDM – PCM (Time Division Multiplexing – Pulse Code Modulation) merupakan proses multiplexing sinyal yang menggunakan teknik pengkodean PCM



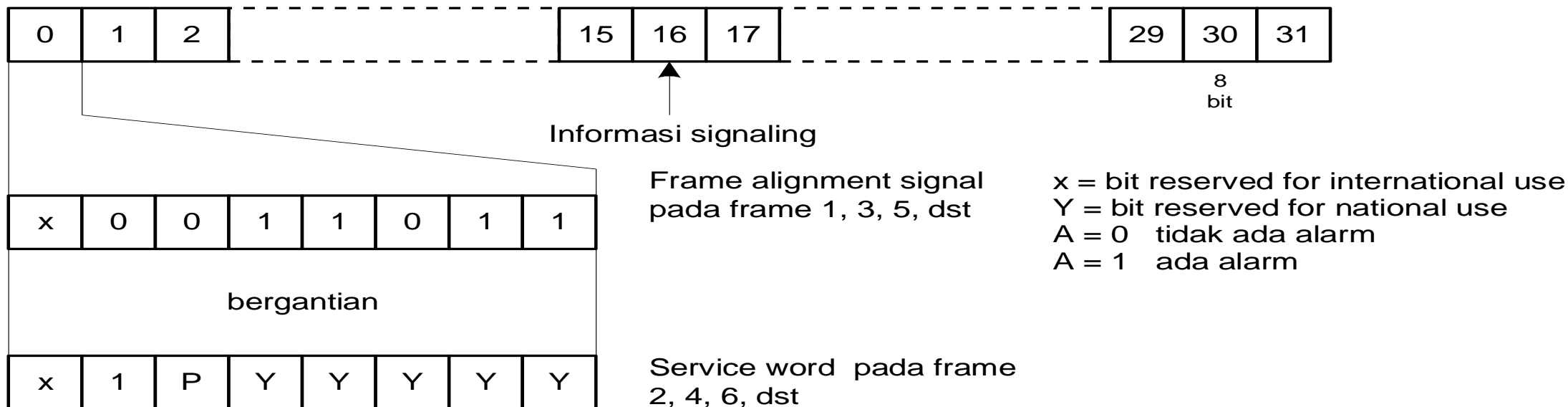
# Multiplexing TDM ( cont )



Standar TDM yang digunakan Indonesia adalah PCM-30 ( E1 ) yang mampu menggabungkan 30 kanal ( masing-masing 64 kbps ) menjadi sebuah sinyal multiplex TDM PCM dengan laju 2,048 Mbps

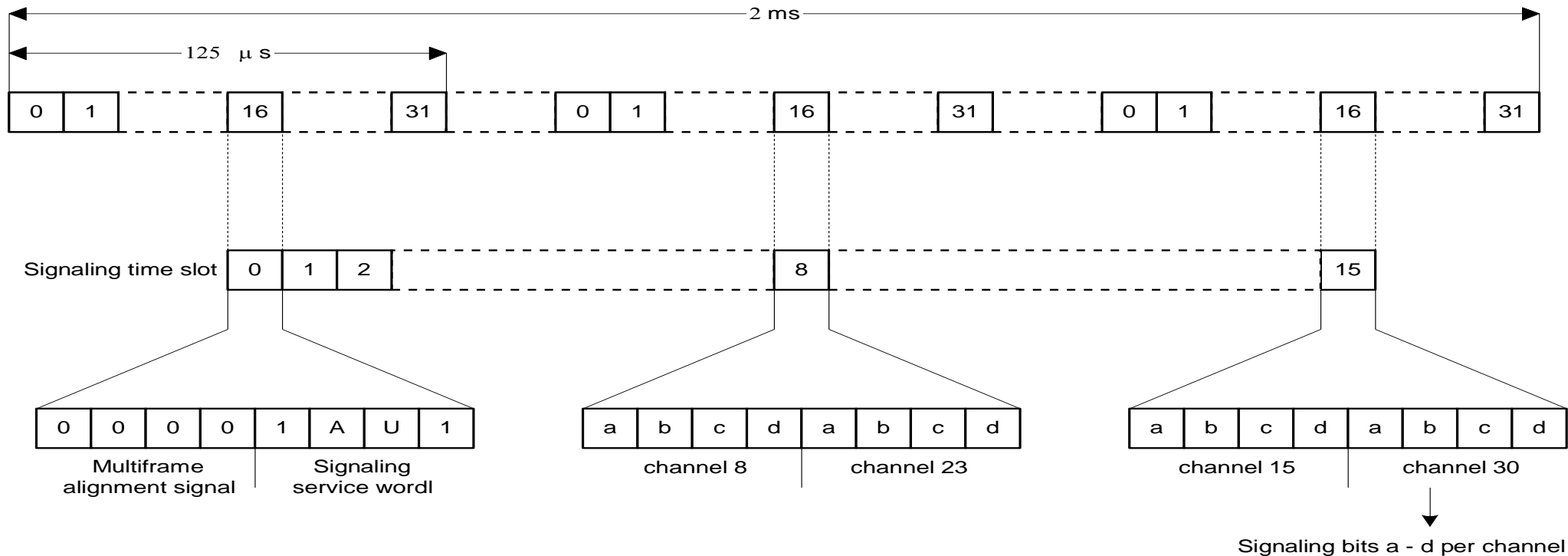
# PCM-30 (E-1, Standar Eropa)

1 - 15 dan 17 - 30 adalah sinyal  
 telephone yang dikodekan/ data digital



- 1 TS = 8 bit
- Terdiri dari 32 TS = 30 kanal suara + 1 sinkronisasi + 1 signaling
  - Sinkronisasi : TS 0
  - Signaling : TS 16
  - Voice : TS 1 – 15 + TS 17 – 31
- Dalam 1 detik tdp 8000 sample, sehingga :
  - Bit rate =  $(8 \times 8000) \times 32 = 2048 \text{ kbps}$

# Multiframe PCM-30



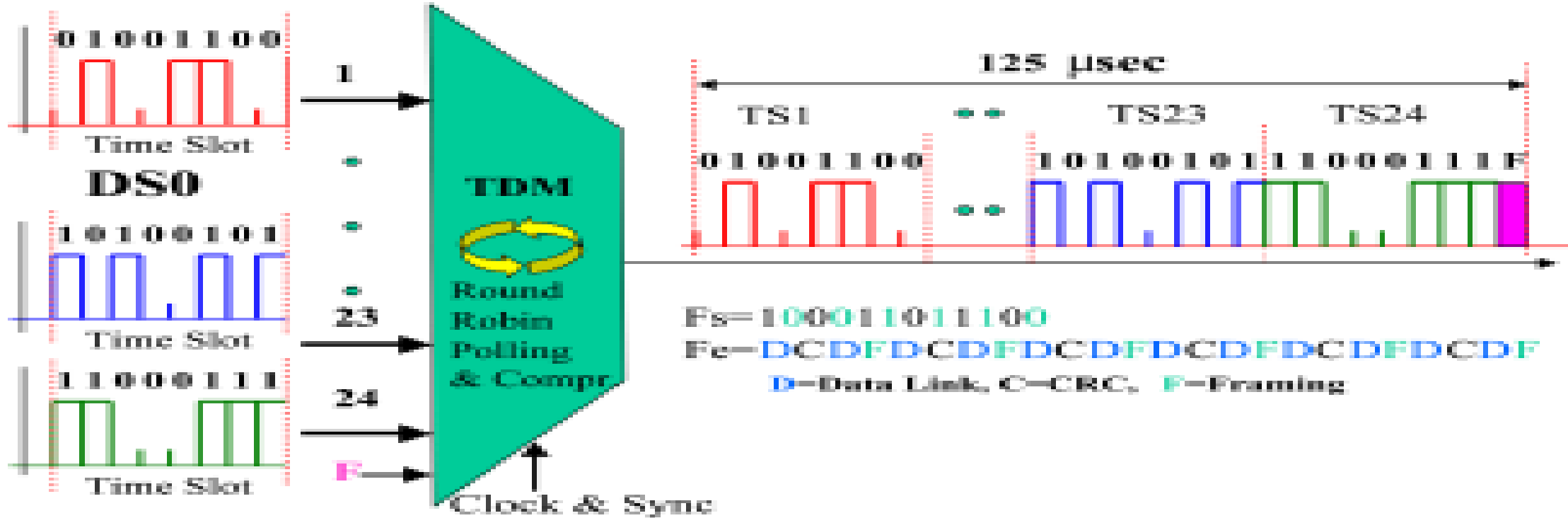
A = 0    Tidak ada Alarm  
 U = 1

A = 1    urgent alarm  
 U = 0    non urgent alarm

- 1 MF = 16 frame
- Signaling lengkap untuk 30 kanal voice (1 TS 16 untuk signaling 2 kanal voice)
- TS-16 untuk frame ke-0 digunakan untuk alignment / sinkronisasi multiframe



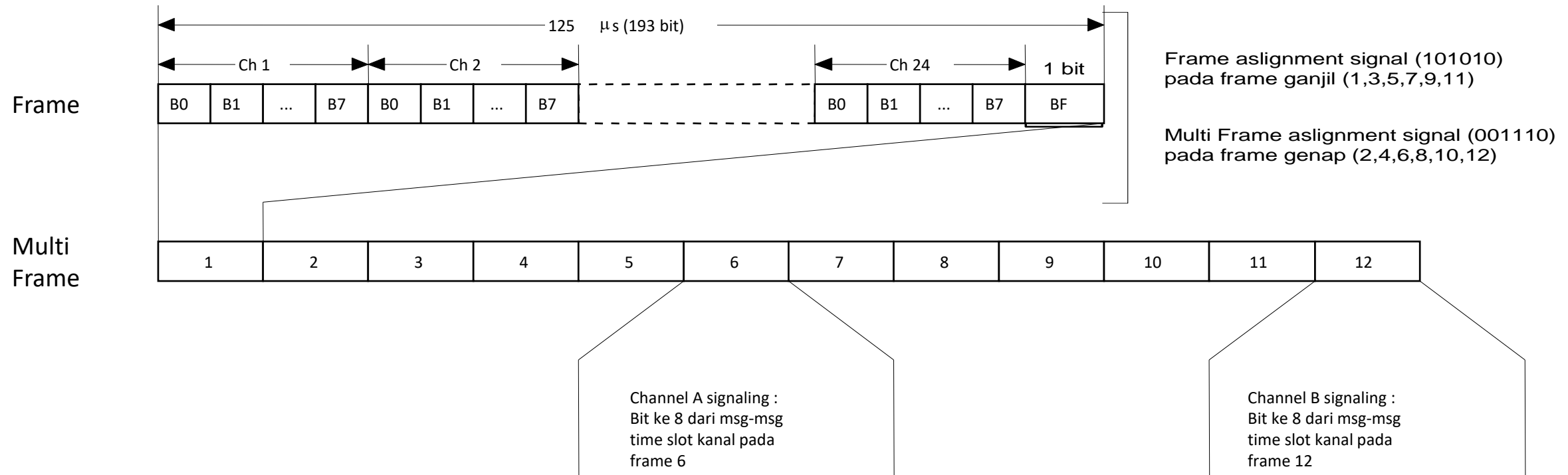
# PCM-24 (T-1, Standar Amerika)



## ■ T1 (DS-0) System

- 24 voice channels are time-division multiplexed
- Each voice signal is sampled at a rate of 8000 samples/sec. (sample duration = 125 μsec)
- Each sample is quantized in amplitude into one of 256 levels (8 bits are used to represent each level)
- T1 rate =  $(24 \cdot 8 + 1) / 125 \mu\text{sec} = 1.544 \text{ Mbps}$

# PCM-24 (T-1, Standar Amerika)



- 1 TS = 8 bit
- Terdiri dari 24 TS = 24 kanal suara Dalam 1 detik tdp 8000 sample
- Sinkronisasi menggunakan 1 bit tambahan (=BF)
- Signaling diambil pada bit ke-8 tiap TS pada frame ke-6 dan kelipatannya
- Bit Rate =  $((24 \times 8) + 1) \times 8000 = 193 \times 8000 = 1544 \text{ kbps}$
- 1 MF = 12 frame

## Perbandingan 3 standar (Amerika, Eropa, Jepang)

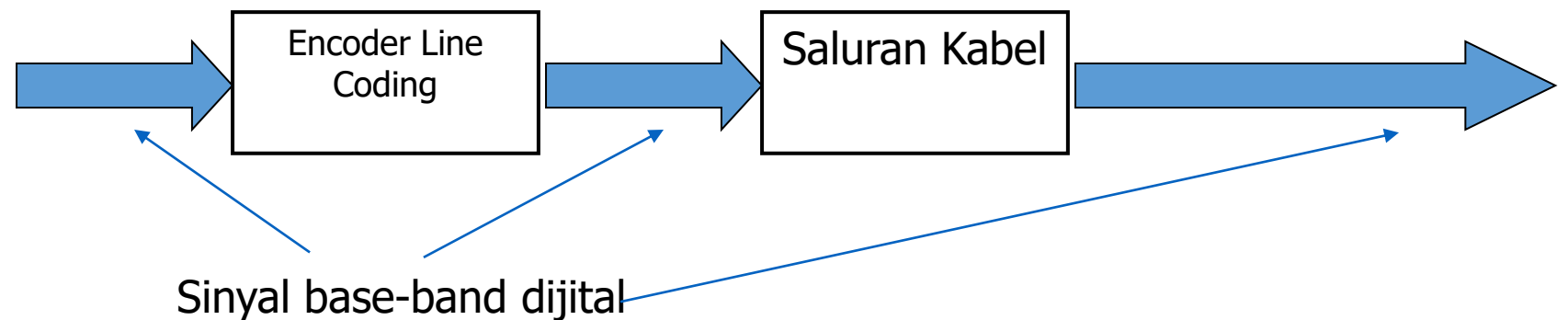
| Level | Eropa           | Amerika Utara | Jepang  |
|-------|-----------------|---------------|---------|
|       | Bit Rate (Mbps) |               |         |
| 1     | 2.048           | 1.544         | 1544    |
| 1C    | -               | 3.152         | -       |
| 2     | 8.448           | 6.312         | 6.312   |
| 3     | 34.368          | 44.736        | 32.064  |
| 4     | 139.264         | 274.176       | 97.728  |
| 5     | 564.992         |               | 400.352 |

- 1.544 Mbps = T1 = PCM-24 (Amerika)
- 2.048 Mbps = E-1 = PCM-30 (Eropa)
- Standar Jepang kurang populer
- Indonesia menggunakan sistem Eropa
- Internasional menggunakan Standard PCM-30

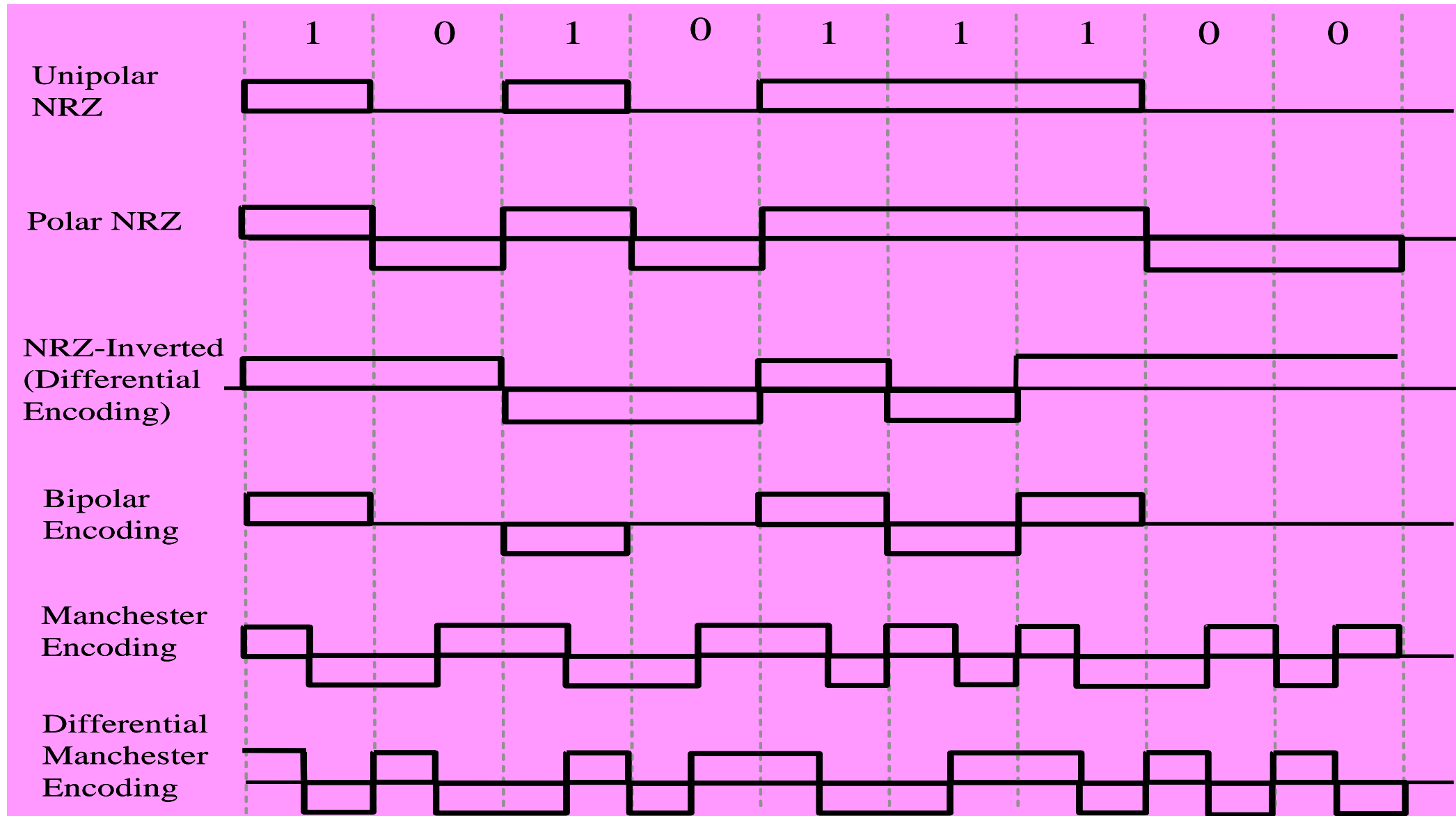
# Encoder (Konverter) Line coding

|                           | Two wire BW kecil<br>( misal kabel telepon ) | Two wire BW sedang<br>( misal kabel 2 Mbps )       | Coaxial                       |
|---------------------------|--|--|-------------------------------|
| <b>Output Line coding</b> | Rate kecil : bipolar , AMI , HDB-3 , B6ZS    | Rate kecil / sedang : bipolar , AMI , HDB-3 , B6ZS | bilpolar , AMI , HDB-3 , B6ZS |
|                           | Rate sedang / besar : Sinyal multi level     | Rate besar : sinyal multi level                    |                               |

- output ADC
- sinyal TDM
- Sinyal data text
- Output scrambler
- Output FEC



# Line coding

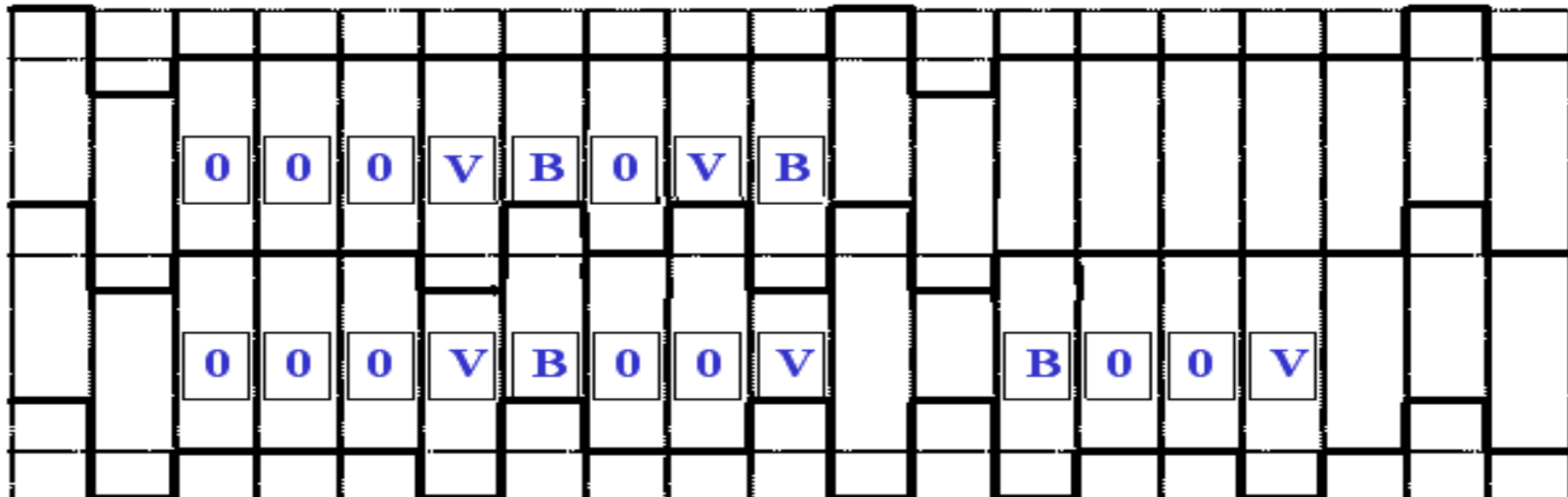


# Line Coding: Bipolar-AMI vs HDB3 dan B8ZS

- Deretan panjang nol dikodekan sbg ketidakadaan sinyal yg panjang. Clock receiver dpt kehilangan sync.
- Deretan nol yg panjang diganti dg pelanggaran (violation) transisi sinyal yang
  - Menghasilkan transisi sinyal yg cukup utk clock resynchronization,
  - Mengkodekan jumlah nol muncul

1 1 0 0 0 0 0 0 0 0 1 1 0 0 0 0 0 1 0

**Bipolar AMI**



# End of Module 9

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