Communication Systems

Xiong Shu Hua

Associate Professor

School of Electronics and Information Engineering

Sichuan University

E-mail：shxiong@scu.edu.cn

Chapter 3

Pulse Modulation
脉冲调制

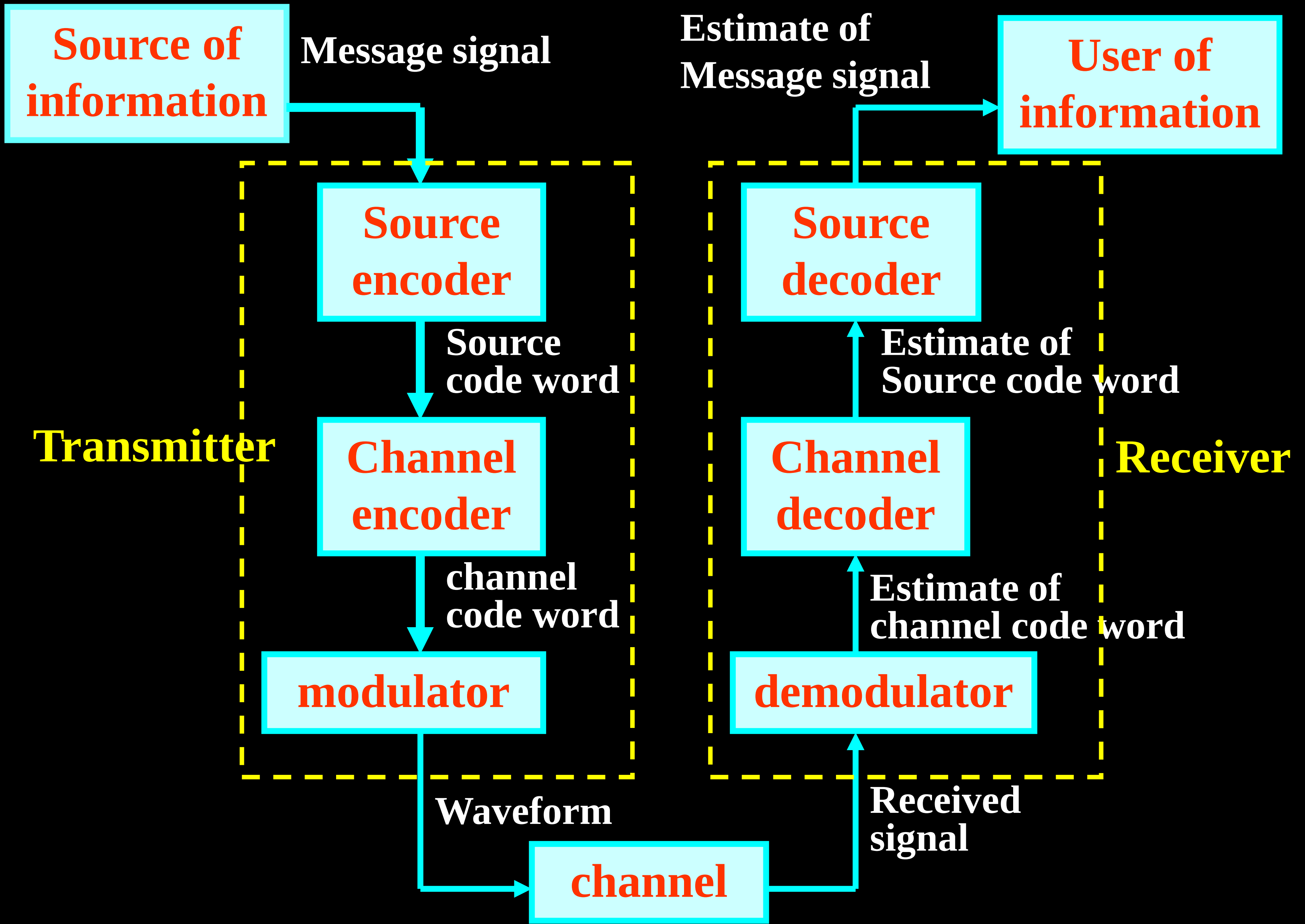
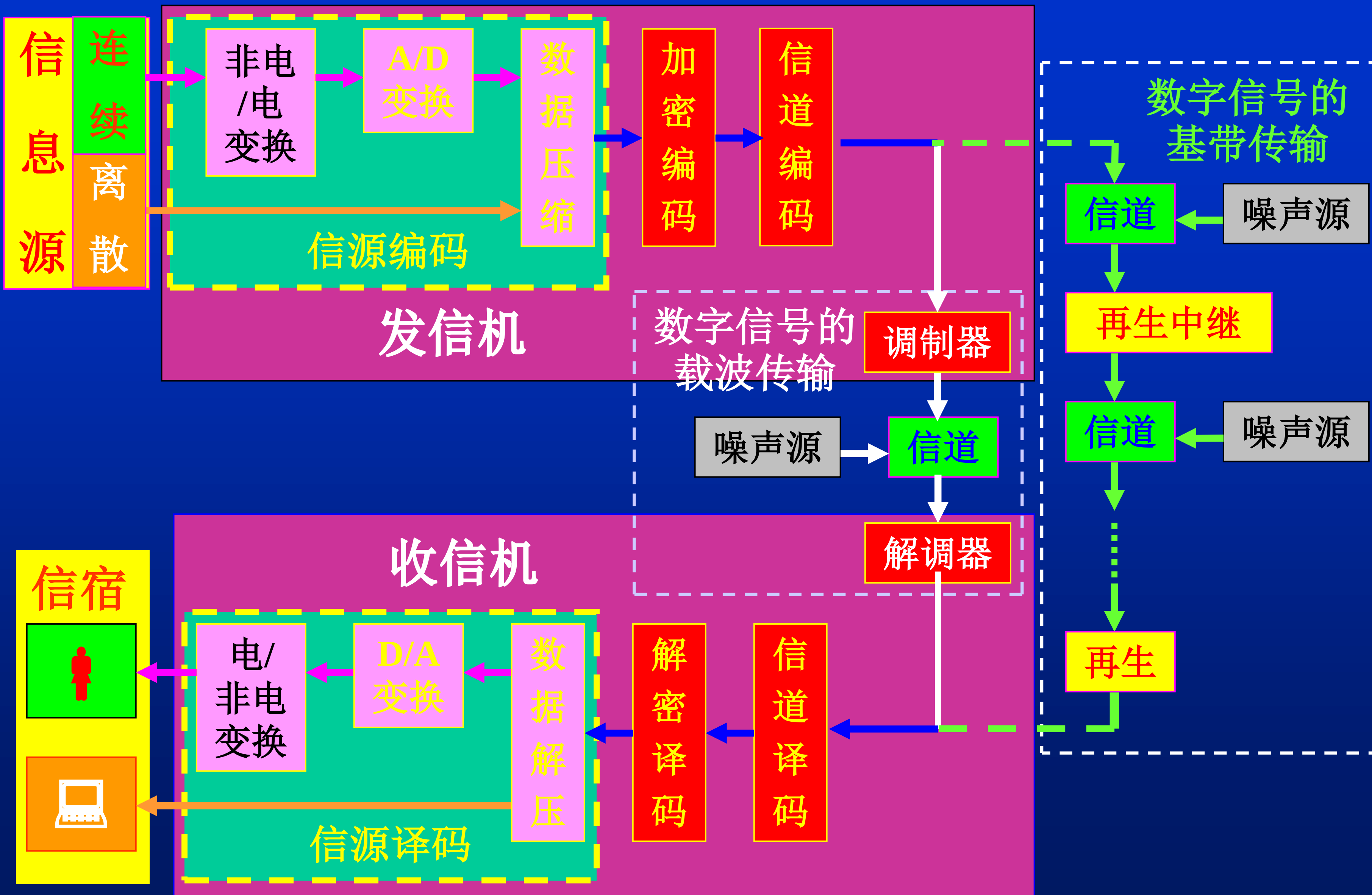


Figure 9 Block diagram of digital communication system.

数字通信系统



信息源
连续
离散
非电/电变换
A/D变换
数据压缩
信源编码
加密编码
信道编码
发信机
数字信号的载波传输
调制器
噪声源
信道
解调器
收信机
数字信号的基带传输
信道
噪声源
再生中继
信道
噪声源
再生
信宿
电/非电变换
D/A变换
数据解压
信源译码
解密译码
信道译码

信源编码的目的与任务

目的：提高通信的有效性

任务：模拟信号数字化

数据压缩

contents

- Sampling
- Pulse-amplitude modulation PAM
- Quantization
- Pulse-code modulation PCM
- Time-division multiplexing TDM
- Digital multiplexers
- Improved PCM: DM , DPCM, ADPCM
- MPEG-1/audio coding standard

3.1 Introduction

- Continuous-wave modulation (CW)
- Pulse modulation (PM)

The carrier consists of a periodic sequence of rectangular pulses. some parameter of a pulse train 脉冲序列 is varied with the message signal.

Pulse parameters:

1. Amplitude
2. Position
3. Width/duration

Pulse modulation types

Analog pulse modulation
脉冲模拟调制

- **Pulse-amplitude modulation (PAM)**
  脉冲幅度调制
- **Pulse-duration modulation (PDM)**
  脉冲宽度（持续时间）调制
- **Pulse-position modulation (PPM)**
  脉冲位置调制

Digital pulse modulation
脉冲数字调制

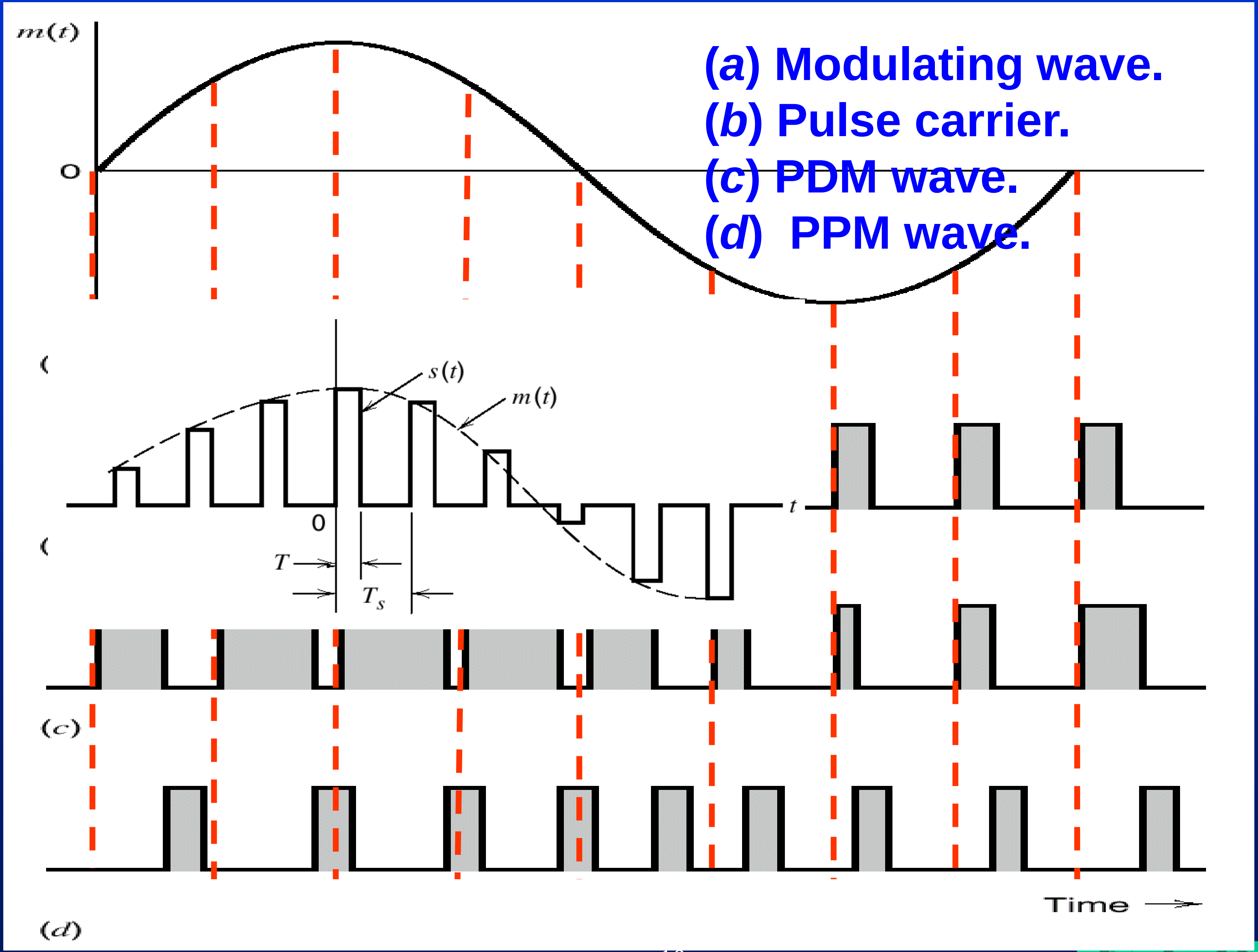
- **Pulse code modulation (PCM)**
  脉冲编码调制
- 改进形式：**DPCM, ADPCM, ΔM**

3.4 Other Forms of Pulse Modulation

1. Pulse amplitude modulation
   PAM 脉冲幅度调制

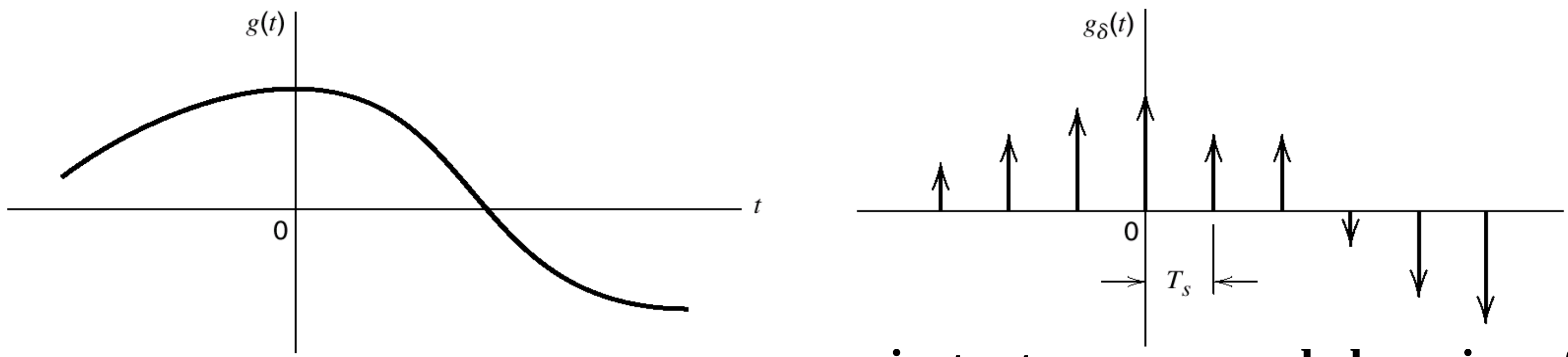
2. Pulse-duration modulation
   (Pulse-width modulation)
   PDM 脉冲宽度调制

3. Pulse-position modulation
   PPM 脉冲位置调制


m(t)
O
(a) Modulating wave.
(b) Pulse carrier.
(c) PDM wave.
(d) PPM wave.
s(t)
m(t)
0
t
T
T_s
(c)
(d)
Time

3.2 Sampling Process

Sampling: an analog signal is converted into a corresponding sequence of samples.



Analog signal

instantaneous sampled version of the analog signal

Ts: sampling period ***fs*=1/Ts: sampling rate**

Instantaneous sampling 瞬时抽样
ideal sampling 理想抽样

Question: how to choose the sampling rate properly so that the sequence of samples uniquely define the original analog signal?

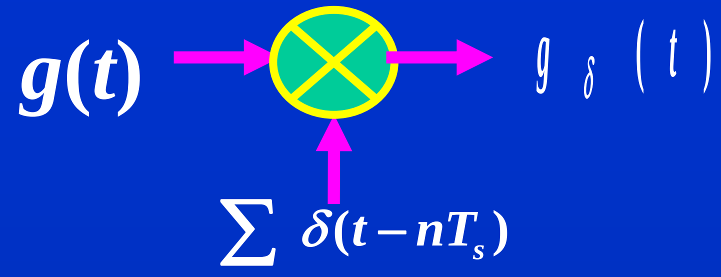
Sampling theorem answers this question.

$$\text{抽样}\begin{cases}\text{低通信号的抽样}\\\text{带通信号的抽样}\end{cases}$$

低通信号抽样定理

一个频带限制在（0，W）内的模拟信号$g(t)$，如果抽样频率 $f_s \geq 2W$ 则可以由抽样间隔为 $T_s \leq \frac{1}{2W}$ 的抽样值序列无失真地重建原始信号$g(t)$。

Ideal sampling process



Ideal sampled signal:

$$g_\delta(t) = g(t)\cdot \sum_{n=-\infty}^{\infty} \delta(t-nT_s) = \sum_{n=-\infty}^{\infty} g(nT_s)\delta(t-nT_s) \qquad 3.1$$

Fourier-transform pairs:

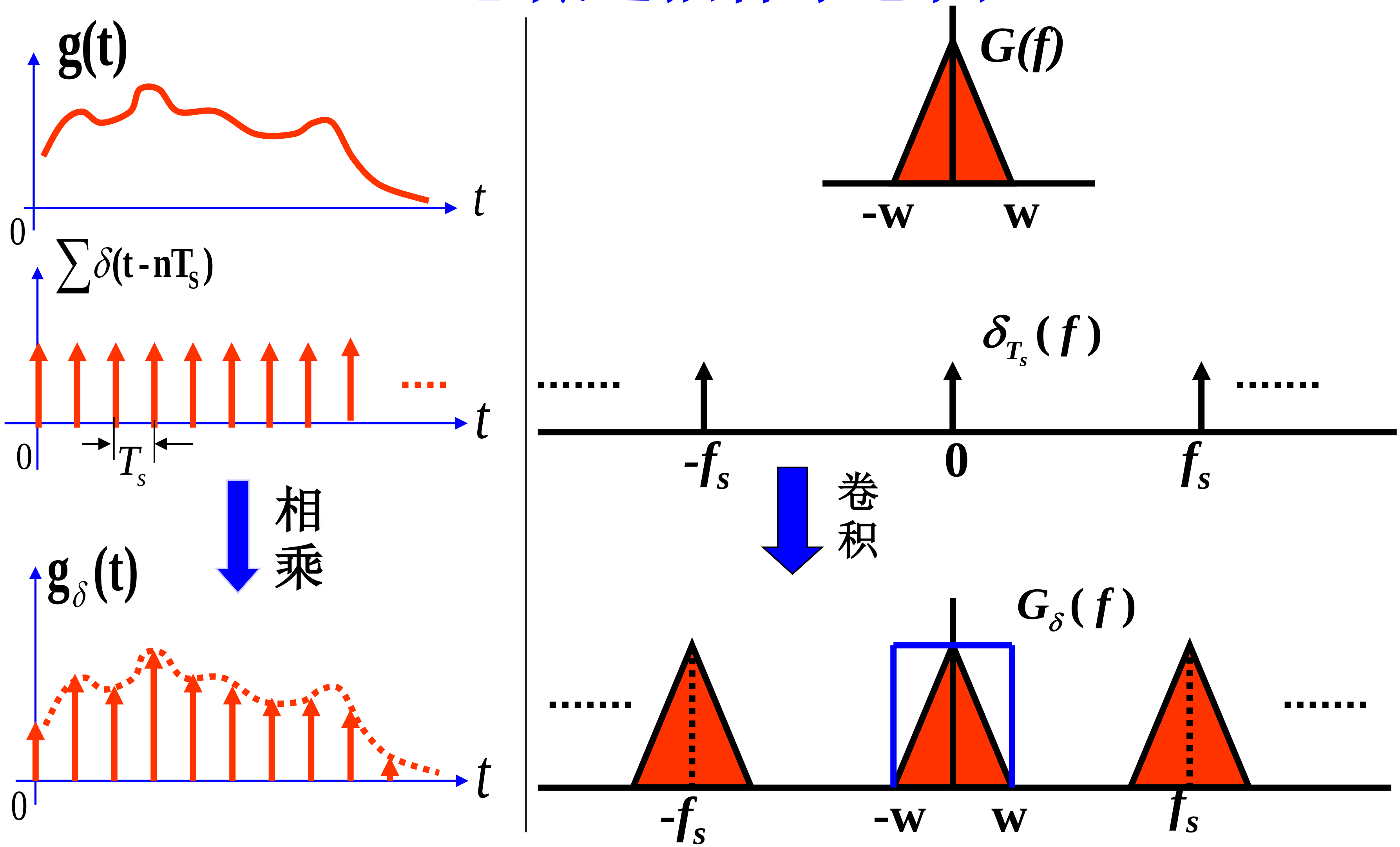
$$g(t) \xleftrightarrow{FT} G(f)$$

$$\sum_{n=-\infty}^{\infty} \delta(t-nT_s) \Leftrightarrow \frac{1}{T_s}\sum_{m=-\infty}^{\infty} \delta\left(f-m\frac{1}{T_s}\right) = f_s \sum_{m=-\infty}^{\infty} \delta(f-mf_s)$$

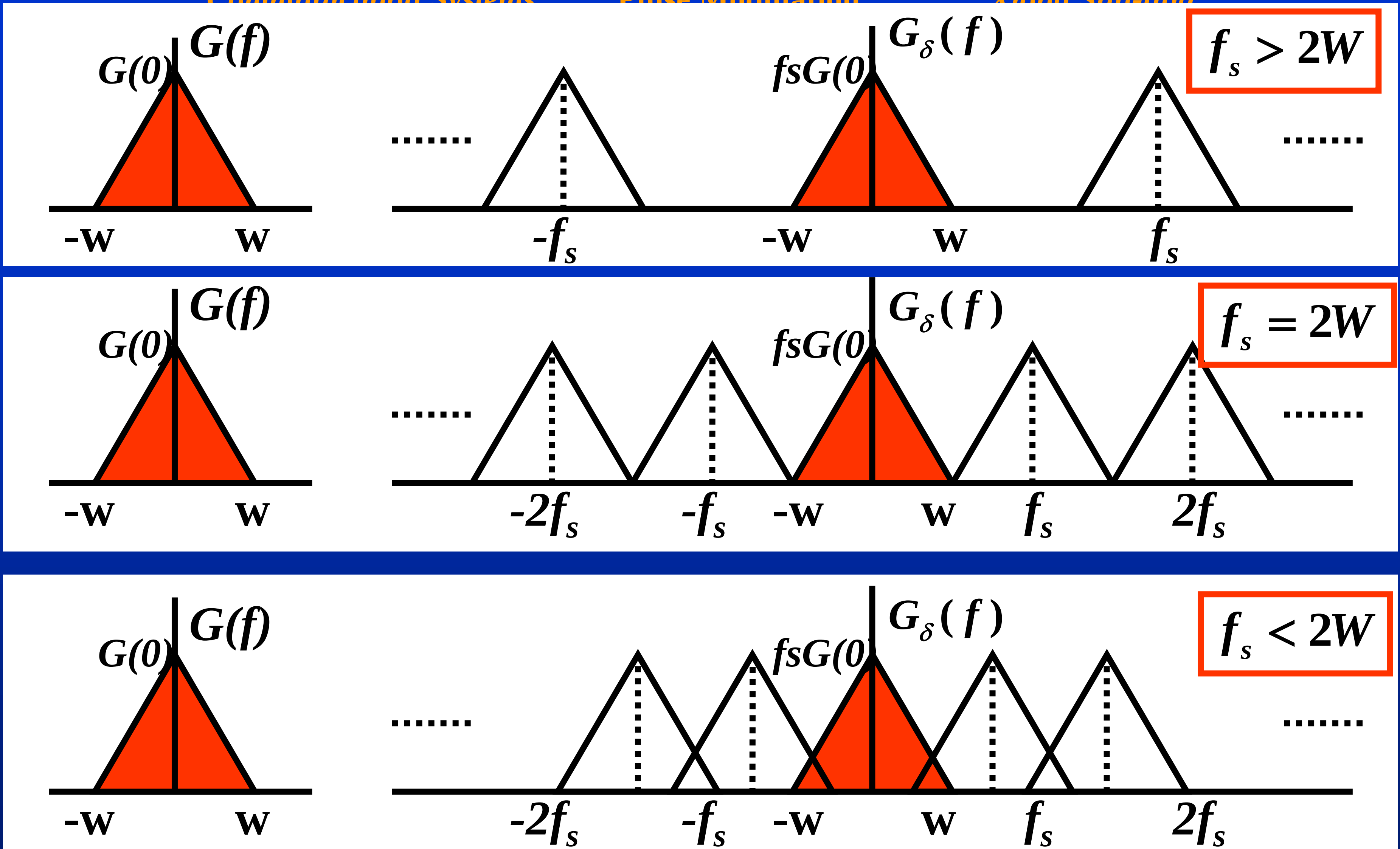
Then

$$g_\delta(t) \xleftrightarrow{FT} G_\delta(f) = f_s \sum_{m=-\infty}^{\infty} G(f-mf_s) \qquad 3.2$$

理想低通抽样示意图



How to reconstruct the original signal *g(t)* ?



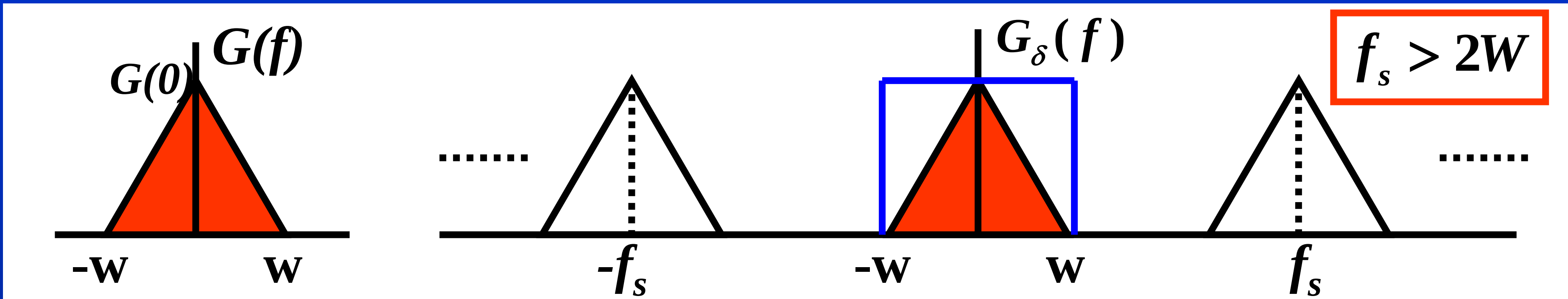
Nyquist rate 奈奎斯特速率: 2W

Nyquist interval 奈奎斯特间隔: 1/(2W)

Undersampling 欠采样

Aliasing 混迭现象

Reconstruction of the original signal *g(t)*

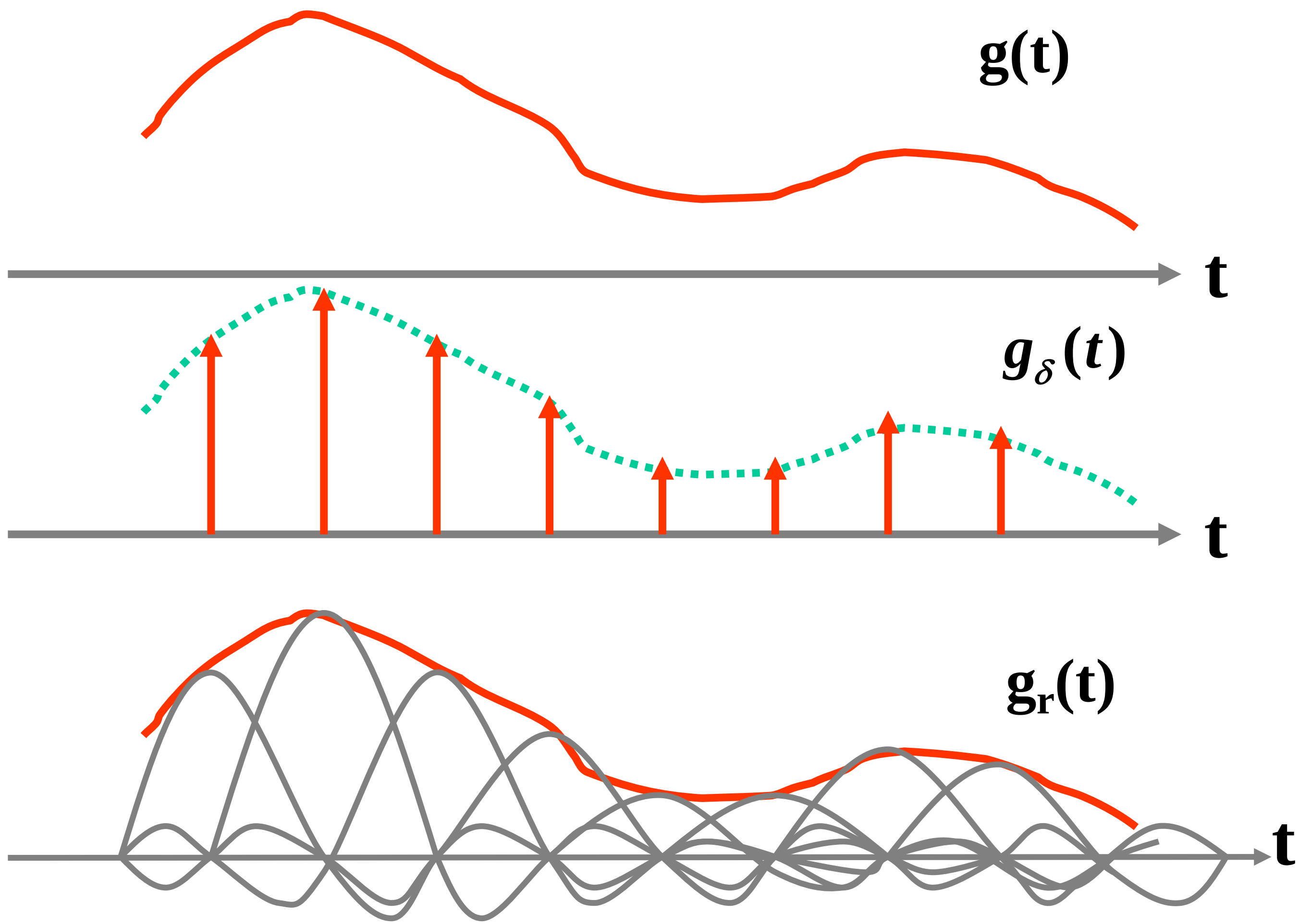


$$g(t) = \sum_{n=-\infty}^{\infty} g\left(\frac{n}{2W}\right)\frac{\sin(2\pi Wt - n\pi)}{(2\pi Wt - n\pi)} = \sum_{n=-\infty}^{\infty} g\left(\frac{n}{2W}\right)\operatorname{sinc}(2\pi Wt - n\pi) \quad 3.9$$

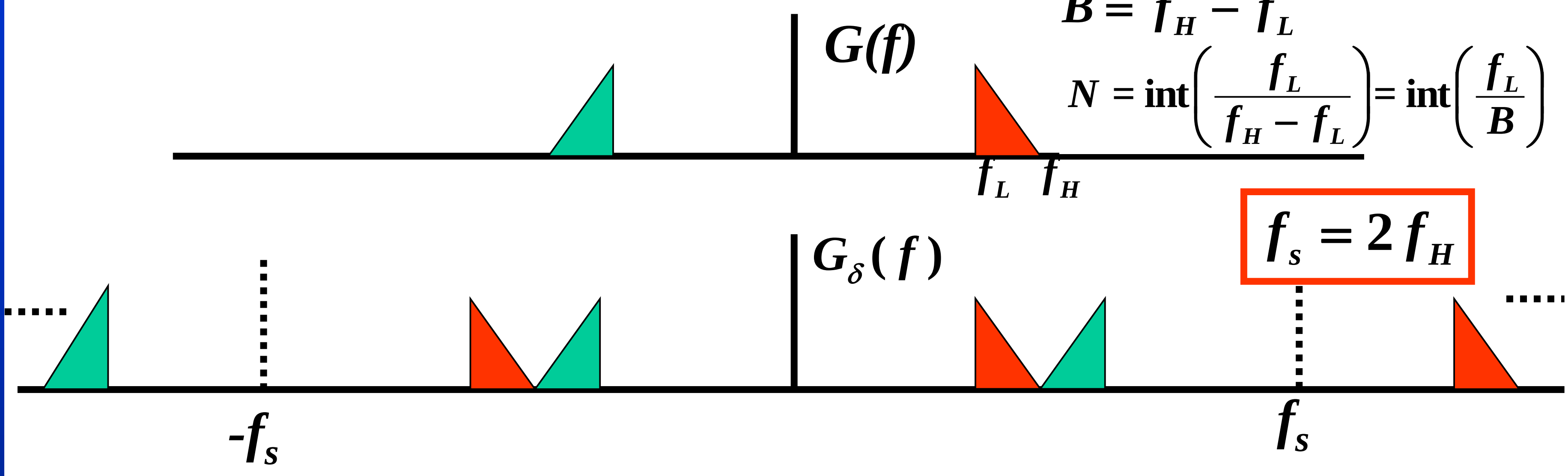
Interpolation formula 内插公式

Interpolation function 内插函数

理想采样与理想带限内插示意图



补充：带通信号的抽样



$$B = f_H - f_L$$

$$N = \text{int}\left(\frac{f_L}{f_H - f_L}\right) = \text{int}\left(\frac{f_L}{B}\right)$$

$$\boxed{f_s = 2f_H}$$

带通信号，从原理上讲仍可按低通信号的抽样频率来抽样，但这时抽样频率将会很高，频谱中 0~ f_L 频段为空隙，没有被充分利用，使得信道利用率不高。

带通信号的抽样频率：

$$\frac{2f_H}{N+1} \le f_s \le \frac{2f_L}{N}$$

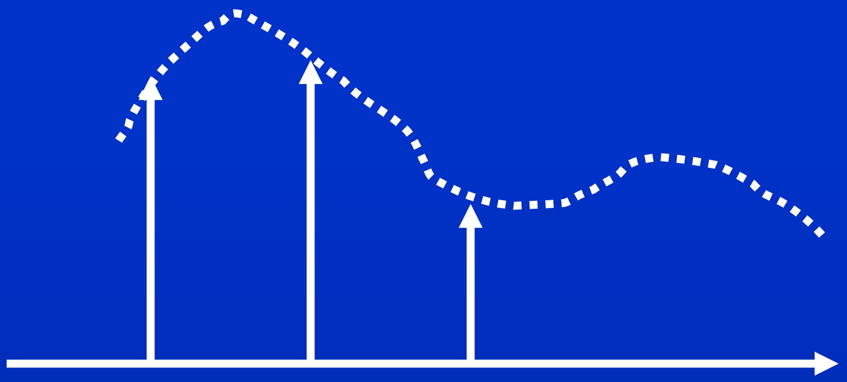
$$f_s = \frac{2(f_L + f_H)}{2N+1}$$

Practical sampling

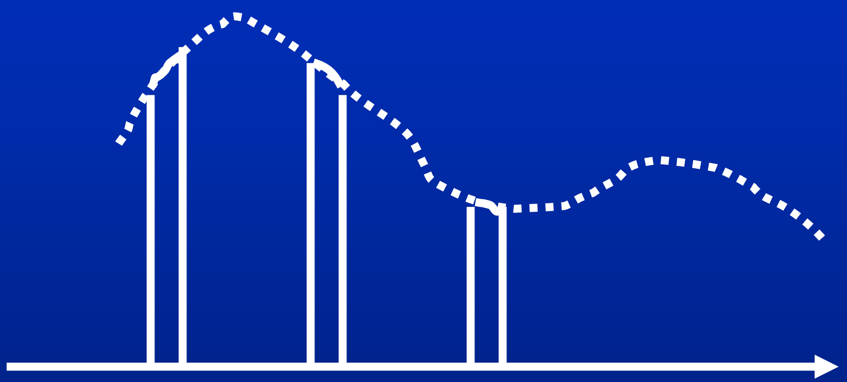
In practical, delta function is closely approximated by **a rectangular pulse**. The smaller the duration of pulse the better will be the approximation.

- Ideal sampling (理想抽样）
- Practical sampling （实际抽样）
  - Natural sampling （自然抽样）
  - Flat-top sampling: PAM （平顶抽样）

理想抽样

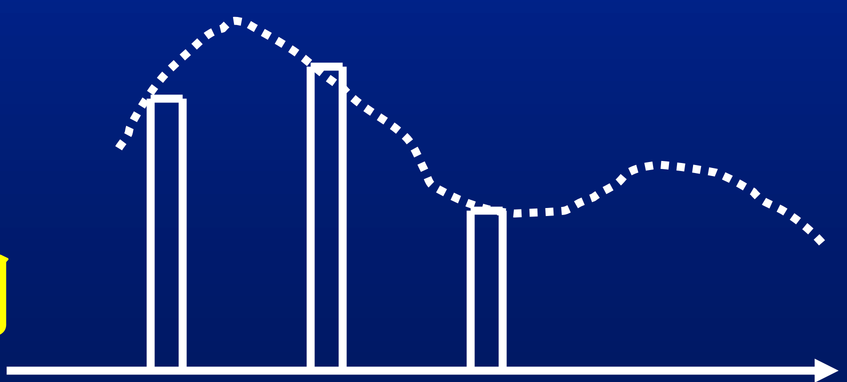


自然抽样



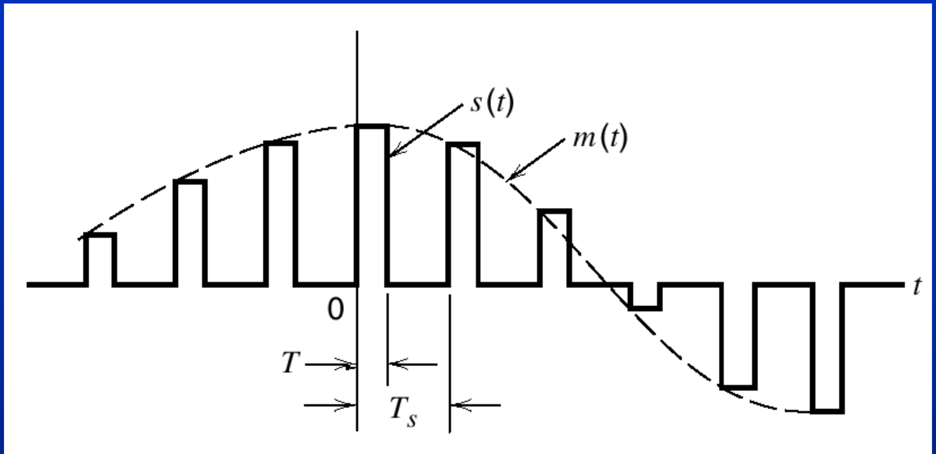
平顶抽样

PAM 脉冲幅度调制



3.3 Pulse-Amplitude Modulation PAM

PAM is the simplest and most basic form of **analog** pulse modulation.



PAM signal:

$$s(t) = \sum_{n=-\infty}^{\infty} m(nT_s)h(t - nT_s) \quad (3.10)$$

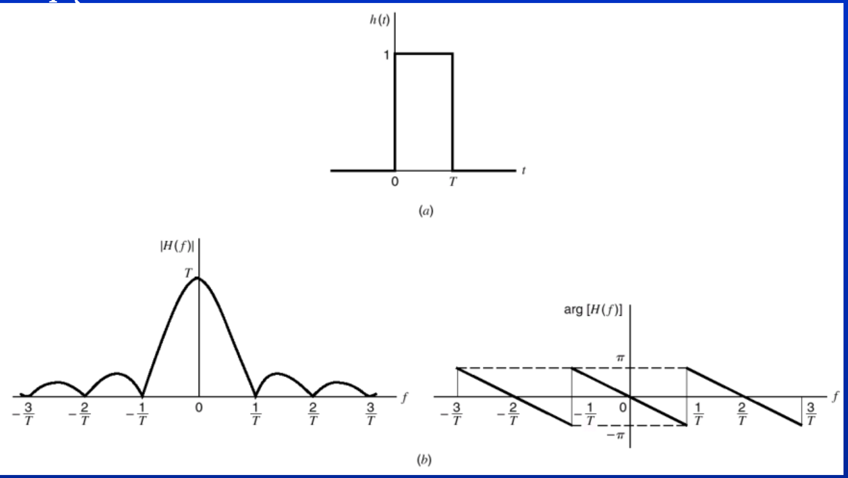
Two operations for generation of PAM signal:

1. Instantaneous sampling 2. Lengthening

Sample and hold 抽样-保持

Properties of Sample and hold circuit

$$h(t)=\begin{cases} 1, & 0<t<T \\ \frac{1}{2}, & t=0, t=T \\ 0, & \text{otherwise} \end{cases}$$



Instantaneously sampled version of m(t) is

$$m_\delta(t)=\sum_{n=-\infty}^{\infty} m(nT_s)\delta(t-nT_s)$$

$$s(t)=m_\delta(t)*h(t)$$

$$s(t)=\sum_{n=-\infty}^{\infty} m(nT_s)h(t-nT_s) \Rightarrow S(f)=f_s\sum_{k=-\infty}^{\infty} M(f-kf_s)H(f)$$

Comparison

Ideal sampling

$$G_\delta(f) = f_s \sum_{m=-\infty}^{\infty} G(f - mf_s) \qquad 3.2$$

Flat-top sampling or PAM

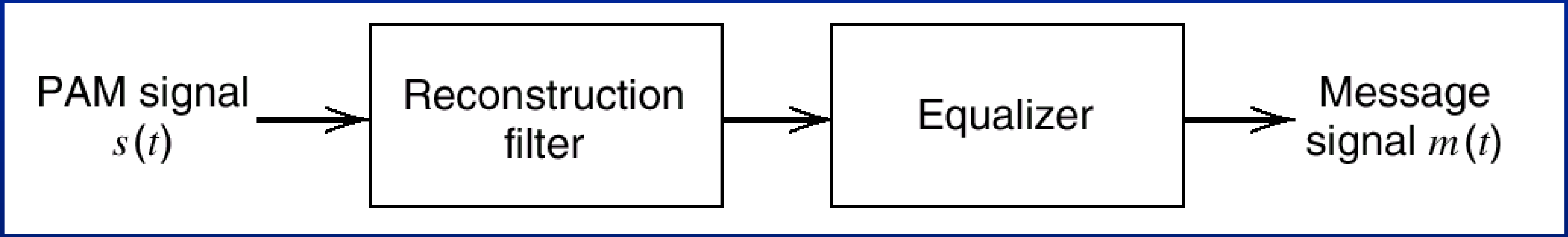
$$S(f) = f_s \sum_{k=-\infty}^{\infty} M(f - kf_s)H(f) \qquad 3.18$$

Compared to ideal sampling, we have *H(f)* here. It produces distortion.

Aperture effect 孔径效应

The distortion caused by the use of PAM to transmit an analog signal is referred to as the aperture effect.

Aperture distortion can be corrected by connecting an equalizer 均衡器 in cascade with the low pass reconstruction filter 重建滤波器.



The magnitude response of the equalizer is given by

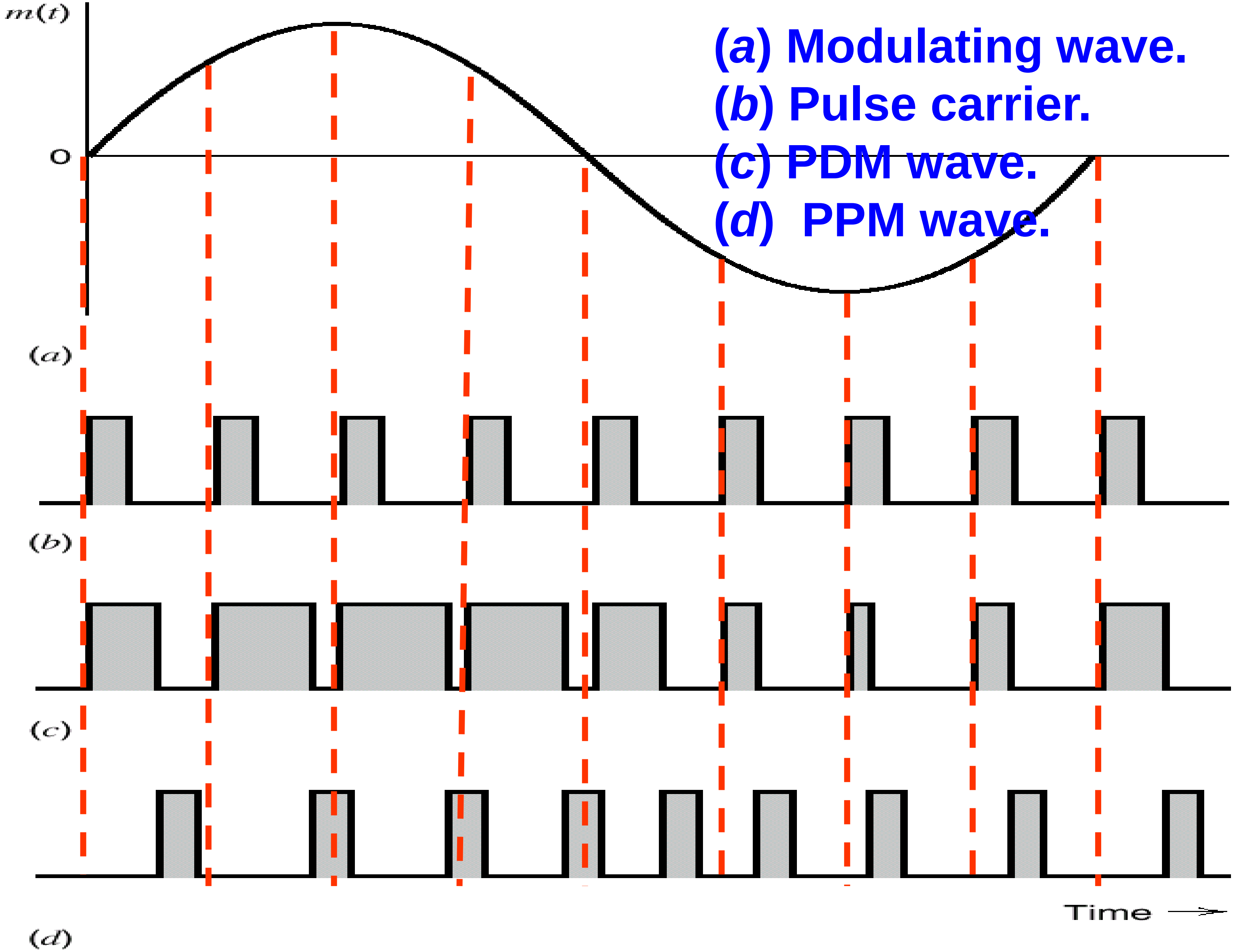
$$\frac{1}{H(f)} = \frac{1}{T\,\mathrm{sinc}(fT)} = \frac{\pi f}{\sin(\pi fT)}$$

3.4 Other Forms of Pulse Modulation

1. **Pulse amplitude modulation**
   PAM 脉冲幅度调制

2. **Pulse-duration modulation**
   (Pulse-width modulation)
   PDM 脉冲宽度调制

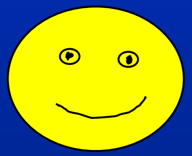
3. **Pulse-position modulation**
   PPM 脉冲位置调制



(*a*) Modulating wave.
(*b*) Pulse carrier.
(*c*) PDM wave.
(*d*) PPM wave.

3.5 Bandwidth-Noise Trade-off
带宽—噪声权衡

PPM and FM can improve noise performance by increasing transmission bandwidth. Both of them have a figure of merit proportional to the square (平方) of transmission bandwidth B_T.



Question: Can we produce a trade-off better than a square law?

YES !

Digital pulse modulation is the way to do it. The use of such a method is a radical departure from CW modulation. PCM is a basic form of digital pulse modulation.

PCM

In PCM, a message signal is represented in discrete form in both time and amplitude.

Sampling → **Quantizing** → **Encoding** →

PCM

Analog signal —sampling→ **Sampled signal** —Quantizing→ **Digital signal**

Continuous time continuous amplitude

Discrete time continuous amplitude

Discrete time discrete amplitude

3.6 Quantization Process

Definition: Amplitude quantization is defined as the process of transforming the sample amplitude into a discrete amplitude taken from a finite set of possible amplitudes.

- **Sampling process:**
  In theory, it can be a lossless process.
- **Quantization process:**
  It uses finite numbers of amplitudes to represent infinite sample amplitudes, which must cause loss of information. It is a lossy process even in theory.

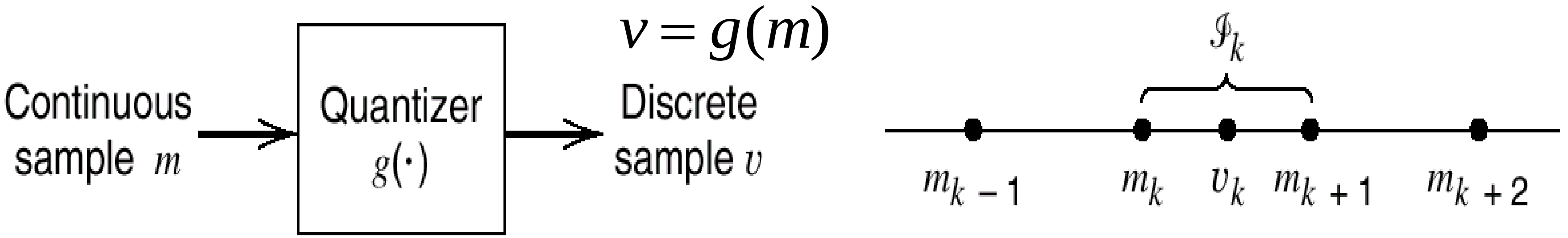


Fig. 3.9 Description of a quantizer

$$\varphi_k : \{ m_k < m \le m_{k+1} \} \qquad k = 1, 2, \cdots\cdots, L$$

m_k **(k=1,2,⋯L)** **decision levels 判决电平**
or decision thresholds 判决门限

V_k **(k=1,2,⋯L)** **representation levels表示电平**
or reconstruction levels 重建电平

$\Delta_k = v_{k+1} - v_k \quad (k = 1, 2, \cdots L-1)$ **quantum量阶**
or step-size 量化步长

Quantizer characteristic 量化特性

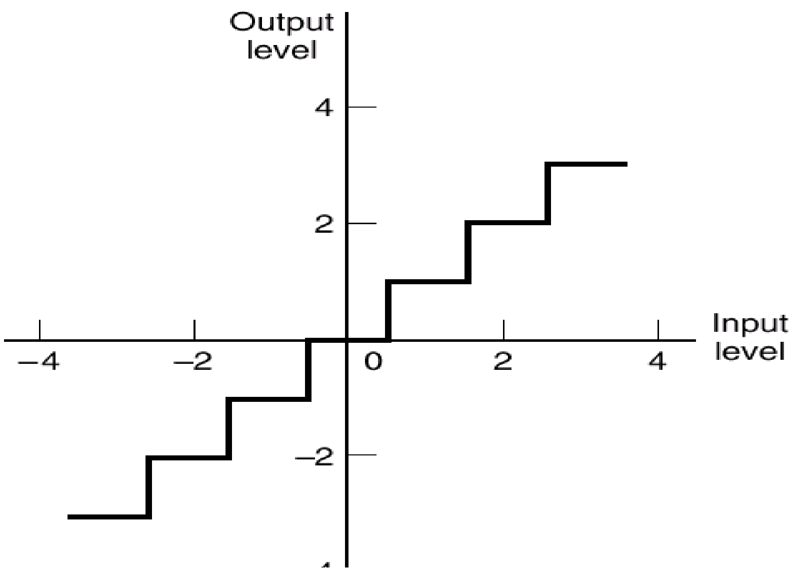
$$v = g(m)$$

It is a staircase function.

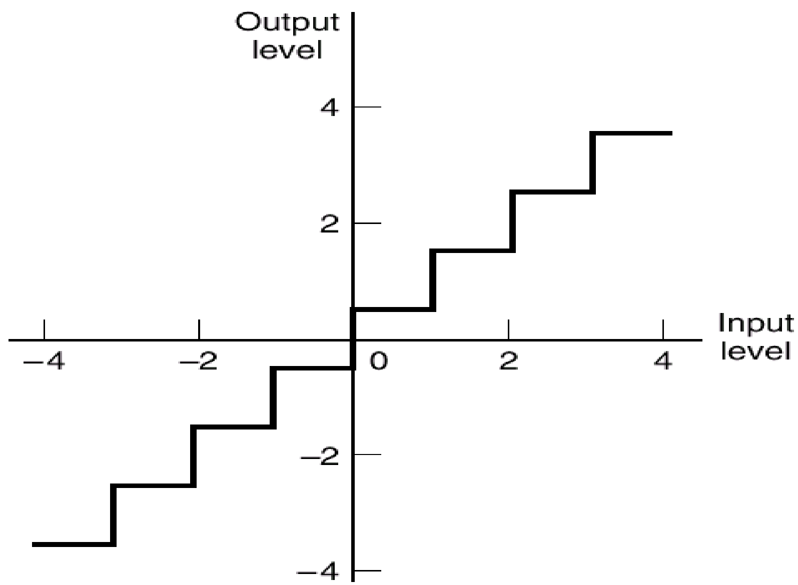
Quantizer classification:

- Uniform quantizer
- Nonuniform quantizer

- Midtread type(中平型）
- Midrise type（中升型）

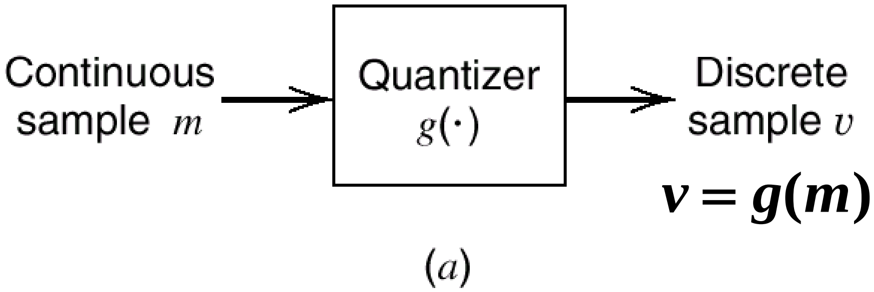


(*a*) midtread



(*b*) midrise

Quantization Noise 量化噪声



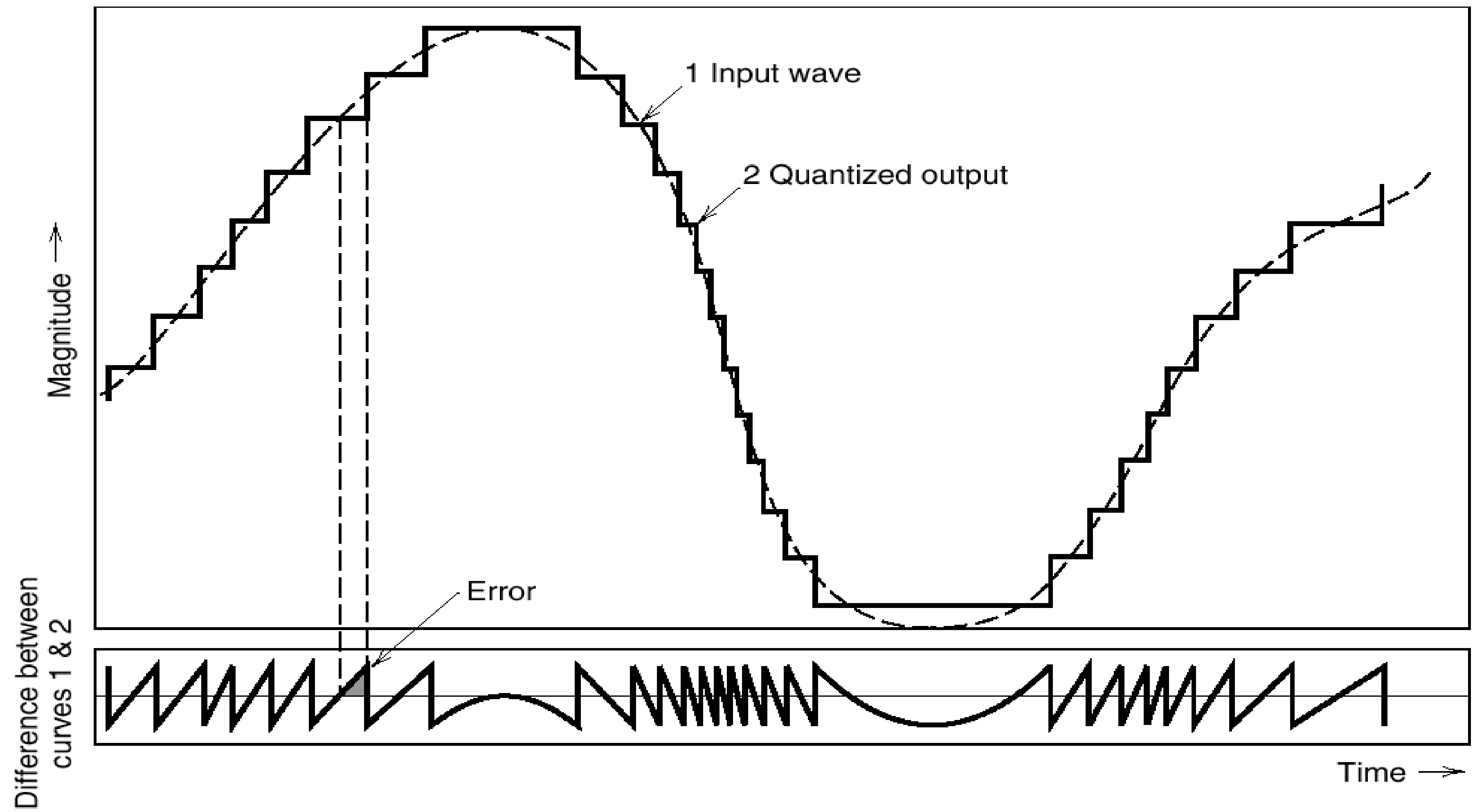
$$v = g(m)$$

$$q = m - v$$

$$Q = M - V$$

Definition: **The use of quantization introduces an error defined as the difference between the input signal m and the output signal v. The error is called quantization noise.**

Fig. 3.11 中平型均匀量化器的量化噪声示意图



How to Calculate Quantization Noise ?

Suppose

$m \in (-m_{max}, m_{max})$, uniform quantization, midrise type

Then

Step size:

$$\Delta = \frac{2m_{max}}{L}$$

$$-\frac{\Delta}{2} \le q \le \frac{\Delta}{2}$$

If Δ is sufficiently small, the quantization error Q is uniformly distributed. Thus, the probability density function of Q is

$$f_Q(q) = \begin{cases} \frac{1}{\Delta} & -\frac{\Delta}{2} < q \le \frac{\Delta}{2} \\ 0 & otherwise \end{cases} \qquad 3.26$$

Q has zero mean. So, the mean-square value equals variance.

$$\sigma_Q^2 = E[Q^2] = \int_{-\frac{\Delta}{2}}^{\frac{\Delta}{2}} q^2 f_Q(q)dq = \frac{1}{\Delta}\int_{-\frac{\Delta}{2}}^{\frac{\Delta}{2}} q^2 dq = \frac{\Delta^2}{12} \qquad 3.28$$

SNR of Uniform Quantizaer

Typically, each sample is represented by *R* binary bits, then $L=2^R$ or $R=\log_2 L$, so

$$\Delta = \frac{2m_{\max}}{2^R}$$

$$\sigma_Q^2 = \frac{1}{3} m_{\max}^2 2^{-2R}$$

P: average power of the message signal m(t), then the output SNR of a uniform quantizer:

$$(SNR)_o = \frac{P}{\sigma_Q^2} = \left(\frac{3P}{m_{\max}^2}\right) 2^{2R}$$

SNR increase exponentially with R. 指数级增长

Example 3.1 Sinusoidal Modulating Signal

Consider a full-load sinusoidal modulating wave

$$m(t) = A_m \cos 2\pi f_m t$$

$$P = \frac{A_m^2}{2}$$

By using uniform quantizer and $m_{max}=A_m$, we get

$$\sigma_Q^2 = \frac{1}{3} A_m^2 2^{-2R}$$

$$(SNR)_o = \frac{A_m^2 / 2}{A_m^2 2^{-2R} / 3} = \frac{3}{2} 2^{2R}$$

Describe SNR in decibel: 用分贝表示

$$10 \log_{10}(SNR)_o = 1.8 + 6R$$

As R increases 1 bit, SNR increases 6 dB.

例：模拟信号的数字化传输。

信号m(t)按三倍信号最高频率采样，用均匀量化器进行量化，量化后的每个样本用R比特自然二进制码表示。

求：量化信噪比，码元速率，波特率，比特率。

$$m(t) = A_m \cos 2\pi f_m t$$

Optimal Scalar Quantizer
最佳标量量化器

- ❖ Scalar quantization 标量量化
- ❖ Vector quantization 矢量量化

何谓最佳?

就是选择分层电平与重建电平，使得量化信噪比最大。选择方法与量化器输入信号的统计分布有关。当输入信号服从均匀分布时，均匀量化器是最佳量化器。

Review for Last Class

Pulse modulation types

Analog pulse modulation
脉冲模拟调制

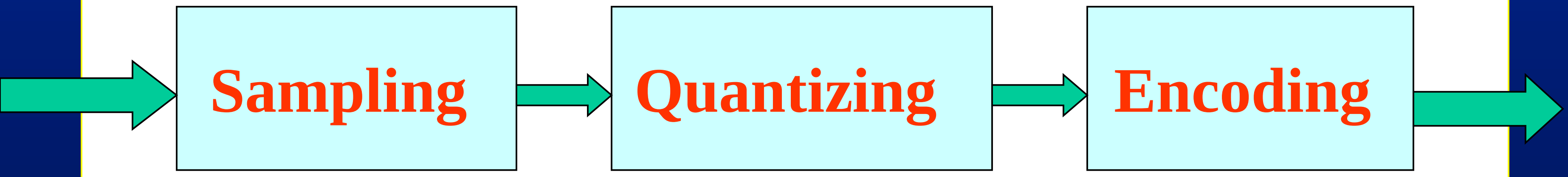
- **Pulse-amplitude modulation (PAM)**
  脉冲幅度调制
- **Pulse-duration modulation (PDM)**
  脉冲宽度（持续时间）调制
- **Pulse-position modulation (PPM)**
  脉冲位置调制

Digital pulse modulation
脉冲数字调制

- **Pulse code modulation (PCM)**
  脉冲编码调制
- **改进形式：DPCM, ADPCM, ΔM**

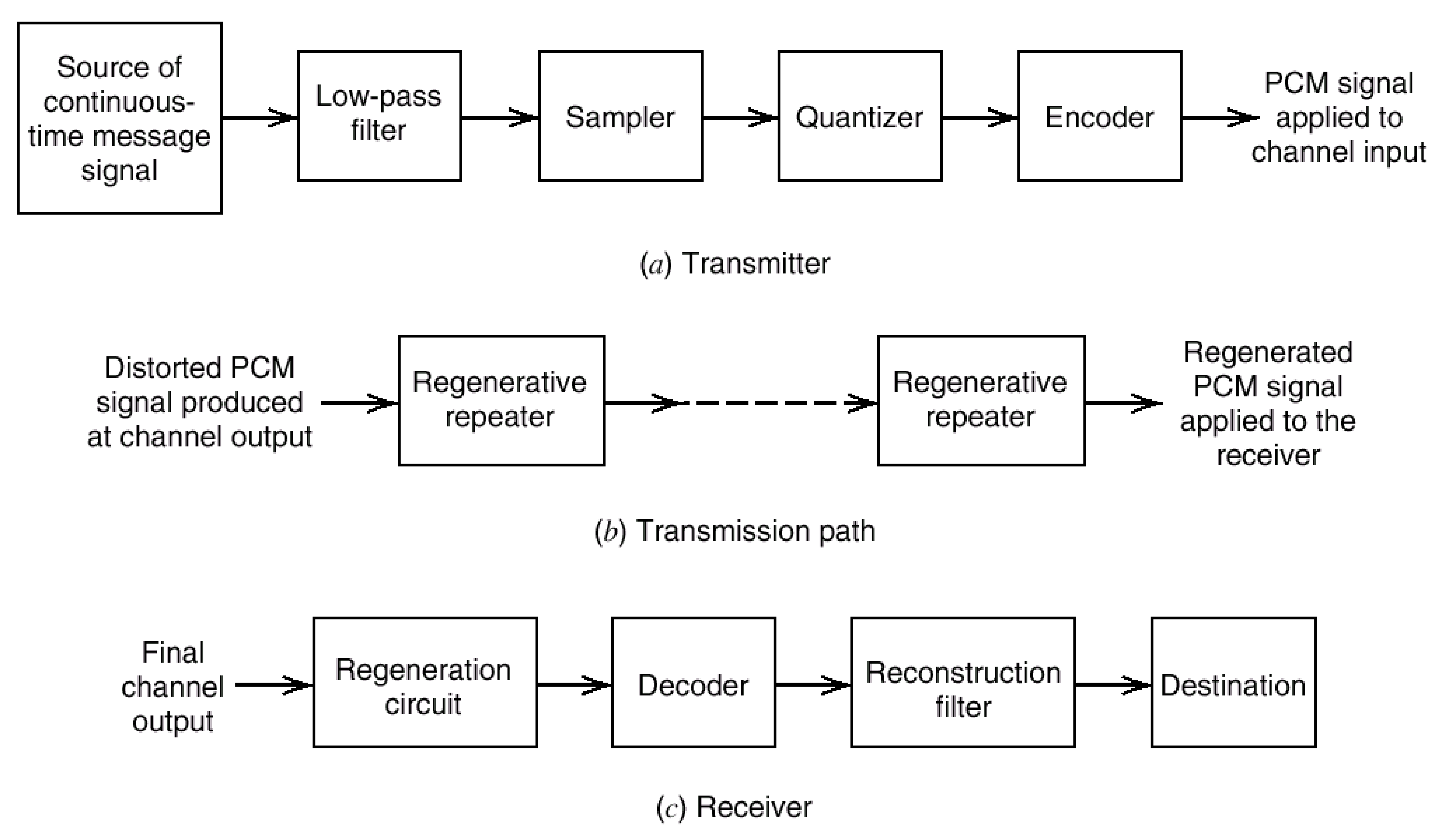
3.7 Pulse-Code Modulation
脉冲编码调制

PCM is the most basic form of digital pulse modulation. In PCM, a message signal is represented by a sequence of coded pulses, which is accomplished by representing the signal in discrete form both in time and amplitude.



Basic operations of PCM

Figure 3.13
The basic elements of a PCM system.



Sampling follows sampling theorem.

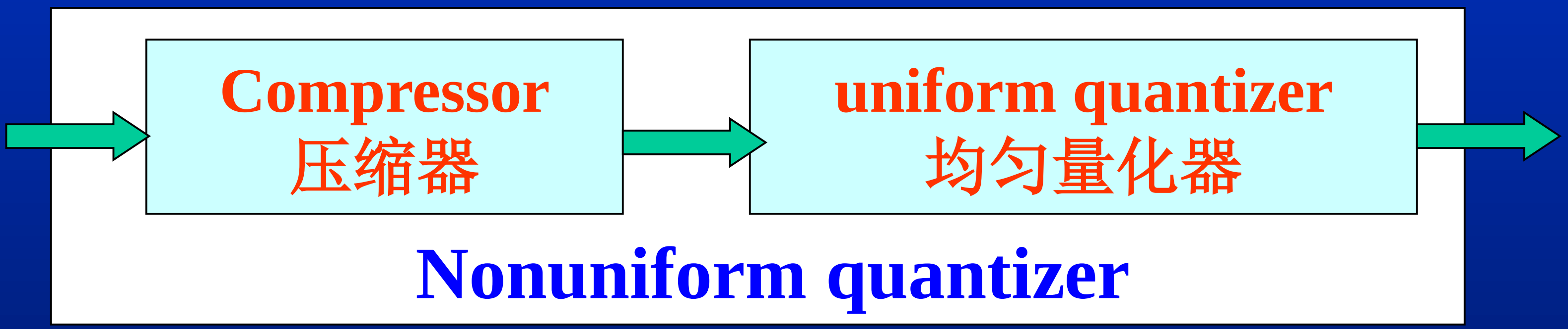
Quantization can employ uniform quantizer and nonuniform quantizer in theory.

In telephonic communication, it is preferable to use nonuniform quantizer. *Why?*

- 当输入量化器的信号具有非均匀分布的概率密度时，非均匀量化器的量化信噪比得以改善；
- 非均匀量化时，量化噪声的均方根值基本上与信号抽样值成比例，因此，量化噪声对大、小信号的影响大致相同，改善了小信号时的量化信噪比。

How to realize nonuniform quantization?

The use of nonuniform quantizer is equivalent to passing the baseband signal through a compressor and then applying the compressed signal to a uniform quantizer.



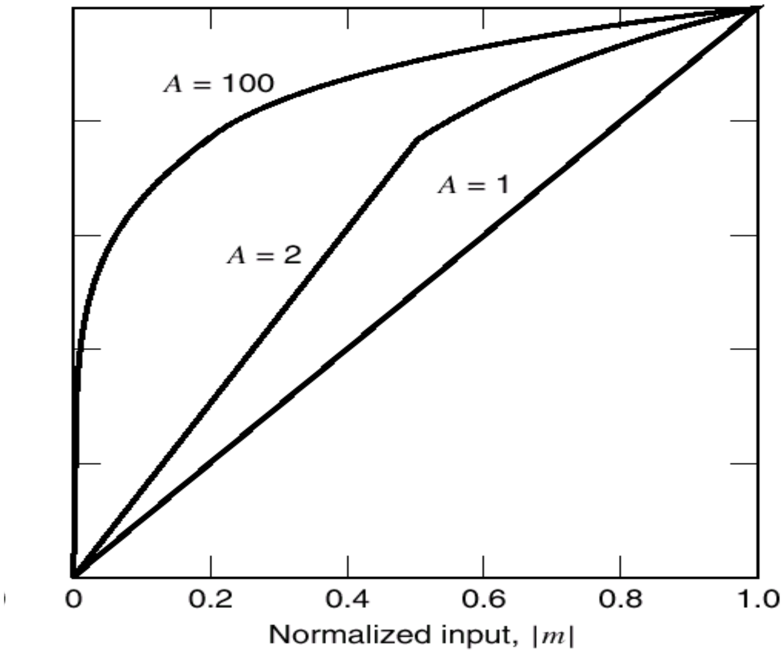
What kinds of nonuniform quantizers are used in telephonic communication ?

A-Law and μ-Law

Compression Law: A-Law

$$
|v| = \begin{cases} \dfrac{A|m|}{1+\log A} & 0 \le |m| \le \dfrac{1}{A} \\[2ex] \dfrac{1+\log(A|m|)}{1+\log A} & \dfrac{1}{A} \le |m| \le 1 \end{cases}
$$

- **A>=1**
- **A=1 uniform quantizer**
- **A=87.6 international standard**

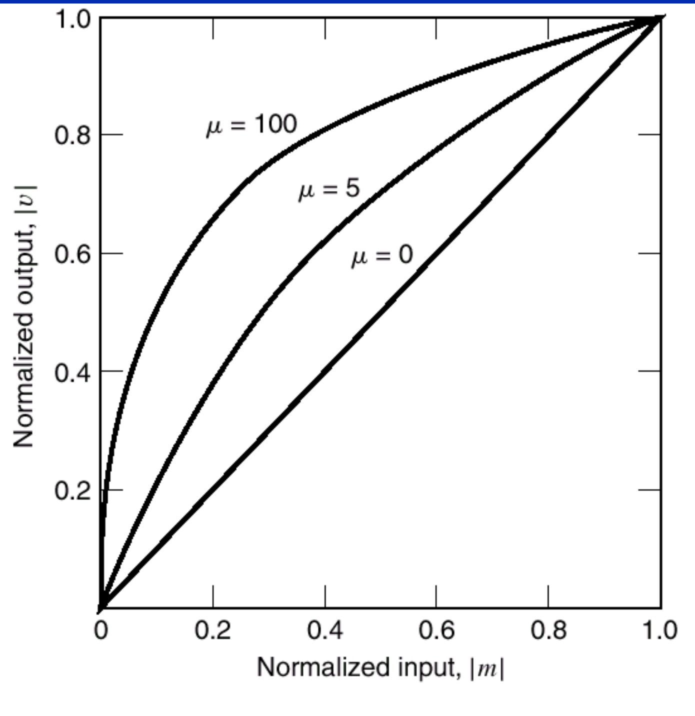


(b)

Compression Law: μ-Law

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

μ-Law is neither strictly linear nor strictly logarithmic.



(*a*)

μ>=0

μ=0 uniform quantizer

μ=255 international standard

Compressor+expander=compander
压缩器+扩张器=压扩器

There is a need to use a device in the receiver with a characteristic complementary to the compressor. Such a device is called an expander. Ideally, the compression and expansion laws are exactly inverse so that the expander output is equal to the compressor input.

理想情况下，压缩与扩张互为逆运算．扩张器是压缩器的逆系统．

非均匀量化的实现方法有两种：

（1）模拟压扩法

（2）数字压扩法。数字压扩法就是用数字电路产生许多相连接的折线段来近似对数压扩特性，直接对抽样信号进行非均匀量化编码。

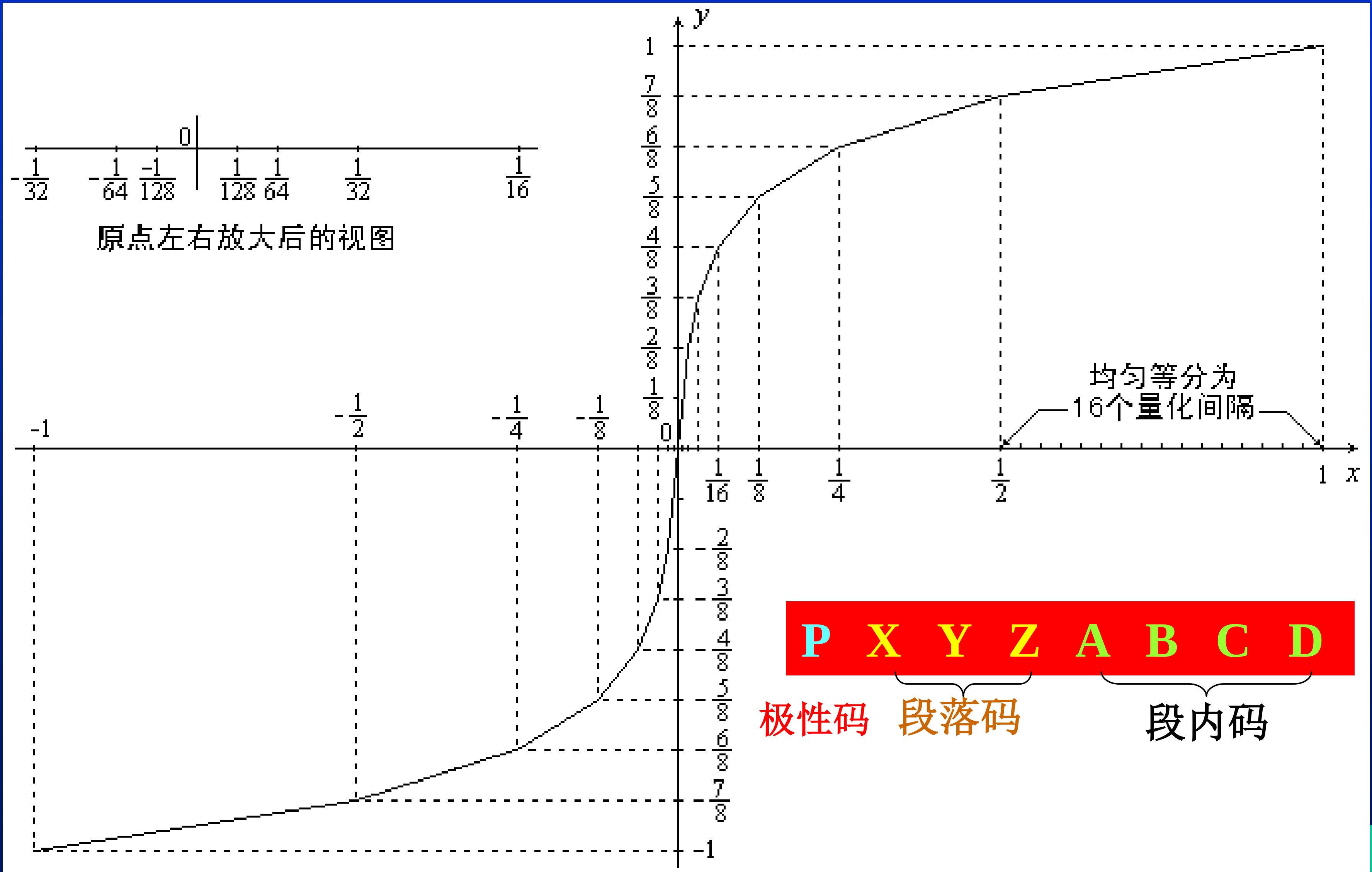
Piecewise linear approximation
分段线性近似

- It is also of interest to note that in actual PCM systems, the companding circuitry does not produce an exact replica of the nonlinear compression curves.
- Rather, it provides a piecewise linear approximation to the desired curve.

A律13折线近似

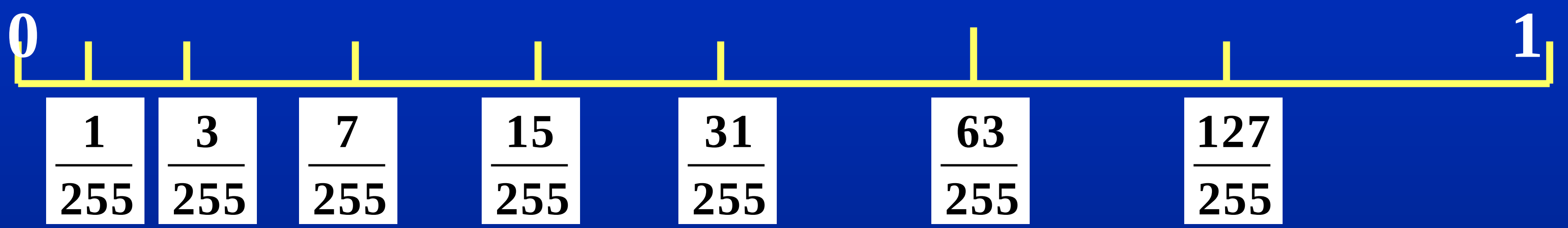
μ律15折线近似

Why are 13 linear segments used to approximate the logarithmic A-Law with the constant A=87.6?



Why are 15 linear segments used to approximate the logarithmic μ-Law with the constant μ=255?

(0,1)范围内非均匀地分成8段



(-1,0)范围内也对称地非均匀地分成8段

正负区间靠0的两段斜率相同,合并为一条折线段,故共有15条折线. 这15条折线组成的曲线与μ=255的μ律对数量化特性很接近, 故称为μ律15折线近似.

Encoding

Why encoding?

- After sampling and quantizing, the specification of a continuous message signal becomes limited to a discrete set of values, but not in the form best suited to transmission over a telephone line or radio path.

- We require the use of an encoding process to translate the discrete set of sample values to a more appropriate form of signal.

Terms about encoding

Any plan for representing each of the discrete set of values as a particular arrangement of discrete events is called **a code**.

One of the discrete events in a code is called **a code element码元 or symbol符号.**

A particular arrangement of symbols used in a code to represent a single value is called **code word码字 or character字符.**

Binary code: 1 and 0

Ternary code: 0, 1, 2.

00	0
01	1
10	2
11	3

量化电平	自然二进码	格雷码	折叠二进码
15	1111	1000	1111
14	1110	1001	1110
13	1101	1011	1101
12	1100	1010	1100
11	1011	1110	1011
10	1010	1111	1010
9	1001	1101	1001
8	1000	1100	1000
7	0111	0100	0000
6	0110	0101	0001
5	0101	0111	0010
4	0100	0110	0011
3	0011	0010	0100
2	0010	0011	0101
1	0001	0001	0110
0	0000	0000	0111

（1）常用二进制代码：

格雷码（Gray Code）：

$$\begin{cases} g_{n-1} = b_{n-1} \\ g_i = b_{i+1} \oplus b_i \qquad i = 0, 1, 2, \cdots, n-2 \end{cases}$$

$$\begin{cases} b_{n-1} = g_{n-1} \\ b_i = b_{i+1} \oplus g_i \qquad i = 0, 1, 2, \cdots, n-2 \end{cases}$$

折叠二进制码（Folded Binary Code）：

$$\begin{cases} f_{n-1} = b_{n-1} \\ f_i = \overline{b_{n-1}} \oplus b_i \qquad i = 0, 1, 2, \cdots, n-2 \end{cases}$$

$$\begin{cases} b_{n-1} = f_{n-1} \\ b_i = \overline{f_{n-1}} \oplus f_i \qquad i = 0, 1, 2, \cdots, n-2 \end{cases}$$

例：模拟信号的数字化传输。

某模拟信号m(t)最高频率为10kHz，按奎斯特采样率进行采样。

1）若采用12比特的均匀量化与线性编码，求码元速率；

2）若采用8比特A律对数PCM码，求码元速率；

3）若采用8比特的格雷码，求码元速率。

Line codes 线路码

数字基带信号是数字信息的电脉冲表示，不同形式的数字基带信号或码型具有不同的频谱结构，合理地设计数字基带信号以使数字信息变换为适合于给定信道传输特性的频谱结构，是基带传输首先要考虑的问题。

把数字信息的电脉冲表示过程称为**码型变换**。

在有线信道中传输的数字基带信号又称为**线路传输码**。

线路传输码设计原则：

① 对于传输频带低端受限的信道，一般来说线路传输码型的频谱中应不含直流分量。

② 尽量减少基带信号频谱中的高频分量，以便节省传输频带和减小串扰。

③ 信号的抗噪声能力强，波型间相关性越小越好。产生误码时，在译码中不产生误码的扩散或误差的增值，如果有，也希望越小越好。

④ 便于从信号中提取定时信息；若采用分组形式传输时，不但要从基带信号中提取位定时信息，而且要便于提取分组同步信息。

⑤ 要求基带传输信号具有内在的检错能力。

⑥ 编译码的设备应尽量简单。

Commonly used Line codes 常用线路码

1. Unipolar nonreturn-to-zero (NRZ) signaling
   单极性非归零码 on-off signaling 通-断信号
2. Polar nonreturn-to-zero (NRZ) signaling
   双极性非归零码
3. Unipolar return-to-zero (RZ) signaling
   单极性归零码
4. Bipolar return-to-zero (BRZ) signaling
   双极性归零码
   alternate mark inverse (AMI) signaling
   交替极性反转码
5. Split-phase(Manchester code)
   多相码或曼彻斯特码

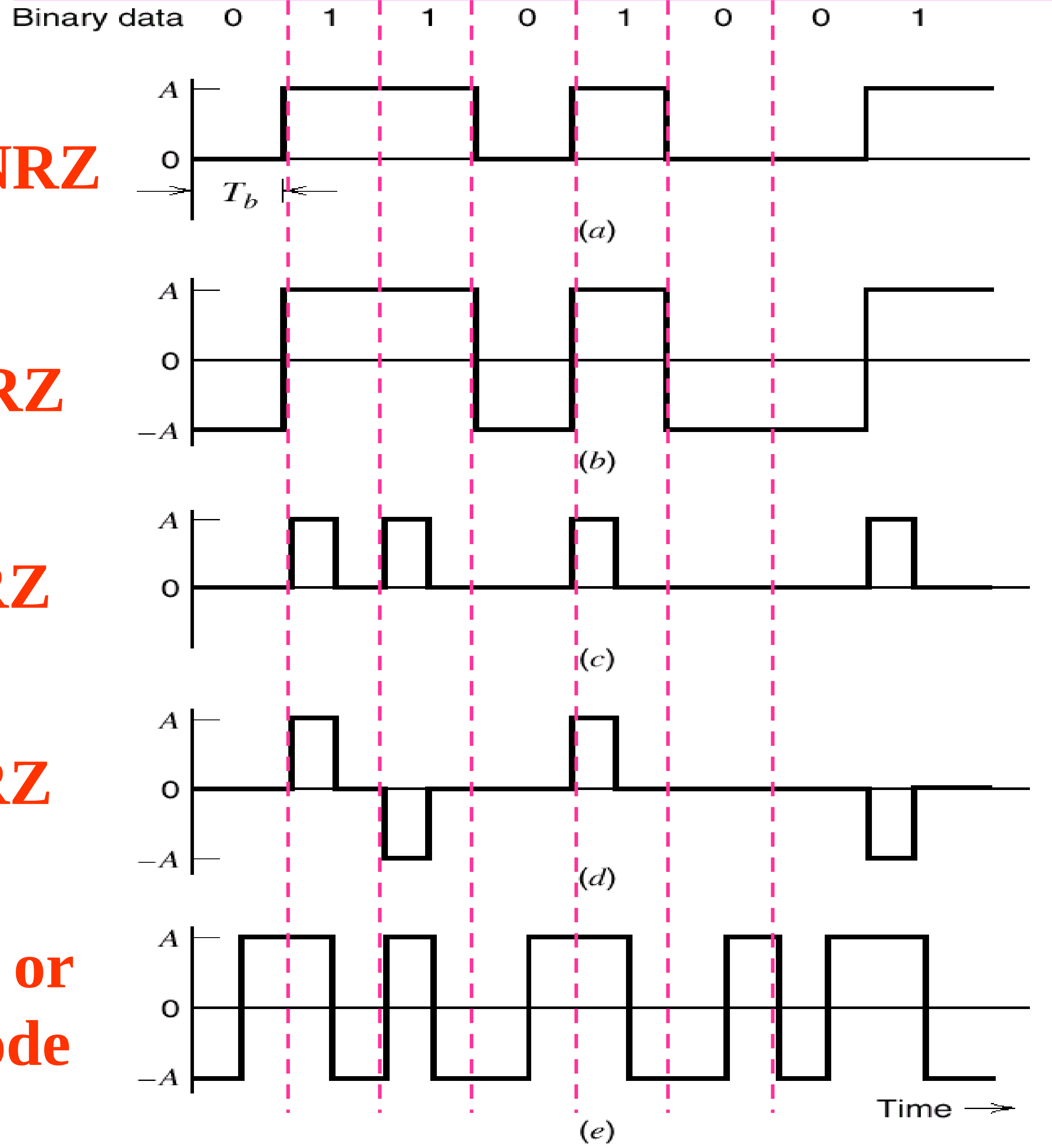
(a) Unipolar NRZ

(b) Bipolar NRZ

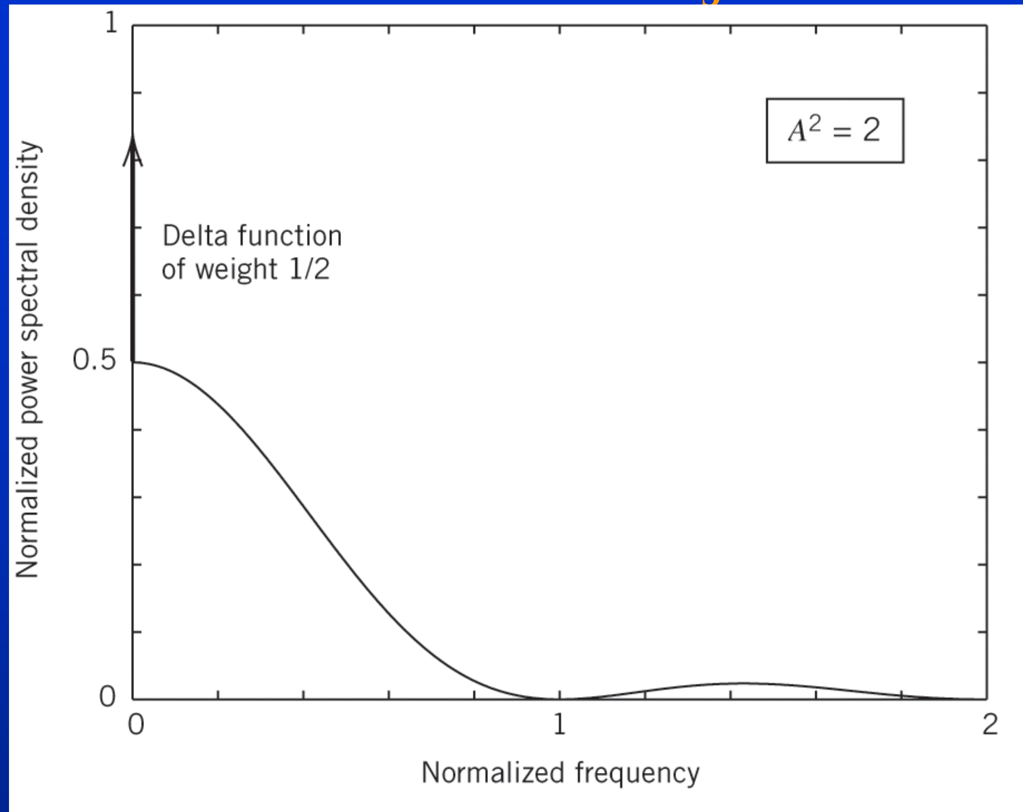
(c) Unipolar RZ

(d) Bipolar NRZ AMI

(e) split-phase or manchester code



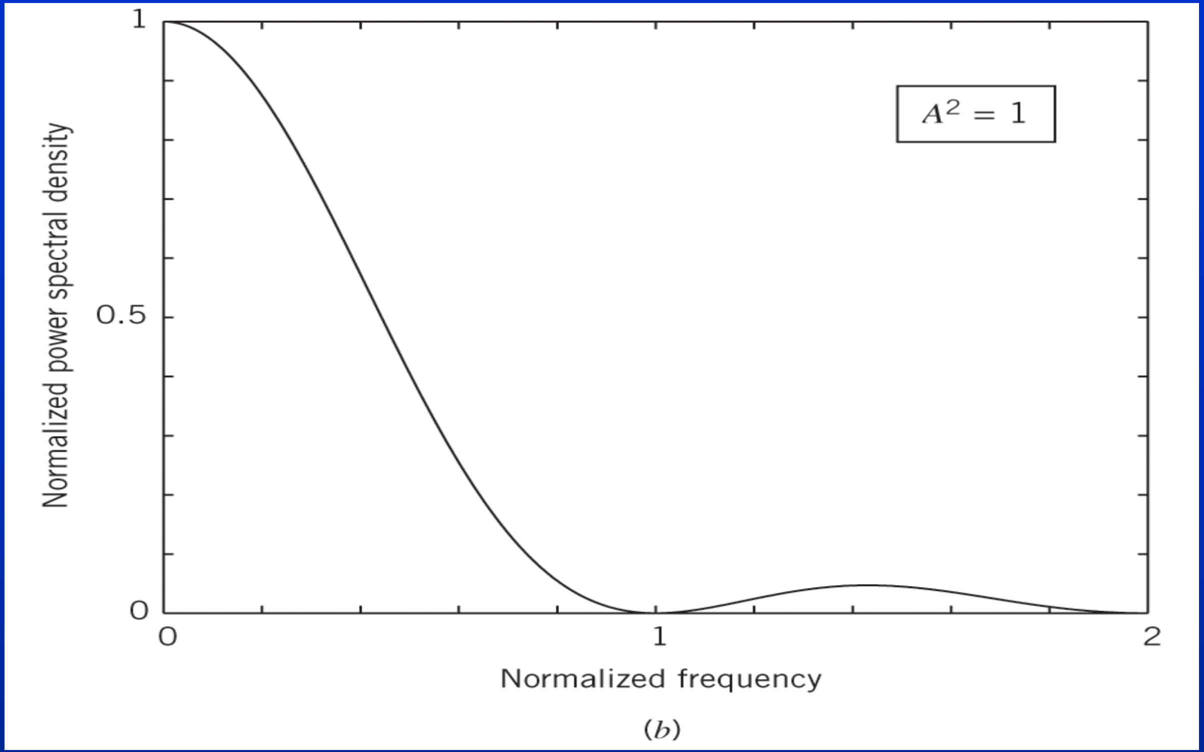
Unipolar NRZ



(*a*)

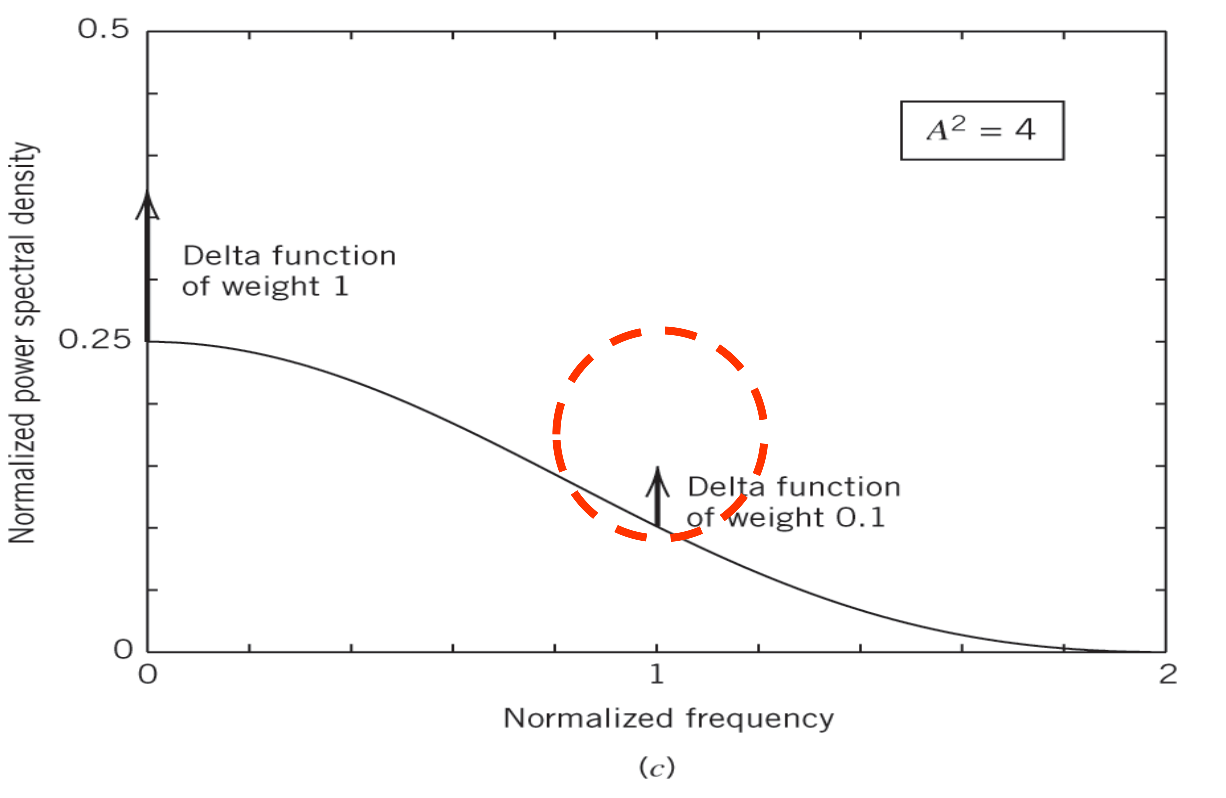
- **Advantage**: **Simplicity**
- **Disadvantage: Waste of power due to DC level and large spectrum value at zero frequency.**

Polar NRZ



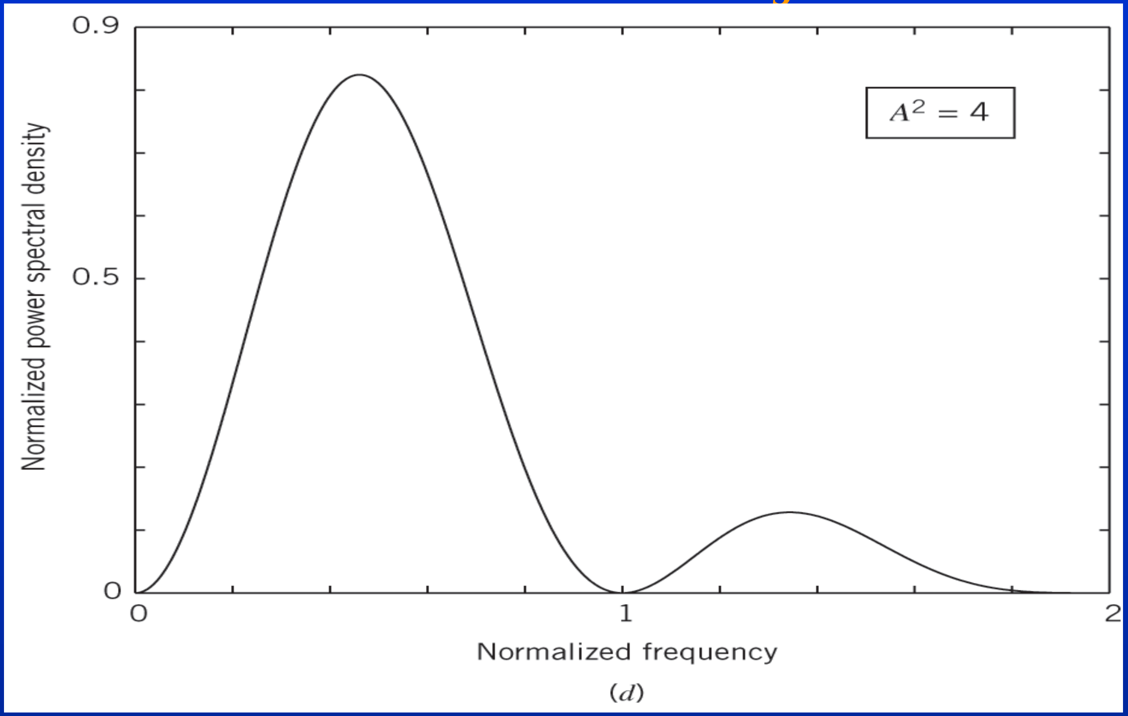
- **Advantage:** Simplicity, no DC level if "1" and "0" are equiprobable.
- **Disadvantage:** Waste of power due to large spectrum value at zero frequency.

Unipolar RZ



- **Advantage:** delta function at f=0, ±1/Tb in the power spectrum can be used for bit-timing recovery. 有利于位定时的提取。
- **Disadvantage:** Waste of power

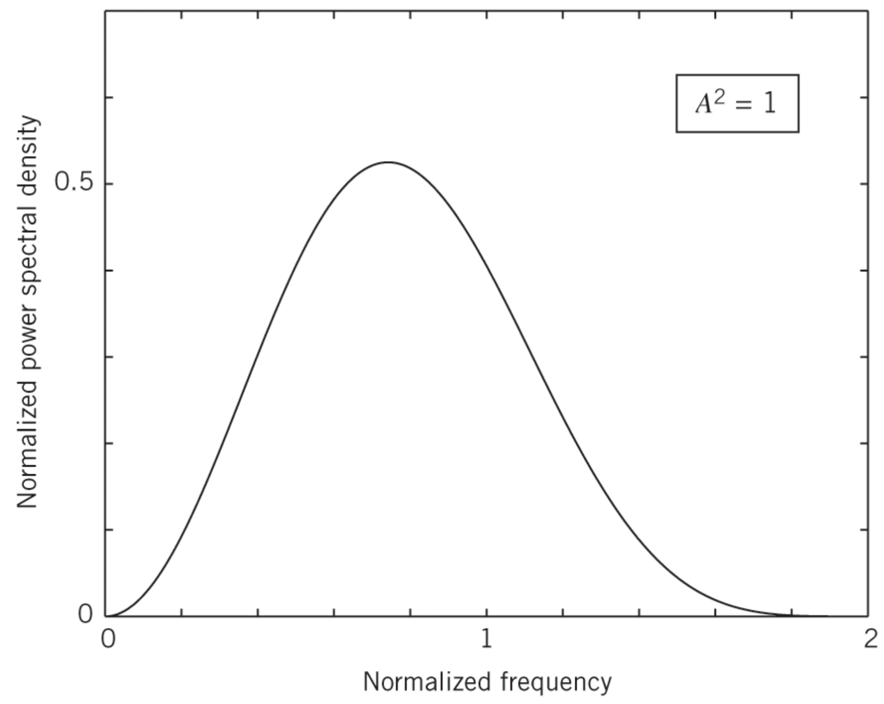
BRZ/AMI



- **Advantage:** **power spectrum has no DC component and relatively insignificant low – frequency components when symbols "1" and "0" occur with equal probability.**
- **Disadvantage:** **complexity.**

Manchester code

Split-phase



(e)

- **Advantage:** power spectrum has no DC component and relatively insignificant low – frequency components regardless of the signal statistics.

- **Disadvantage:** complexity.

差分码 Differential Encoding

差分码又称为相对码，它不是以信号电平的大小取值，而是以电平跳变和不变表示数字信息，分为两种：

- 若以电平跳变表示“1”，不变表示“0”，则称为传号差分码或“1”差分码；（电报通信中常把“1”称为传号，“0”称为空号），
- 若用电平跳变表示“0”，不变表示“1”，则称为空号差分码或“0”差分码。

English Textbook

Differential encoding is defined to encode information in terms of signal transitions（信号跃变）

- **A transition is used to designate symbol 0;**
- **No transition is used to designate symbol 1.**

“1”差分码？“0”差分码？ ----“0”差分码!!!

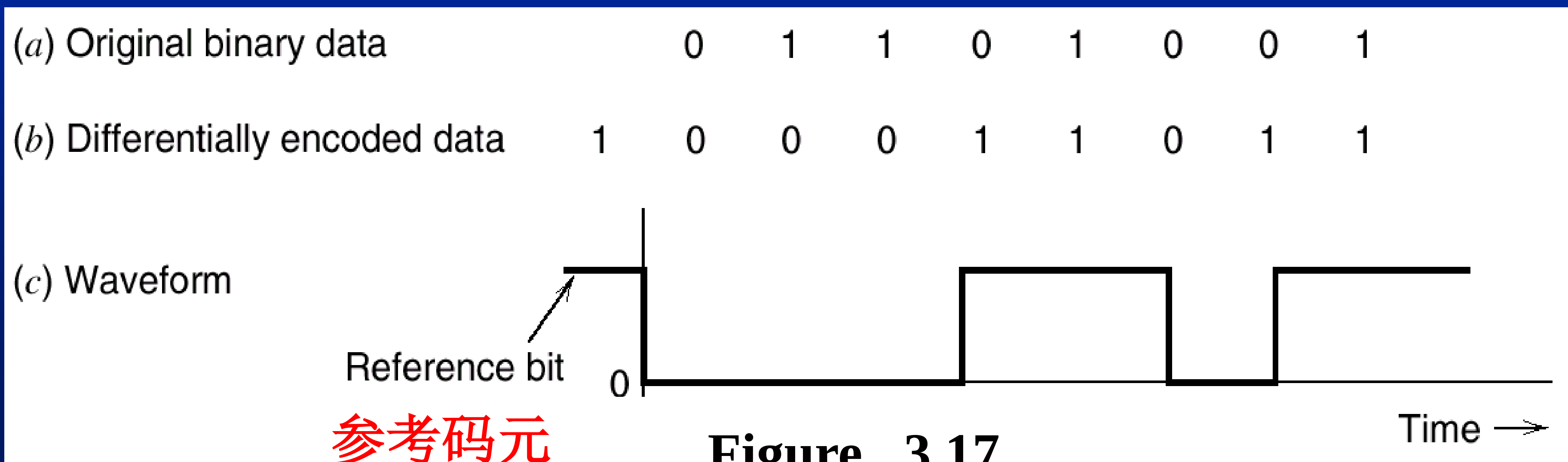
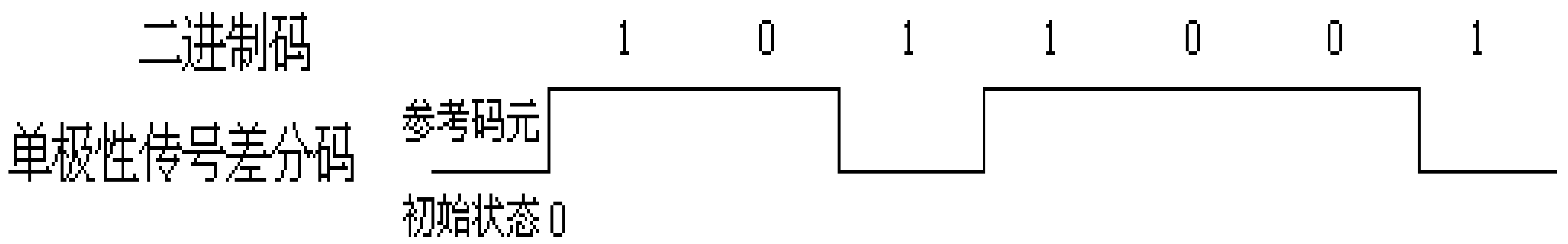


Figure 3.17



"1"差分编码规则：$d_k = a_k \oplus d_{k-1}$

译码规则：$a_k = d_k \oplus d_{k-1}$

绝对码a_k：1 0 1 1 0 0 1

"1"差分码d_k：0* 1 1 0 1 1 1 0

思考题： "0"差分的编码和译码规则是什么？

若只给出绝对码，但不规定初始状态和"1"差分还是"0"差分，问对应的差分码有几种可能波形？

例：

绝对码a_k： 1 1 0 1 1 0 1

“1”差分码d_k： 0*

“0”差分码d_k： 0*

“1”差分码d_k： 1*

“0”差分码d_k： 1*

例:

"1"差分码d_k: 0* 1 0 0 1 0 0 1

绝对码a_k:

"0"差分码d_k: 0* 0 0 1 1 1 0 0

绝对码a_k:

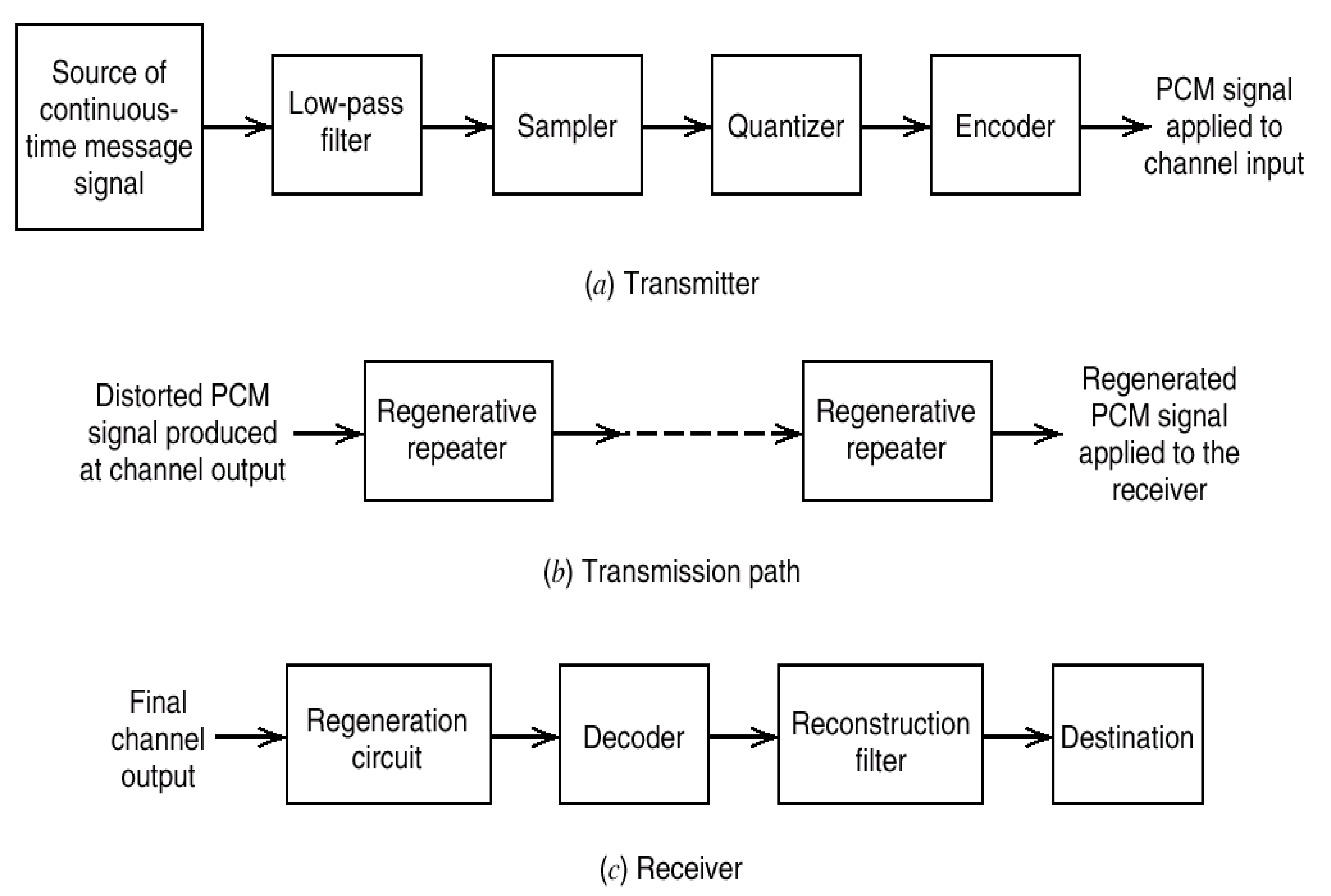
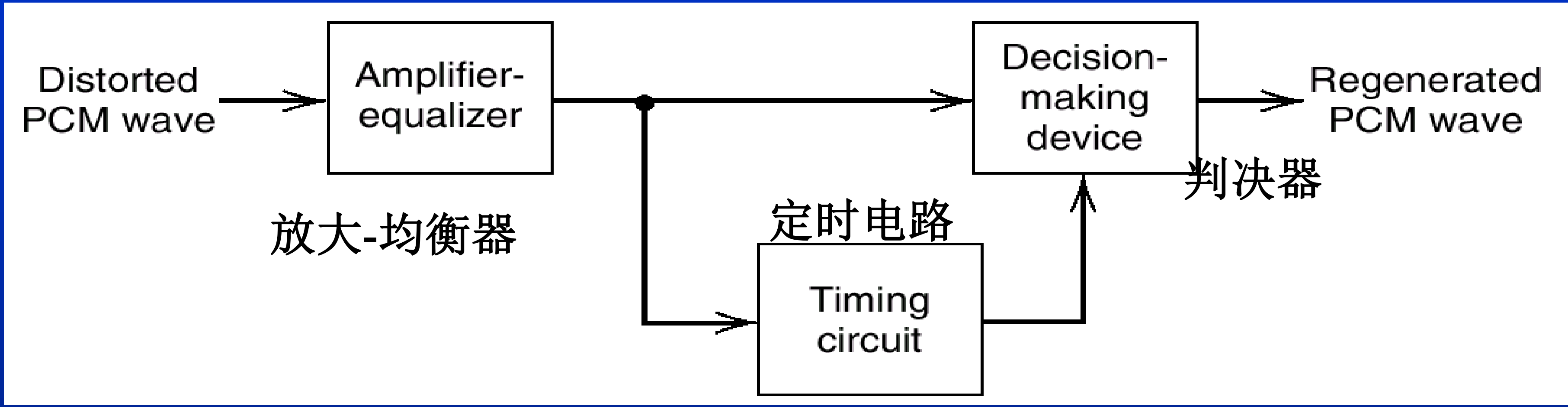


Figure 3.13 The basic elements of a PCM system.

Regeneration 再生

Figure 3.18 Block diagram of regenerative repeater



Repeaters （再生中继器） have three basic functions ：

- **Equalization 均衡**
- **timing 定时**
- **decision making 判决**

3.8 Noise Consideration in PCM Systems

PCM systems have two major sources of noise:

1 Channel noise

It is introduced anywhere between the transmitter output and receiver input. Channel noise is always present, once the equipment is switched on.

2 Quantization noise

It is introduced in the transmitter and is carried all the way along to the receiver output. Unlike channel noise, quantization noise is signal-dependent in the sense that it disappears when the message signal is switched off.

- **Naturally, these two sources of noise appear simultaneously once the PCM system is in operation.**
- **However, the traditional practice is to consider them separately, so that we may develop insight into their individual effects on the system performance.**

Bit Error Rate (BER) 误比特率

Average probability of symbol error （平均符号差错概率，误符号率）

It is defined as the probability that the reconstructed symbol at the receiver output differs from the transmitted symbol.

Bit Error Rate 误比特率

It is defined as the probability that the reconstructed bit at the receiver output differs from the transmitted bit.

二进制系统中，误比特率=误符号率

3.9 Time-Division Multiplexing TDM

Multiplexing: a number of independent signals are combined into a composite signal suitable for transmission over a common channel.

FDM:
separating the signals according to frequency

TDM:
separating the signals according to time

How to understand the principle of TDM?

An important feature of the sampling process is a conservation of time 时间守恒.

That is, the transmission of message samples engages the communication channel for only a fraction of the sampling interval on a periodic basis, and in this way some of the time interval between adjacent samples is cleared for use by other independent message sources on a time-shared basis 时分方式共享.

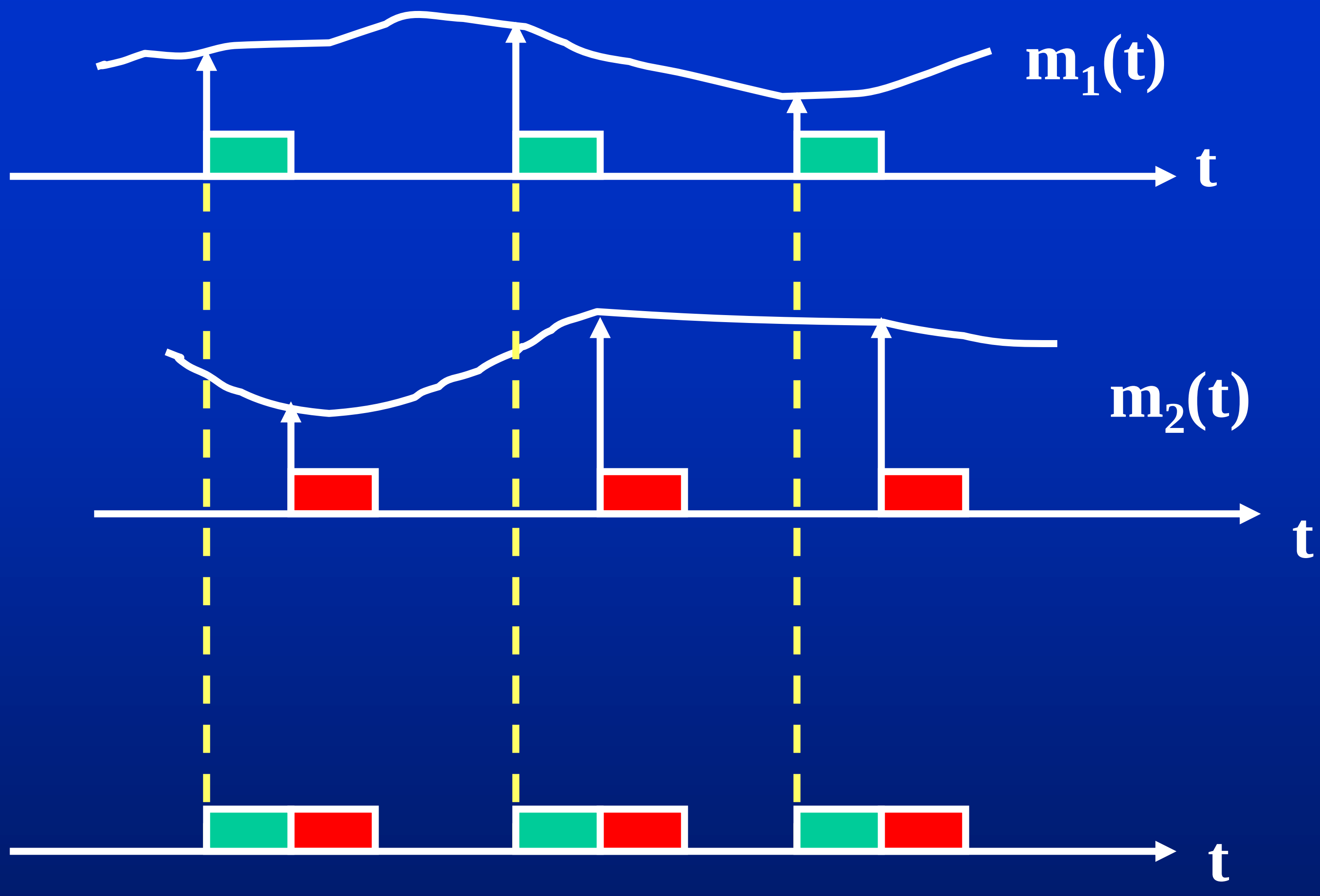
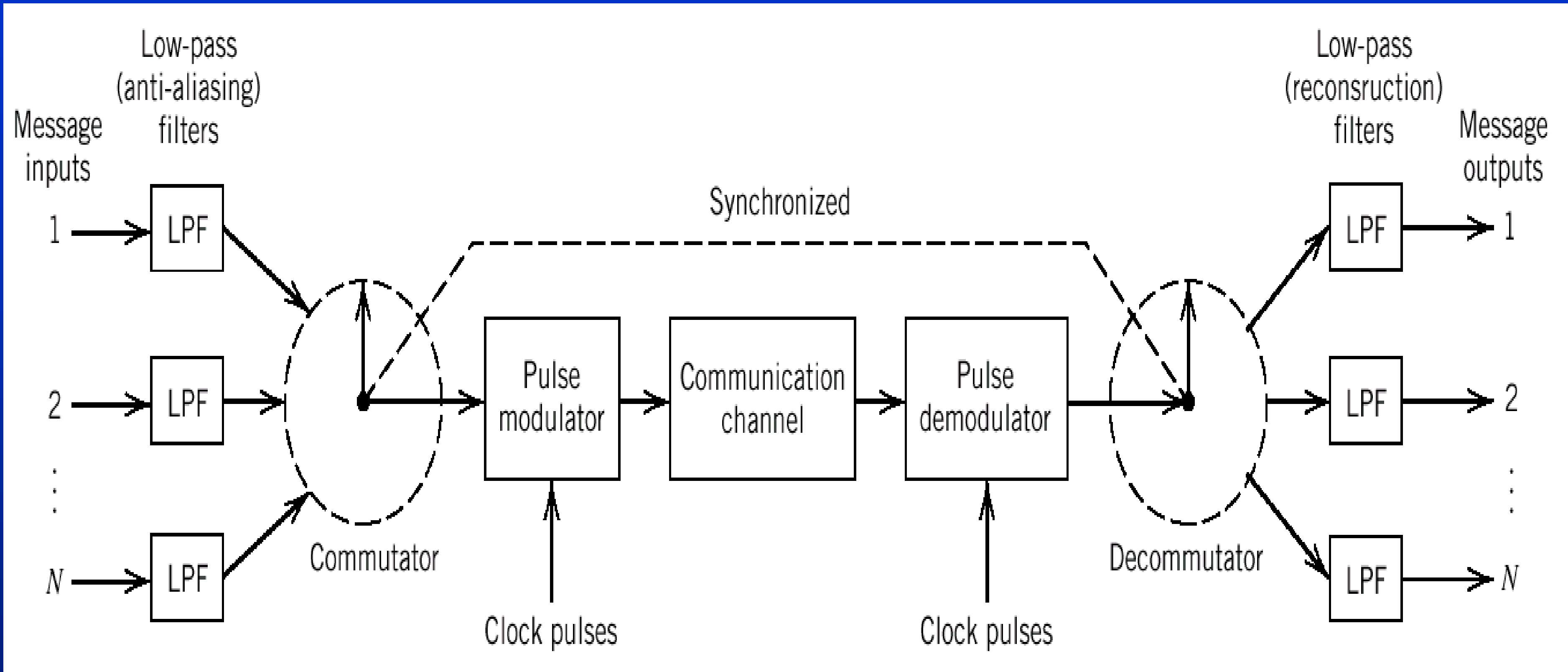

$m_1(t)$
t
$m_2(t)$
t
t

Figure 3.19 Block diagram of TDM system.



Commutator合路器 **Decommutator 分路器**

Pulse modulator 脉冲调制器 **Pulse demodulator**

A commutator, which is usually implemented using electronic switching circuitry 电子开关电路. 完成采样与按时间顺序分配各路信号.

A pulse modulator, the purpose of which is to transform the multiplexed signal into a form suitable for transmission over a common channel.

Synchronization 同步

In a PCM system with time-division multiplexing, there is a need that a local clock at the the receiver keeps the same time as a distant standard clock at the transmitter.

- Bit synchronization 位同步
- Code word synchronization 字同步，路同步
- Frame synchronization 帧同步
- Carrier synchronization 载波同步
- Net synchronization 网同步

Typical application of TDM:
数字话音TDM–PCM系统

话音信号的频带限制在300~3100Hz范围内，根据CCITT建议，采用8kHz的抽样率，抽样周期为125μs，每样值采用8位二进制非线性编码。由于国际上通用的PCM有A律和μ律之分，它们的编码规则不同，所以时分多路复用的基群帧结构不同，形成了:

A律TDM-PCM30/32制式 E1 system
μ律TDM-PCM24制式 T1 system

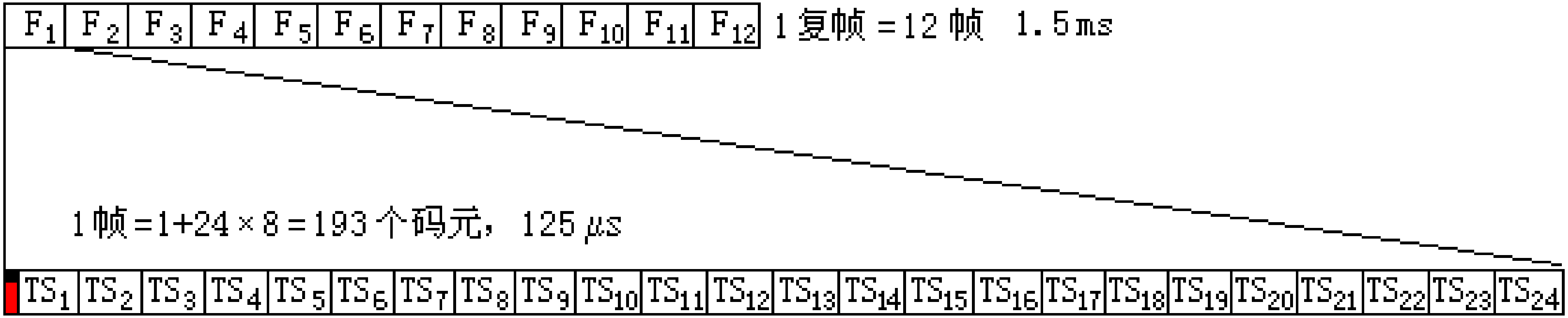
Example 3.2 the T1 System

- T1 system carries 24 voice channels over separate pairs of wires
- T1 system is a typical example of TDM.
- T1 system is basic to the North American Digital Switching Hierarchy.
- T1 system uses a piecewise-linear characteristic (consisting 15 linear segments) to approximate the logarithmic μ-Law with the constant μ=255. (μ律15折线近似）

What is E1 system?

- E1 system carries 30 voice channels over separate pairs of wires.
- E1 system is a typical example of TDM.
- E1 system is used in China and Europe.
- E1 system uses a piecewise-linear characteristic (consisting 13 linear segments) to approximate the logarithmic A-Law with the constant A=87. 56. (A律13折线近似)

T1 system—Frame Structure



在μ律TDM-PCM24制式中，一个抽样周期的125μs被分成193个码元，组成一帧。

12帧构成一个复帧，复帧周期为1.5ms。

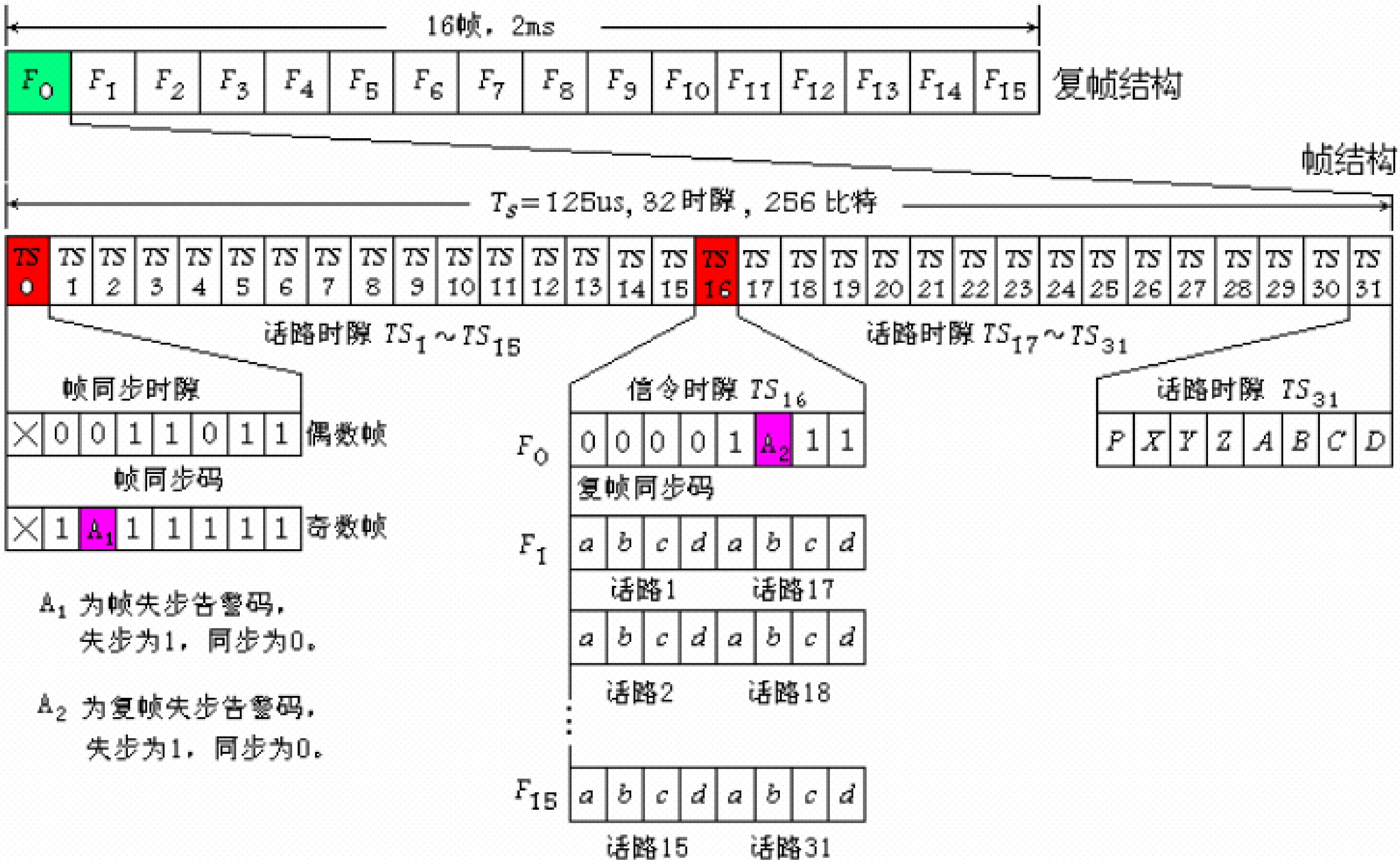
在PCM24系统中，总的数码率为（8×24＋1）×8000＝1544 kbit/s。

Frame Synchronization 帧同步

每帧193个码元中帧首编号为1的位交替传送帧同步码和复帧同步码。其中12帧中的奇数帧的第1位码元构成“101010”帧同步码组，而偶数帧的第1位码元构成复帧同步码“00111”，第12帧的第1位码用作对端告警用。

每帧中其余192位码元每8位构成一路时隙，用于传送24路电话信号。

A律TDM-PCM30/32制式基群帧结构：



在抽样率为8000Hz时，PCM30/32系统的数码率为 f_b=8×32×8000=2048 Kb/s

复帧：一帧中的TS16只有8位码，不足以传送30个话路的标志信号，所以必须将16帧构成一个更大的帧，称为复帧。复帧的重复频率为8000÷16＝500Hz，周期为125×16＝2.0ms。

6. 频分多路复用信号在（ ）。

. 1 .

① 频谱上可分辨，时间上混叠在一起；
② 频谱上可分辨，时间上可分辨；
③ 时间上可分辨，频谱上混叠在一起；
④ 时间上可分辨，频谱上可分辨。

12. PCM30/32 路电话系统在一个取样周期内(即一帧内)传送（ ）个码元。

① 1.544M； ② 2.048M； ③ 256； ④ 193。

思考题：一时分复用传输系统包括三路信源信号和一路帧同步信号。三路信源信号的最高频率分别为1 kHz，2kHz，4kHz。请设计该时分复用系统的帧结构。如果每个采样样本采用8位线性二进制编码，则数据速率是多少？

方案1：按统一的采样率**8kHz**，帧长**125μs**，数据速率？

1	**2**	**3**	**4**	**1**	**2**	**3**	**4**

方案2：各自按奈奎斯特采样率，帧长？数据速率？

500 μs

$f_{s1} = 2\ kHz$	$T_{s1} = 500\ \mu s$	1次
$f_{s2} = 4\ kHz$	$T_{s1} = 250\ \mu s$	2次
$f_{s3} = 8\ kHz$	$T_{s1} = 125\ \mu s$	4次

再加上一路同步信号，一共八个时隙

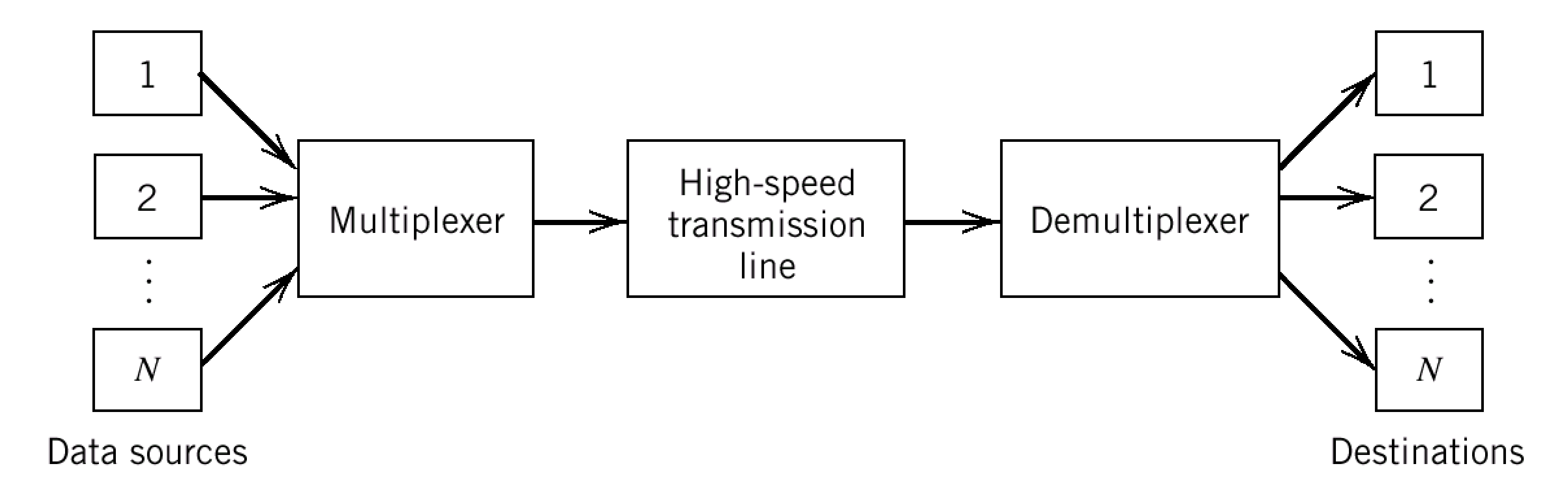
同步	3	2	3	1	3	2	3

500 μ

3.10 Digital Multiplexers 数字复接器

- In this section, we consider the multiplexing of digital signals at different bit rates. The device used to accomplish this operation is referred to as digital multiplexers.

Fig. 3.20 a conceptual diagram of the digital multiplexing-demultiplexing operation



The multiplexing of digital signals is accomplished by using a bit-by-bit interleaving 按位交织 procedure with a selector switch that sequentially takes a bit from each incoming line and then applies it to the high-speed common line.

PDH与SDH数字复接系列

	制式 / 群路等级	PCM30/32		PCM24	
		数码率 Mb/s	话路数	数码率	话路数
PDH准同步数字复接系列	基群	2.048	30	1.544	24
	二次群	8.448	30 ×4=120	6.312	24×4=96
	三次群	34.368	120 ×4=480	44.736	96 ×7=672
	四次群	139.264	480 ×4=1920	274.176	672 ×6=4032
SDH同步数字复接系列	STM-1	155.520Mb/s			
	STM-4	622.080Mb/s			
	STM-16	2.5Gb/s（2488.320Mb/s）			
	STM-64	10Gb/s（9953.28）			
	STM-256	40Gb/s（39813.12）			

3.11 Virtues, Limitations, and Modifications of PCM

Advantages of PCM

1 **Robustness** to channel noise and interference

2 **Efficient regeneration**

3 **Efficient exchange** of B_T for improved SNR

4 **Uniform format** for the transmission of different kinds of baseband signals

5 Comparative **ease** to drop or reinsert message sources in TDM system

6 **Secure communication** through the use of special modulation schemes or encryption.

Disadvantages of PCM

1. Increased system complexity

2. Increased channel bandwidth

However, today these two problems are no longer serious. Why?

For problem 1----complexity

----Very-large-scale integrated (VLSI) circuit
----delta modulation 增量调制 can be used if the simplicity of implementation is a necessary requirement.

For problem 2----bandwidth

----Wideband communication channels, such as satellites and optic fibers, are available.
----Data compression is helpful to reduce bandwidth requirement.

信源编码的目的与任务

目的：提高通信的有效性

任务：模拟信号数字化

数据压缩

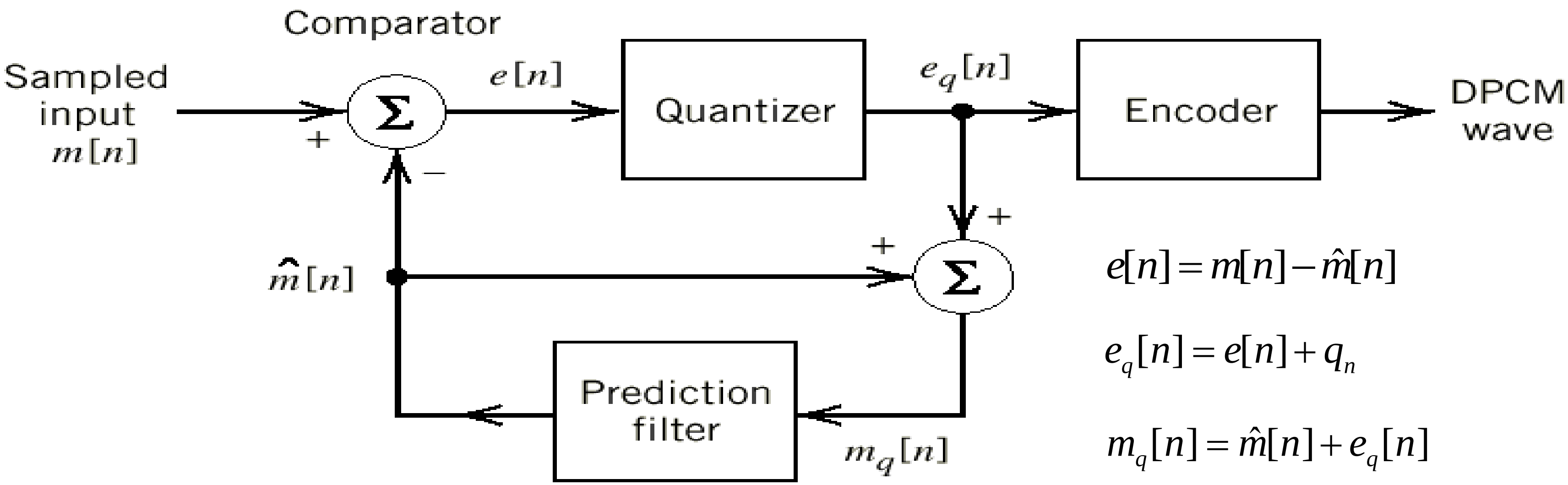
3.14 Differential Pulse-code Modualtion 差分脉冲编码调制(DPCM)

PCM samples have following features:

1. High degree of correlation between adjacent samples 高度相关
2. Redundancy information 冗余信息

When the highly correlated samples are encoded, the resulting encoded signal contains redundant information. By removing this redundancy before encoding, we obtain a more efficient coded signal, which is the basic idea behind DPCM.

Fig. 3.28 DPCM system.

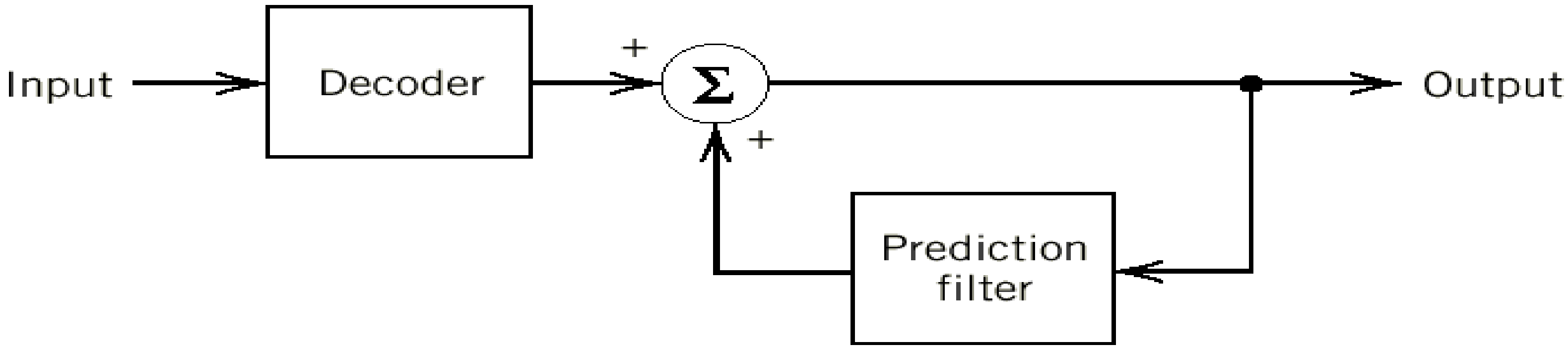


$$e[n] = m[n] - \hat{m}[n]$$

$$e_q[n] = e[n] + q_n$$

$$m_q[n] = \hat{m}[n] + e_q[n]$$

(*a*) Transmitter.



(*b*) Receiver

Virtues of DPCM

- **DPCM can be used to improve noise performance or reduce bit rate.**
- **The latter function is referred to as data compression.**

3.15 Adaptive Differential Pulse-code Modulation ADPCM 自适应差分脉冲编码调制

ADPCM的主要改进是量化器和预测器均采用自适应方式，使量化器和预测器的参数能随输入信号的统计特性自适应于最佳或接近于最佳参数状态。

3.13 linear Prediction
线性预测

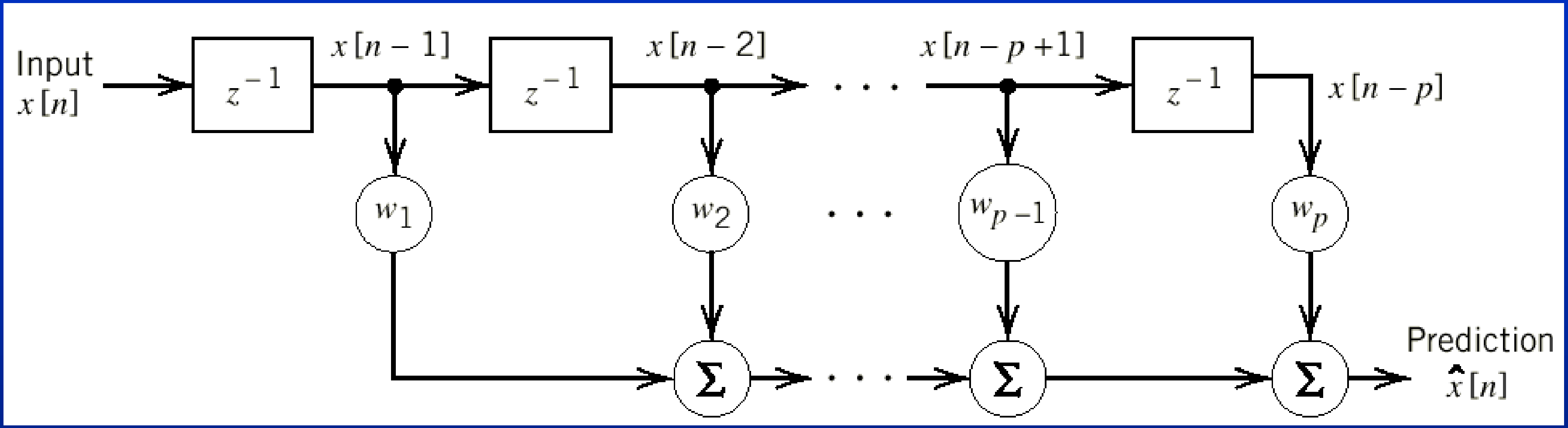


Figure 3.26

Block diagram of a linear prediction filter of order *p*.

$$\hat{x}[n] = \sum_{k=1}^{p} w_k x[n-k]$$

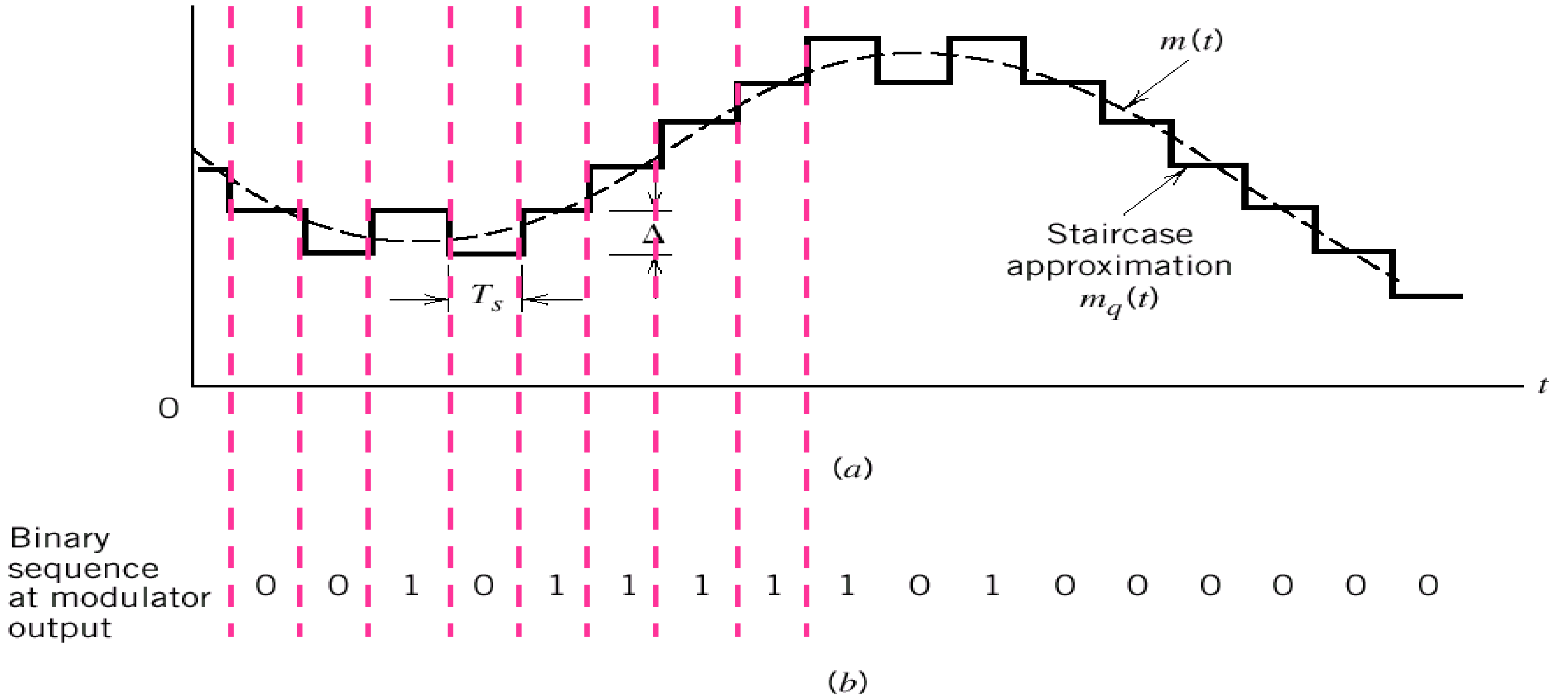
3.12 Delta Modulation DM
ΔM增量调制

DM provides a staircase approximation to the sampled version of the message signal.

The difference between the input and the approximation is quantized into only two levels, namely ±Δ, corresponding to positive and negative differences.

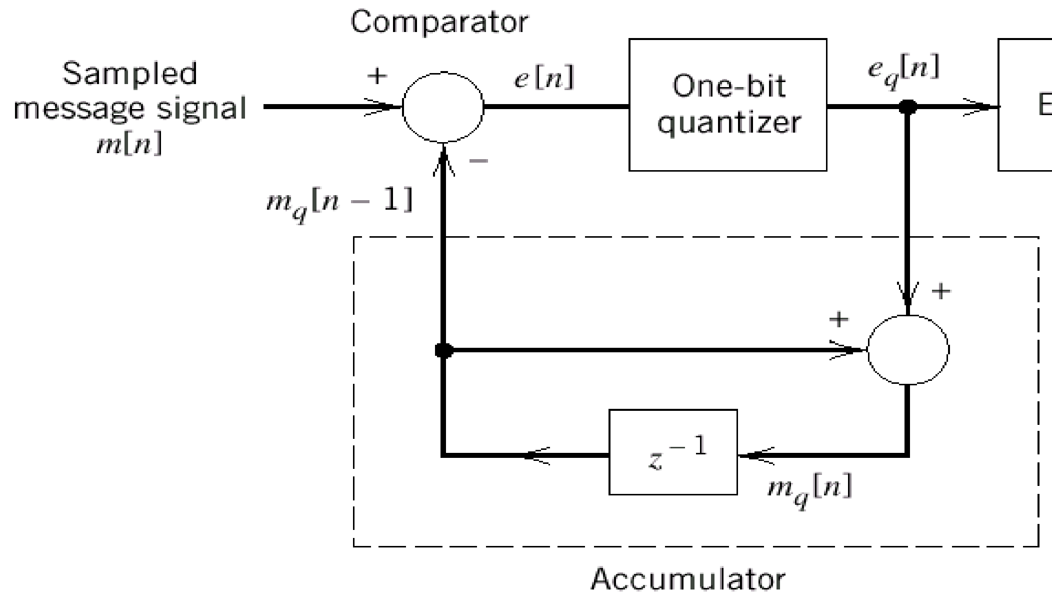
Oversampling (过抽样）： sampling at a rate much higher than the Nyquist rate.

Illustration of delta modulation



- If the approximation falls below the signal at any sampling epoch, it is increased by Δ.
- If the approximation lies above the signal at any sampling epoch, it is diminished by Δ.

DM system. (*a*) Transmitter. (*b*) Receiver.

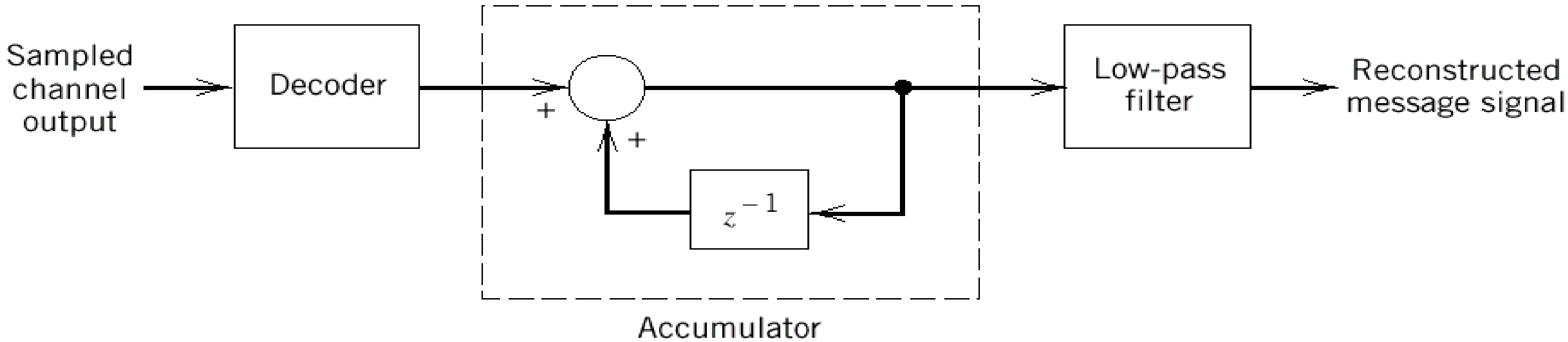


(*a*)

$$e[n] = m[n] - m_q[n-1]$$

$$e_q[n] = \Delta\,\mathrm{sgn}(e[n])$$

$$m_q[n] = m_q[n-1] + e_q[n]$$

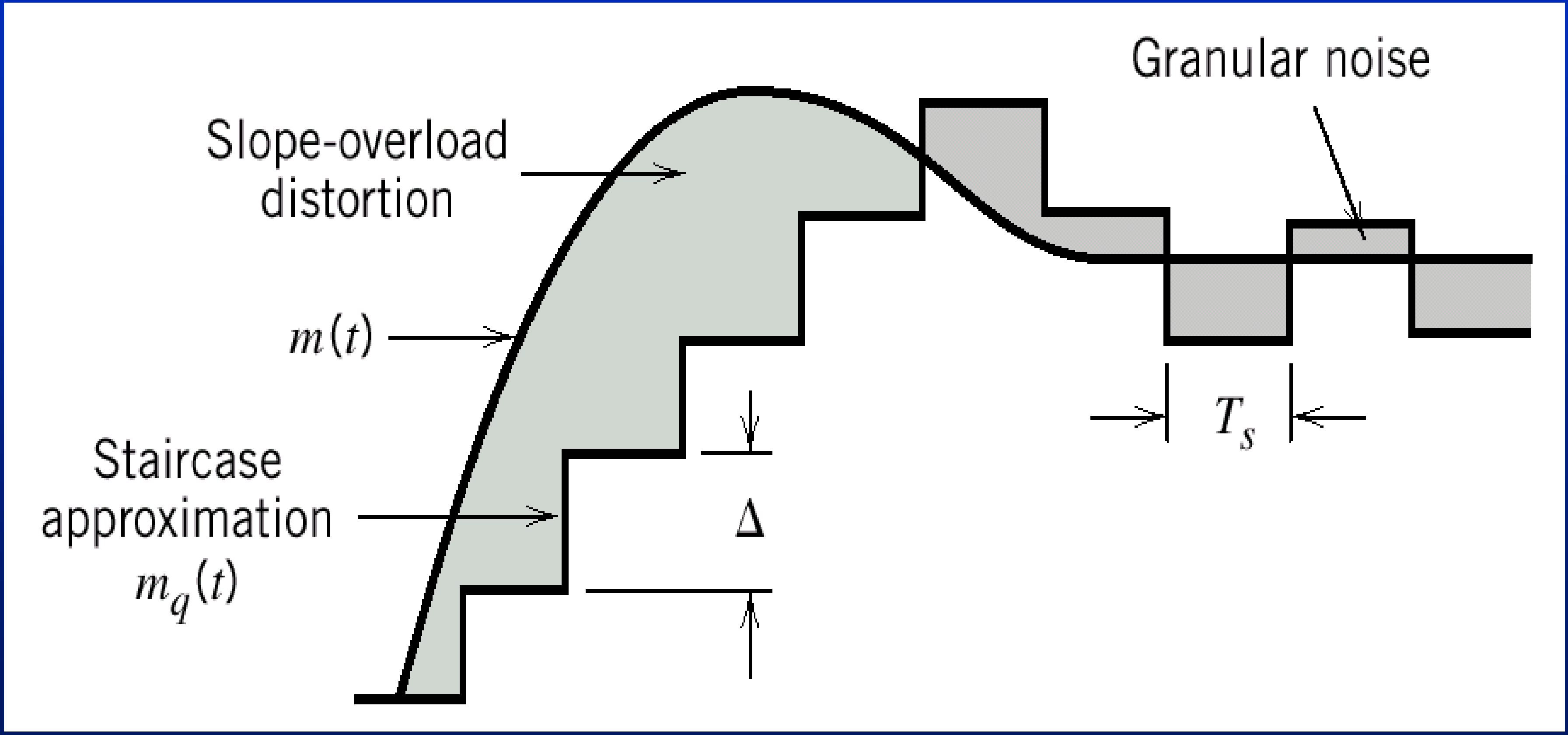


(*b*)

DM 存在两种量化噪声:

---Granular noise 颗粒噪声

---Slope overload distortion 斜率过载失真



Selection of Δ

- **In order to avoid slope overload, we need Large Δ**

$$\frac{\Delta}{T_s} \geq \max\left|\frac{dm(t)}{dt}\right|$$

- **In order to minimize granular noise, we need Small Δ.**

Solution : to make the delta "adaptive" 自适应. That is to say, the step size is varied with the input signal.

Summary

- Sampling
- Pulse-amplitude modulation PAM
- Quantization
- Pulse-code modulation PCM
- Time-division multiplexing TDM
- Digital multiplexers
- Improved PCM: DM , DPCM, ADPCM

18. 若用PXYZABCD表示A律 13 折线编码的码位安排，则其中P表示，XYZ表示________，ABCD表示__________。

19. PCM30/32 路电话系统在一帧内传送______________个码元。

20. 数字加密体制可分为____________、____________和混合密码体制。

21. 若二进制信息码为 10110000101，则当参考码元为“0”时，它的“1”差分码是________________。“0”差分码是______________________。

22. 若二进制信息码为 10110000101，则它的传号交替反转码(AMI)是_______________。HDB_3码是___________________。

homework

3.3	**3.8**	
3.9	**3.14**	**3.18**