

通訊系統

期末考輔導課

2021.01.08

- 1. Remaining part shall be done by yourself**
- 2. Highly recommend to read text book!**

Q1

(1) Draw the model of **FM receiver**

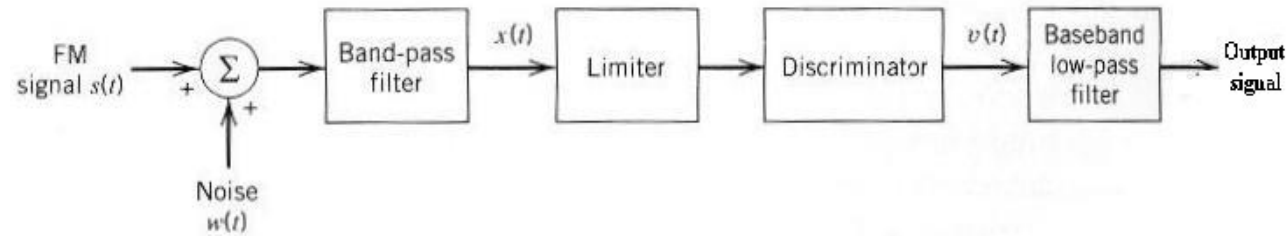


FIGURE 2.40 Model of an FM receiver.

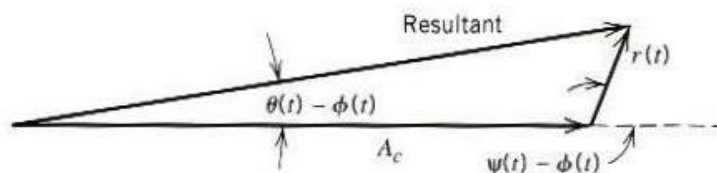
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(2) Why add a limiter in the **FM receiver**

=> 限制訊號大小，砍掉其他雜訊，以避免失真。

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(3) Using the **phasor diagram** to determine the discriminator output $\theta(t)$ for FM in AWGN channel. Assuming **high carrier to noise ratio**.



$$\therefore \theta(t) = \phi(t) + \frac{r(t) \sin[\psi(t) - \phi(t)]}{A_c + 0}$$

$$\phi(t) = 2\pi k_f \int_0^t m(t) dt$$

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Q2 (1) What is the **power spectral** density of **FM RCVR** output ?

$$S_{N_0}(f) = \begin{cases} \frac{N_0 f^2}{A_c^2}, & |f| \leq W \\ 0, & \text{otherwise} \end{cases}$$

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(2) What is the **average signal power**, **average noise power** and **output SNR** ?

$$(SNR)_o = \frac{k_f^2 P}{\frac{2N_0 W^3}{3A_c^2}} = \frac{3A_c^2 k_f^2 P}{2N_0 W^3}$$

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average signal power : $k_f^2 P$

noise power : $\frac{2N_0 W^3}{3A_c^2}$

Q3 (1) what is the **FM threshold effect** ?

⇒ 当載波雜訊比 (P) 小於閾限值 (P_{th}) , 其輸出訊雜比 (SNR)_o 會失真, 明顯產生偏移

⇒ 当載波雜訊比 P 高, 則 $(SNR)_{o, FM} = 3P \left(\frac{Bt}{2W} \right)^3$

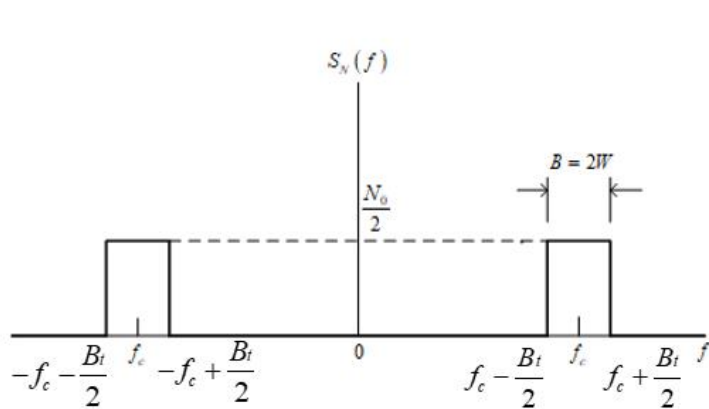
(2) Why FM is **used for satellite communication**? Please state the reasons.

1. FM 調頻展延能使 P_{th} 下降, 可以降低傳輸功率, 就算收到弱訊號也不失真
2. 發射機 PA 的非線性效應, 只會改變震幅大小, 不會改變相位

Q4

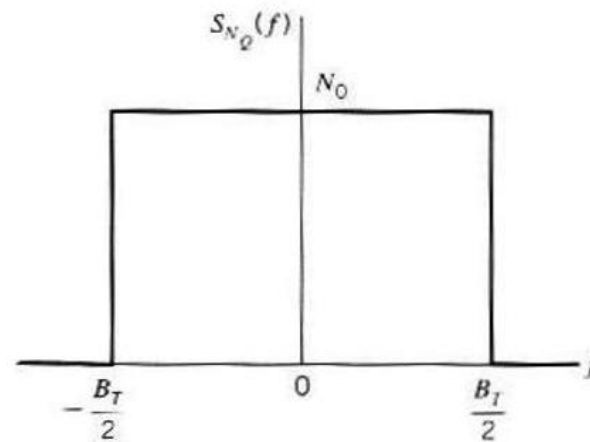
An FM signal with message bandwidth W Hz is passed through an AWGN channel with the white noise $w(t)$.

Write the equation to generate power spectral density of $w(t)$, $n(t)$, $n_Q(t)$, $n_d(t)$, $n_o(t)$

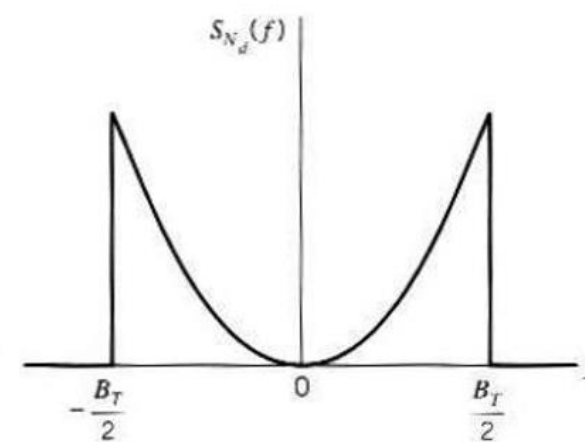


$$S_w(f) = \frac{N_0}{2}$$

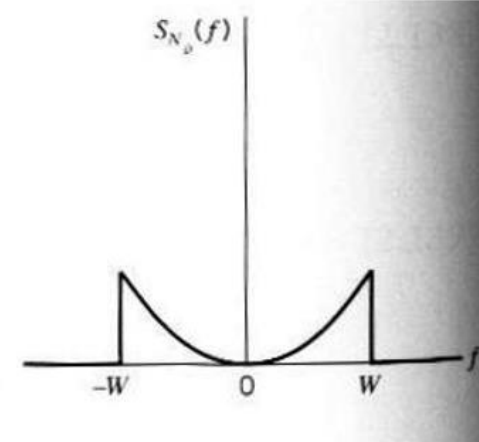
$$S_n(f) = \frac{N_0}{2}, -\frac{B_t}{2} \leq f \leq \frac{B_t}{2}$$



$$S_{N_Q}(f) = N_0, -\frac{B_t}{2} \leq f \leq \frac{B_t}{2}$$



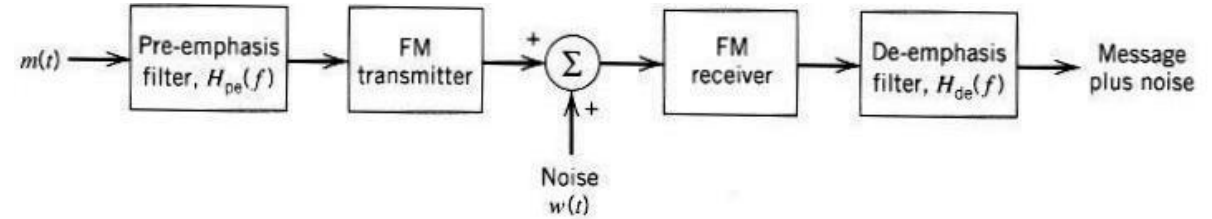
$$S_{N_d}(f) = \begin{cases} \frac{N_0 f^2}{A_c^2}, & -\frac{B_t}{2} \leq f \leq \frac{B_t}{2} \\ 0, & \text{otherwise} \end{cases}$$



$$S_{N_o}(f) = \begin{cases} \frac{N_0 f^2}{A_c^2}, & |f| \leq W \\ 0, & \text{otherwise} \end{cases}$$

Q5 (1) What is the **function** of **pre-emphasis filter** and **De-emphasis filter** in FM transceiver.

$$H_{Pe}(f) = 1 + \frac{jf}{f_0} \quad H_{De}(f) = \frac{1}{1 + \frac{jf}{f_0}}$$



(2) **Why they are needed.**

pre-emphasis filter(預強化濾波器): 強化高頻訊號

De-emphasis filter(去強化濾波器): 將高頻部分雜訊抑制掉，使訊號還原

Q6 Please draw the block diagram of PLL and state the major components.

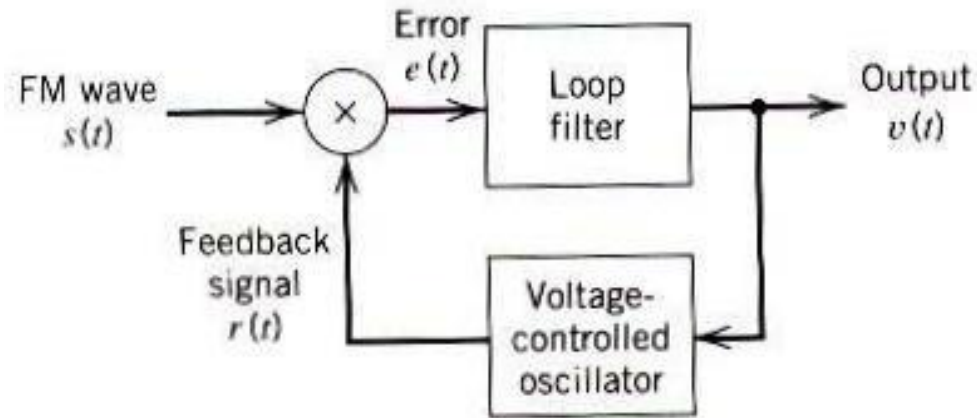
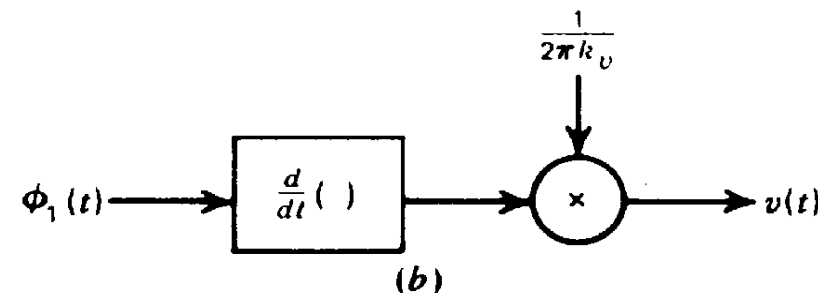


FIGURE 2.51 Phase-locked loop.

- ① 相位偵測器 (phase detector)
- ② 迴路濾波器 (LPF loop filter)
- ③ 電壓控制震盪器 (voltage - controlled oscillator)

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Q7 (1). Please draw the block diagram of simplified linear model of PLL, when loop gain is greater than 1.



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Q8: Compare the (1) figure of merit, (2) output signal to noise ratio for a fixed channel signal to noise ratio and (3) Transmission bandwidth for AM, DSBSC, LSBSC with $\beta = 2.5$ respectively

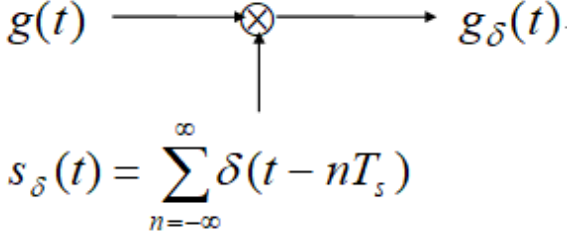
$$\begin{aligned}
 \text{1) } FOM_{AM} &= \frac{u^2}{2+u^2} < 1, \quad FOM_{AM, 100\%} = \frac{1}{3} \quad (u=1) \\
 FOM_{DSBSC} &= 1 \\
 FOM_{LSBSC} &= 1 \\
 FOM_{FM} &= \frac{3}{2} D^2 = \frac{3}{2} \beta^2 = 6 \\
 \text{2) } SNR_{o, AM} &= \frac{A_c^2 k_a^2 P_{av, m}}{2 W N_0} \\
 SNR_{o, DSBSC} &= \frac{C^2 A_c^2 P_{av, m}}{2 N_0 W} \\
 SNR_{o, LSBSC} &= \frac{C^2 A_c^2 P_{av, m}}{4 N_0 W} \\
 SNR_{o, FM} &= \frac{3 A_c^2 k_f^2 P_{av, m}}{2 N_0 W^3} \\
 \text{3) } B_{t, AM} &= 2W \text{ (Hz)} \\
 B_{t, DSBSC} &= 2W \text{ (Hz)} \\
 B_{t, LSBSC} &= W \text{ (Hz)} \\
 B_{t, FM} &= 2(W + \Delta f) = 2W(\beta + 1) = 6W \text{ Hz}
 \end{aligned}$$

課後問答10

Q1

If the bandwidth of analog signal $g(t)$ is W Hz, the Nyquist sampling rate is used to **get PAM signal**.

(1) Write the **spectrum equation** of $G_\delta(f)$ in terms of $G(f)$.



The diagram shows an analog signal $g(t)$ entering a multiplier block (represented by a circle with an 'X'). A sampling signal $s_\delta(t) = \sum_{n=-\infty}^{\infty} \delta(t - nT_s)$ is also input to the multiplier. The output is the PAM signal $g_\delta(t)$.

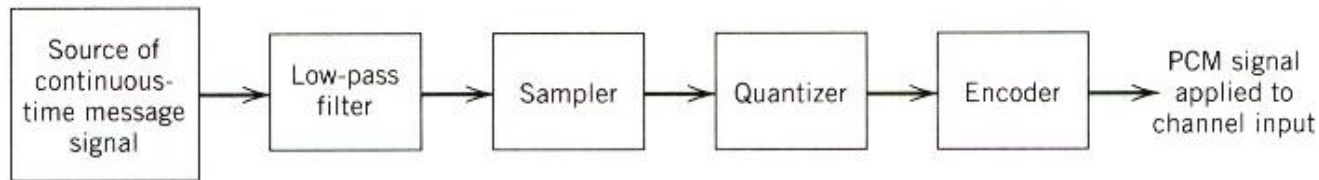
$$g_\delta(t) = g(t)s_\delta(t) = \sum_{n=-\infty}^{\infty} g(nT_s)\delta(t - nT_s)$$

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

$$= 2W G(f+2W) + 2W G(f) + 2W G(f-2W) + \dots$$

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(2) Please draw the **block diagram of a PCM transmitter**.



(a) Transmitter

(3) Determine the frequency **transfer function** of the **LPF**.

$$H(f) = \frac{1}{f_s} \text{rect}\left(\frac{f}{2W}\right) = \frac{1}{f_s}, -W \leq f \leq W$$

$$h(t) = \text{sinc}(2Wt)$$

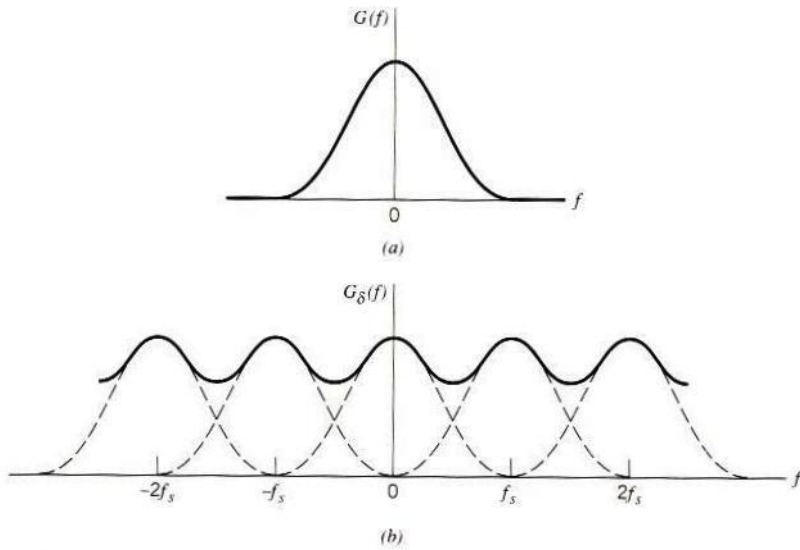
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Q2

(1) How do you **avoid the aliasing**?

取樣頻率要大於兩倍 Nyquist rate $f_s \geq 2W$

(2) Draw the **sampled spectrum with aliasing** error.



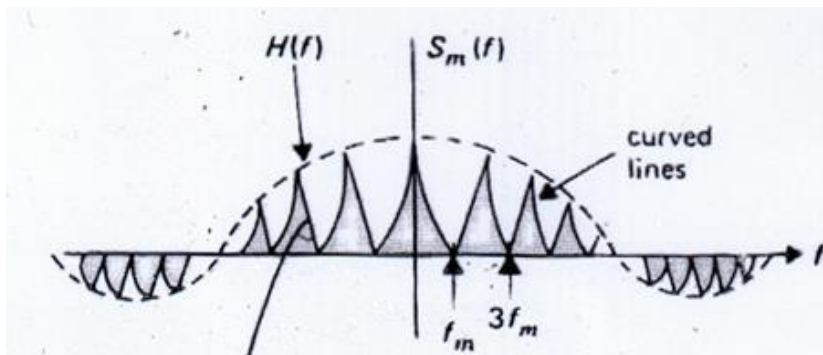
Q3

(1) If the impulse response of **zero order hold circuit** is $h(t)=1, 0 \leq t \leq T$, please use the **transfer function** explain the aperture effect.

$$\begin{aligned} s(t) &= h(t) * g(t) \\ S(f) &= H(f) G(f) \\ &= f_s \sum_{m=-\infty}^{\infty} G(f - m f_s) H(f) \end{aligned}$$

when $H(f) = T \text{sinc}(fT) e^{-j2\pi f \frac{T}{2}}$
 $H(f)$ cause amplitude distortion
and $\frac{T}{2}$ delay called aperture effect

(2) Draw the spectrum of a sampled signal $g_s(t)$ with **aperture effect 孔徑效應**.



aperture effect 孔徑效應:

zero order hold 所造成，導致取樣頻譜的振幅失真和相位失真

(3) Why do you **need zero order hold circuit** ?

使訊號穩定，避免訊號產生氣泡

所以使用**zero order hold** 把訊號**hold**住一段時間

Q4

(1) For a sinusoidal signal $g(t) = A_m \sin 2\pi f t$, the **quantized** sample is encoded with **R bits**, derive the **output signal to noise ratio** of A/D.

$$\begin{aligned}
 g(t) &= A_m \sin 2\pi f t \\
 M_{\max} &= A_m \\
 \Delta &= \frac{2A_m}{2^R} \\
 \Rightarrow \sigma_q^2 &= \frac{1}{3} M_{\max}^2 2^{-2R} = \frac{1}{3} A_m^2 2^{-2R}
 \end{aligned}$$

$$\begin{aligned}
 (SNR)_o &= \frac{P}{\sigma_p^2} = \left(\frac{3P}{M_{\max}^2} \right) 2^{2R} \\
 &= \left(\frac{3P}{A_m^2} \right) 2^{2R} = \frac{3}{2} 2^{2R} \\
 \therefore 10 \log (SNR)_o &= 1.8 + 6R \text{ (dB)}
 \end{aligned}$$

$$\begin{aligned}
 \Delta &= \frac{2 M_{\max}}{2^R} \\
 \sigma_q^2 &= \frac{\Delta^2}{12} \\
 &= \frac{1}{3} M_{\max}^2 2^{-2R}
 \end{aligned}$$

$$SNR_o = \frac{P}{\sigma_p^2} = \left(\frac{3P}{M_{\max}^2} \right) 2^{2R}$$

(2) If $R=8$ bits, what is the gain of A/D.

(3) What is the function of oversampling? Please state the reasons.

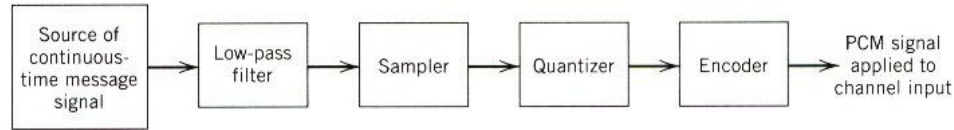
$$\begin{aligned}
 R = 8 \text{ bits} &\Rightarrow 10 \log (SNR)_o = 1.8 + 6 \times 8 \\
 &= 49.8 \text{ (dB)}
 \end{aligned}$$

oversampling A/D $\Rightarrow f_s \gg 2W$
 (取樣頻率大於2倍頻寬)
 量化雜訊平均分佈在 $(-\frac{\Delta}{2}, \frac{\Delta}{2})$, 所以雜訊在
 $W < |f| < \frac{f_s}{2}$ 可被低通濾波器濾除 ($f_s \uparrow, \sigma_q^2 \downarrow$)
 \Rightarrow 目的: 降低雜訊

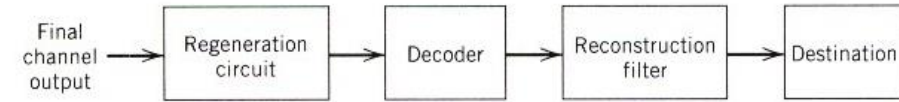
Q5

Please draw the **block diagram** of a **PCM transmitter and receiver**.

should add one blocks respectively.



(a) Transmitter



(c) Receiver

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Write the **impulse response equation** $h_i(t)$ of ideal reconstruction filter.

$$h_i(t) = \text{sinc}(2Wt)$$

Write the **output equation** of ideal reconstruction filter. If the bandwidth of analog signal $g(t)$ is W Hz, the Nyquist sampling rate is used to **get PAM signal**.

$$\begin{aligned} g(t) &= g_s(t) * h_i(t) \\ &= \sum_{n=-\infty}^{\infty} g(nT_s) \delta(t - nT_s) * h_i(t) \\ &= \sum_{n=-\infty}^{\infty} g(nT_s) h_i(t - nT_s) \\ &= \sum_{n=-\infty}^{\infty} g(nT_s) \text{sinc}(2W(t - nT_s)) \end{aligned}$$

By teacher's note

Q6

A TDM (分時多工) is used for four users, which have bandwidth of $w_1=w_2=10\text{k Hz}$, $w_3=20\text{k Hz}$, $w_4=40\text{k Hz}$

(1) What is the **maximum sample interval**? (用戶最取頻寬去求得最大取樣區間)

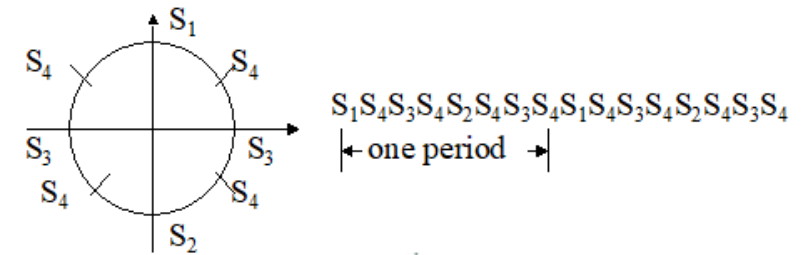
$$\boxed{T = \frac{1}{2W}}$$

$$\begin{aligned} w_1 = 10\text{k(Hz)} &\Rightarrow \frac{1}{20\text{k}} = T_1 \\ w_2 = 10\text{k(Hz)} &\Rightarrow \frac{1}{20\text{k}} = T_2 \\ w_3 = 20\text{k(Hz)} &\Rightarrow \frac{1}{40\text{k}} = T_3 \\ w_4 = 40\text{k(Hz)} &\Rightarrow \frac{1}{80\text{k}} = T_4 \end{aligned}$$

$$\therefore T_s = \frac{1}{20\text{k}} \text{ (sec)}$$

(2) Determine the number of **samples per sampling interval**. (每個取樣區間的樣本)

$$\begin{aligned} n_s &= \frac{T_s}{T_1} + \frac{T_s}{T_2} + \frac{T_s}{T_3} + \frac{T_s}{T_4} \\ &= \frac{\frac{1}{20}}{\frac{1}{20}} + \frac{\frac{1}{20}}{\frac{1}{20}} + \frac{\frac{1}{20}}{\frac{1}{40}} + \frac{\frac{1}{20}}{\frac{1}{80}} = 1 + 1 + 2 + 4 = 8 \end{aligned}$$



(3) Determine the **data rate** for 8 bits PCM modulation.

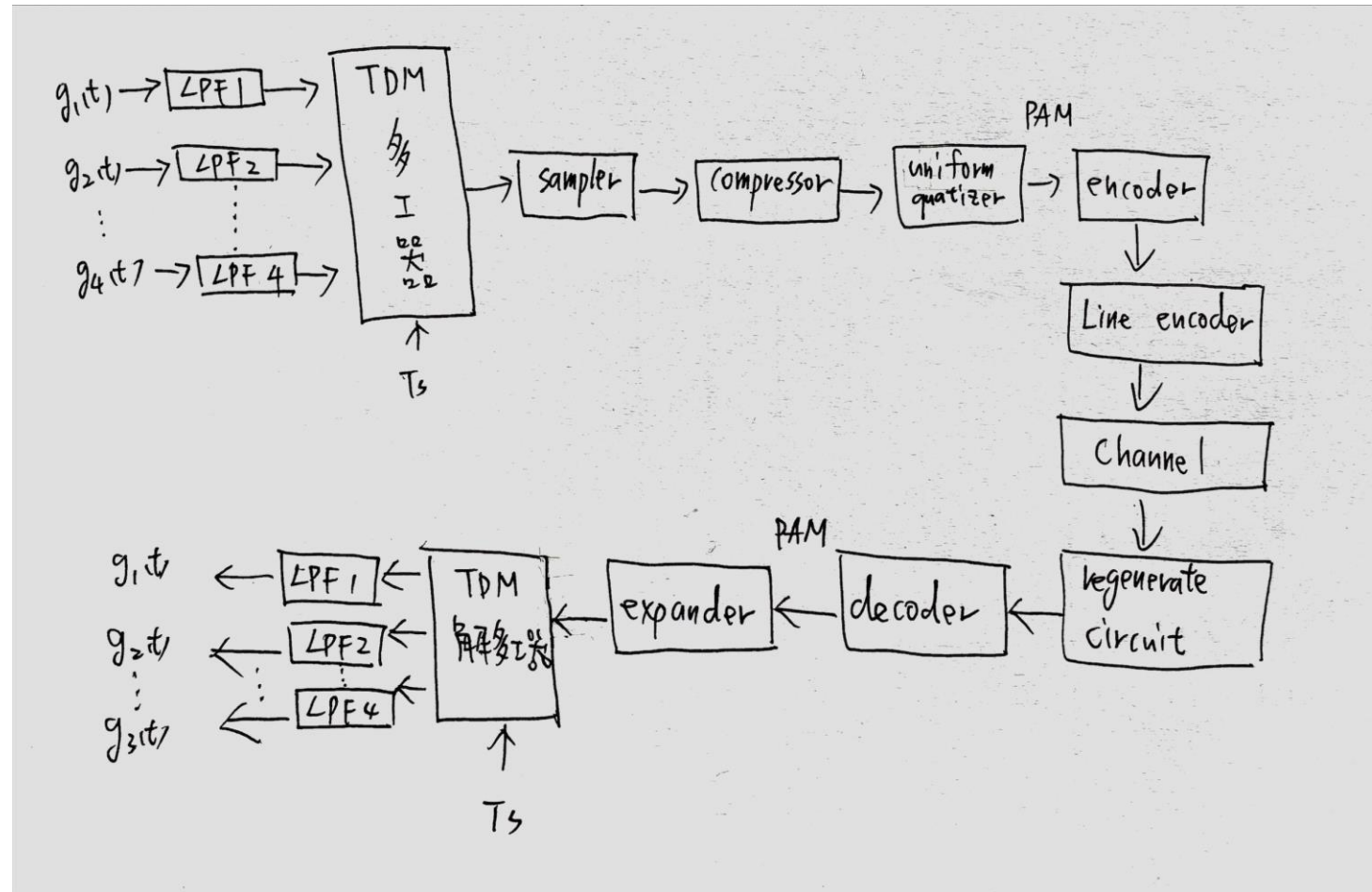
$$R_b = \frac{n_s \times \text{bit}}{T_s}$$

$$R_b = \frac{8 \times 8}{T_s} = \frac{8 \times 8}{\frac{1}{20k}}$$

$$= 1280 \text{ k bit/sec}$$

Q7

Please draw the block diagram of a PCM transmitter and receiver, where TDM accommodates 4 voice channel and the compander can get high resolution for small speech signal



What is the type of compression law used in T1 modem?

μ -law, $n=255$, 15 piecewise-linear segment

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What is the physical meaning of 8 bit PCM codeword at the decoding output of T1?

B1 \Rightarrow sign bit

B2~B4 \Rightarrow identify the segment number

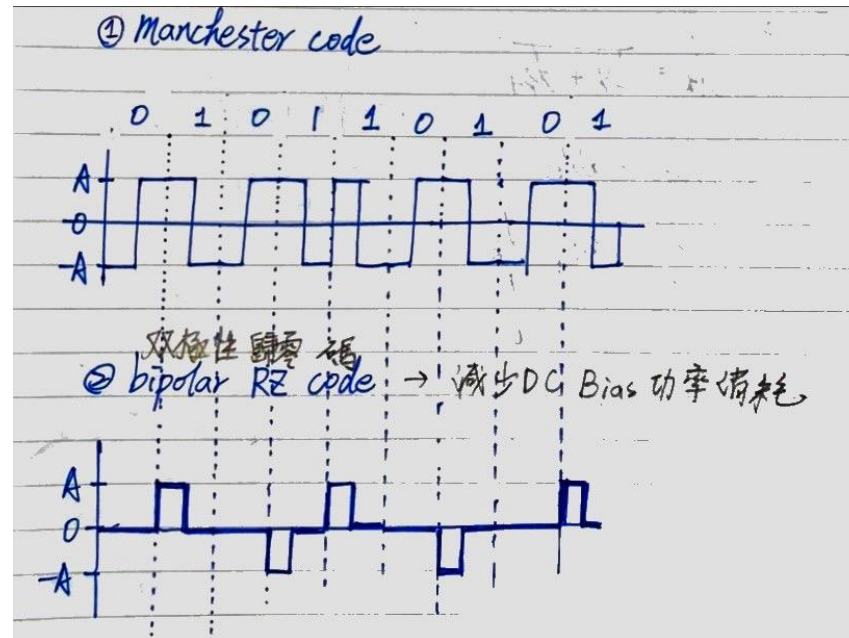
B5~B8 \Rightarrow identify the quantization level in each segment

If the PCM codeword is 01100110, what is sign, segment and quantization level for the decoding output of T1 modem?

$$\begin{array}{lll} \text{sign: } 0 & 110 & 0110 \\ \text{segment: } 7 & & \text{quantization level: } 2+4+1=7 \end{array}$$

Q8

Draw the linear code waveform of 010110101 using
(1) Manchester code, bipolar RZ code with the reference bit of 1.



Please write the formula of the differential encoding if b_k is the current bit and d_{k-1} is the previous difference bit and write the truth table

$$d_k = \overline{b_k \oplus d_{k-1}}$$

b_k	d_{k-1}	d_k
0	0	1
0	1	0
1	0	0
1	1	1