QAM解調之MATLAB專題實作

大綱

QAM調變與解調

- 目的
- 原理
- 實驗步驟
  - 音訊取樣及調變
  - 訊號及頻譜分析
  - 16-QAM解調及播放解調訊號
- 實驗項目

目的

- 了解16-QAM調變之運作原理。
- 利用真實的16-QAM調變訊號來測試接收機之設計。
- 可以利用MATLAB進行聲音播放。

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原理：16-QAM簡介

- QAM(Quadrature amplitude modulation)是由兩個相位相差九十度的正交調幅載波所組成，假設每次傳送區塊中含有k個位元，其可以分割為兩組分別為(k/2)位元的子區塊(假設k是偶數)，這兩個區塊都使用(k/2)位元的D/A轉換器，以提供載波所需的調變電壓。

原理：16-QAM簡介

- 下面描述了一個產生16-QAM訊號的數學模型。
- The 16-QAM signal is represented by
  $s(t) = A_R \cos(2\pi f_c t) + A_I \sin(2\pi f_c t)$

where A_R and A_I are real numbers. The mapping from the information vector to (A_R, A_I) is given in Figure 2 .

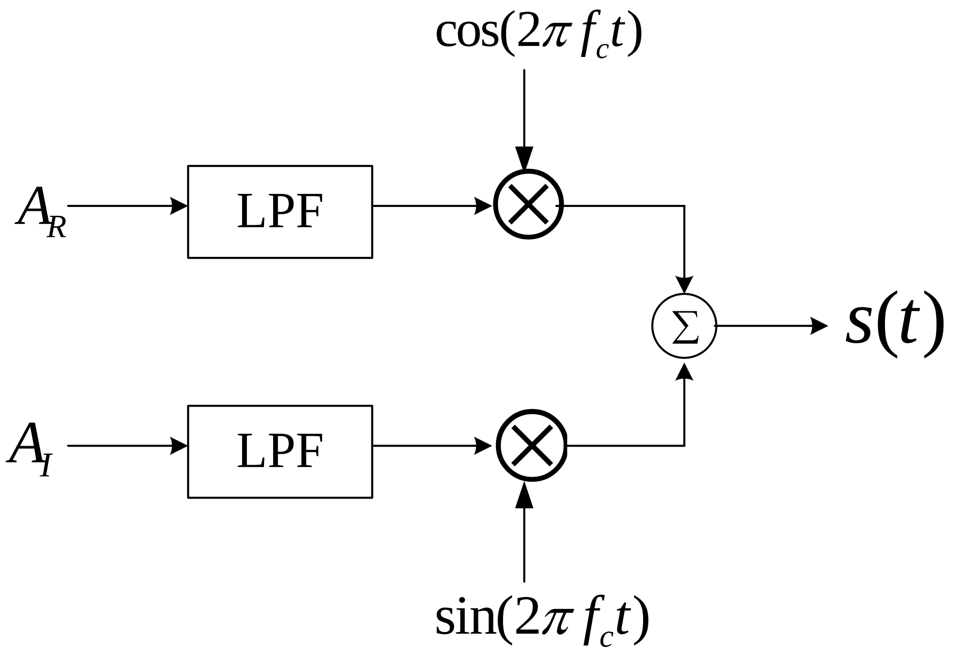
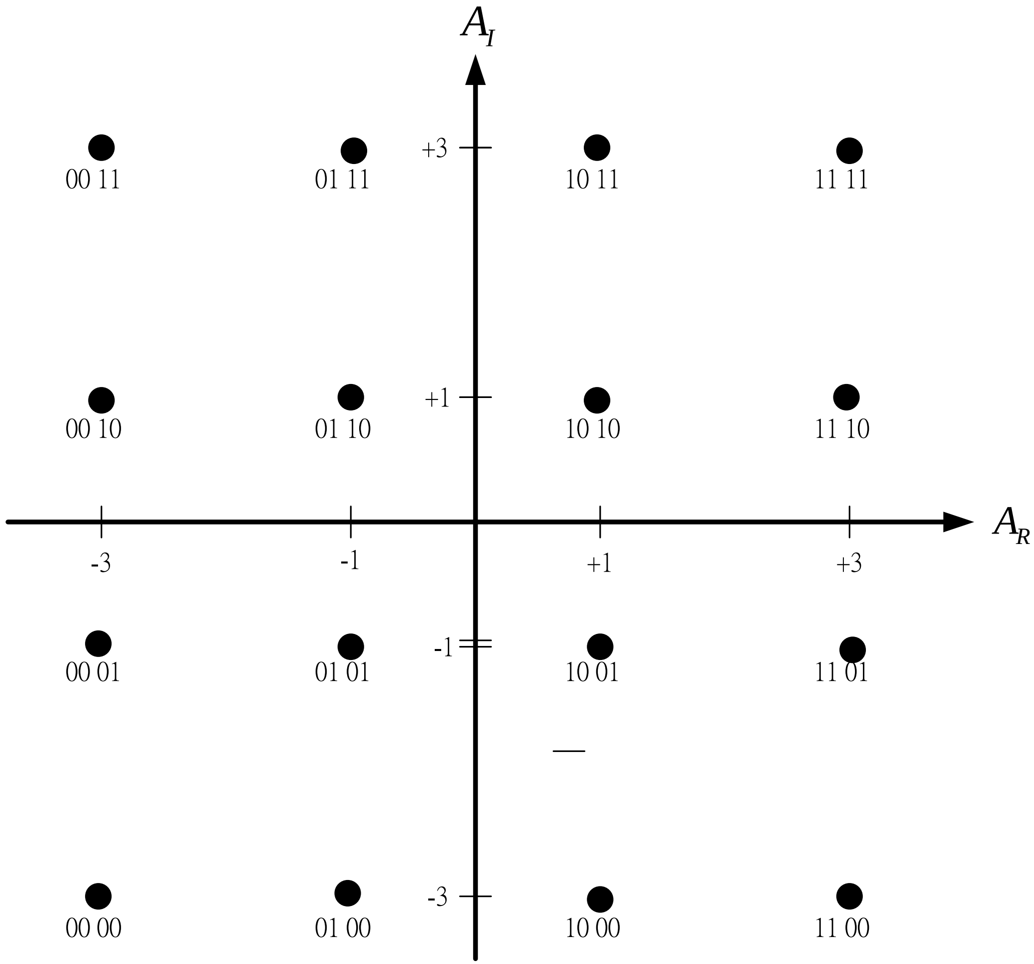


Figure 1: QAM modulator

原理 :16-QAM解調



A_R A_I

$\boxed{0\ 0}$ $\boxed{0\ 0}$

$00 \rightarrow -3$

$01 \rightarrow -1$

$10 \rightarrow +1$

$11 \rightarrow +3$

Figure 2: 16-QAM constellation

原理 :16-QAM解調

- 16-QAM 解調

即為projection on orthogonal basis

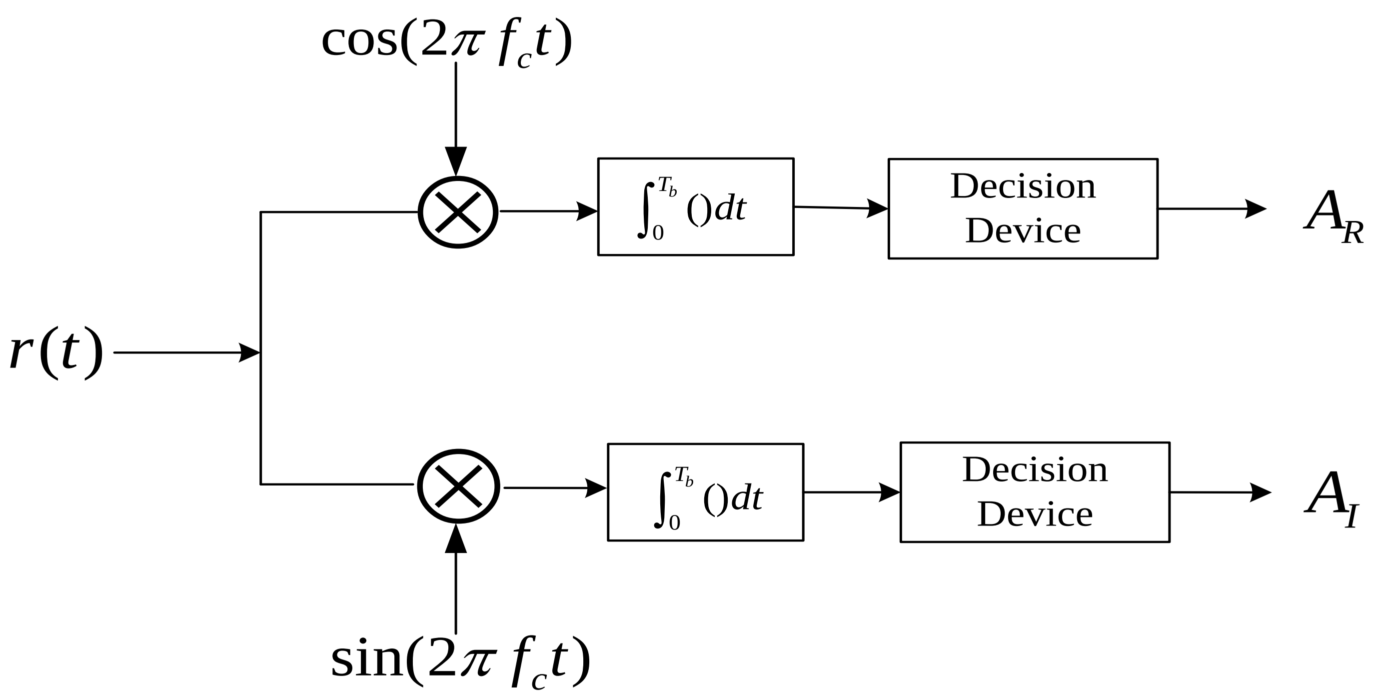


Figure 3: QAM demodulator

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參考文獻

步驟：音訊取樣進行調變

- A voice transmission system with digital communication is shown in Figure 4.

- The voice signal is sampled with sampling rate of 8kHz，and Each sampled signal is then quantized to 8 bits (0~255)。

- The total bit rate for the voice stream = $64\text{kbps}\left(8\text{bit}\times 8\text{k}\right)$

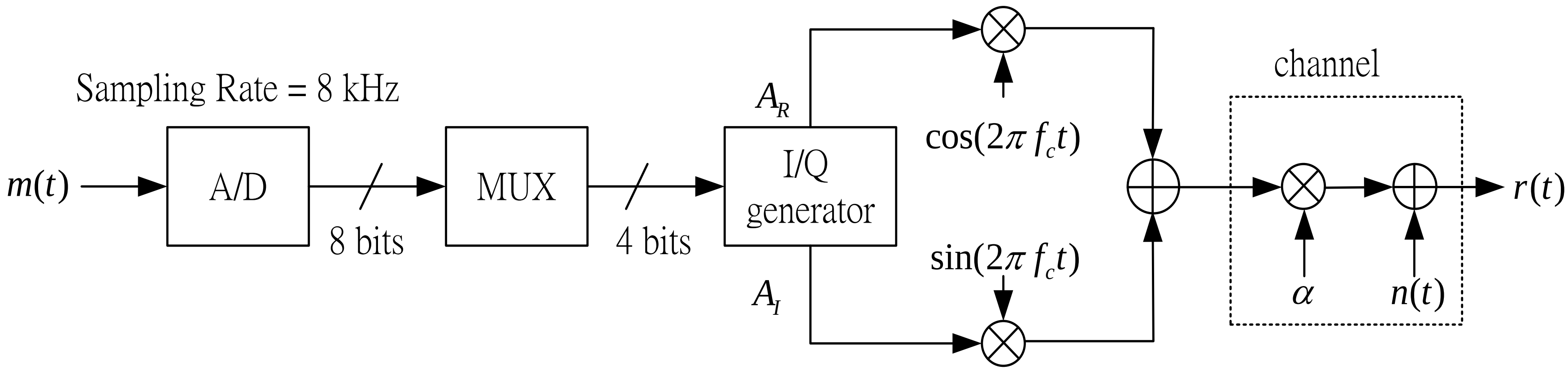


Figure 4: The voice transmission system with 16-QAM.

步驟：音訊取樣進行調變

- The range and levels of the quantizer is shown in Figure 5.

0~255

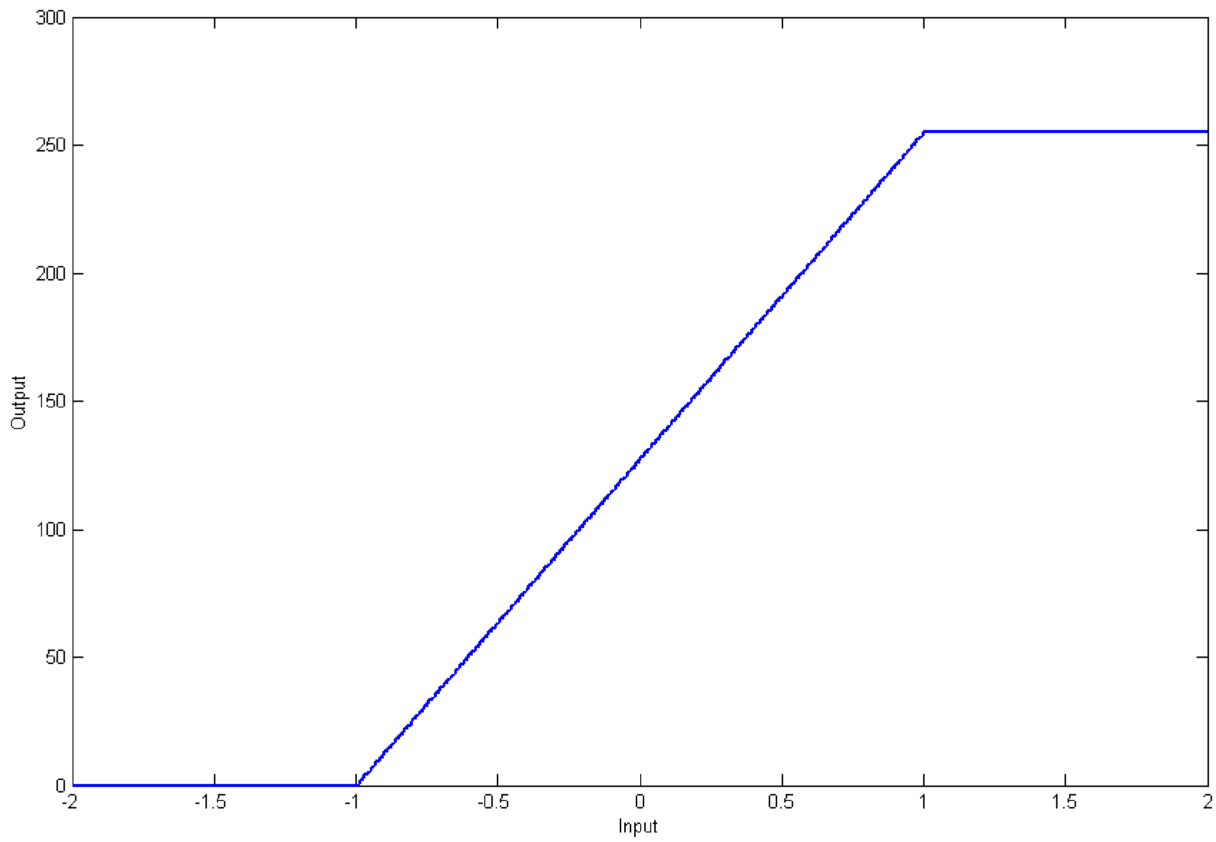


Figure 5: Input/Output relation of the quantizer.

步驟：音訊取樣進行調變

- The voice stream is then modulated by a 16-QAM modulator.

- Let $(\boxed{b_7,b_6,b_5,b_4},\boxed{b_3,b_2,b_1,b_0})$ be a voice sample. Each voice sample is then divided into two vectors, (b_3,b_2,b_1,b_0) and (b_7,b_6,b_5,b_4) .

- The first vector (b_3,b_2,b_1,b_0) is modulated into the first 16-QAM signal and the second vector (b_7,b_6,b_5,b_4) is modulated into the second 16-QAM signal.

步驟：音訊取樣進行調變

- In our system, the carrier frequency is assumed to be $f_c = 64$ kHz, the symbol interval is 62.5 μs which corresponds to a baud rate of 16kHz $(64k\text{bps}/4\text{bits})$.

$$\frac{1}{(64k\text{bps}/4\text{bits})}$$

- The signal $s(t)$ is then transmitted over the channel.

- In the channel, the signal is attenuated by a factor α.

步驟：音訊取樣進行調變

- In addition, an additive white Gaussian noise is added resulting the final received signal $r(t)$ expressed as

$$r(t) = \alpha(A_R \cos(2\pi f_c t) + A_I \sin(2\pi f_c t)) + n(t)$$

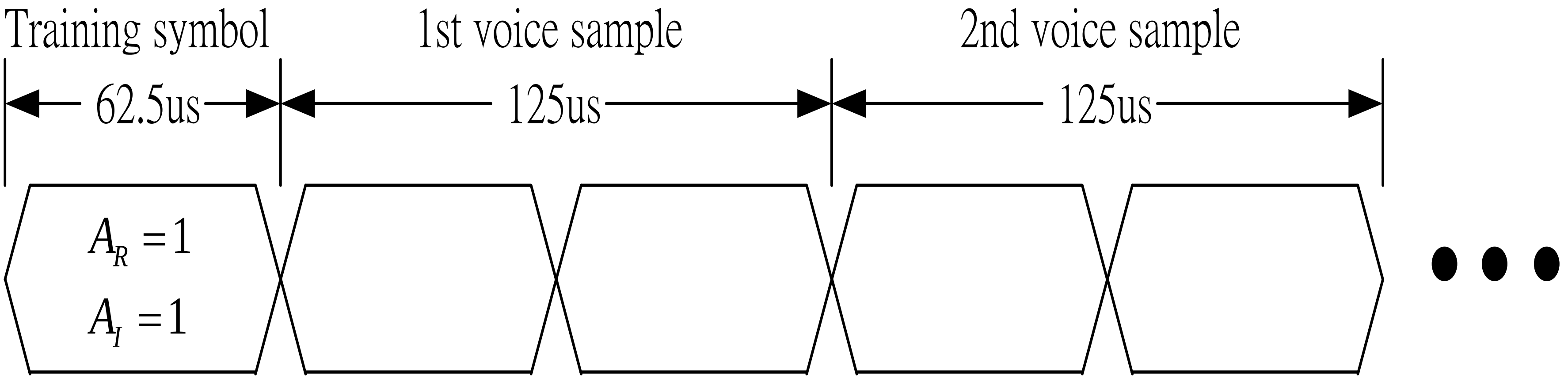
- In order to estimate the attenuation factor α , a training symbol with

$A_R = 1$ and $A_I = 1$ is transmitted prior to the actual data.

- Figure 7 illustrates the signal format.

步驟一：音訊取樣進行調變

- Figure 7: Signal format for voice transmission



步驟：取樣及調變MATLAB程式

```
clear all;

am=[
-3,
-1,
1,
3
];

fc = 8e3*8; %載波
fsample = 64e3*8; %取樣頻率
brate = 2e3*8;
T = 1/brate;
Tsample = 1/fsample;
phi1 = sin(2*pi*fc*[0:31]*Tsample);
phi2 = cos(2*pi*fc*[0:31]*Tsample);

[y,FS,NBITS]=WAVREAD('sample1.wav',[1+8000, 10*8e3+8000]);

% soundsc(y,FS,NBITS);
```

取其長度80000的部份

步驟：取樣及調變MATLAB程式

- 續上頁程式碼，以下為調變過程。

最大值可能為247(+1)

最小值可能為7 (-1)

每個symbol有32個取樣點

Training symbol length

MUX使8bit的變為4bit，使得symbol數呈兩倍

Training symbol

```
mean_y = mean(y);
max_y = max(abs(y-mean_y));
z = round((y-mean_y)/max_y*120+127);
rxQAM16=zeros(1, length (z)*2*32+32);

rxQAM16(1:32) = 1*phi1+1*phi2;

for i = 1: length (z)
    rxQAM16((i-1)*64+1+32: (i-1)*64+32+32)=
      am(bitget(z(i),1)+2*bitget(z(i),2)+1)*phi1 + am(bitget(z(i),3)
     + 2*bitget(z(i),4)+1)*phi2;
    rxQAM16((i-1)*64+33+32: (i-1)*64+64+32)=
     am(bitget(z(i),5)+ 2*bitget(z(i),6)+1)*phi1 + am(bitget(z(i),7)
    + 2*bitget(z(i),8)+1)*phi2;
end

rxQAM16N = rxQAM16+randn(size(rxQAM16))*0.4;
save rxQAM16N;
```

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振幅調變與解調

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實驗項目

- Estimate the channel attenuation factor α .
- Demodulate the received signal and plot the first 5000 received vectors in a 2D signal space. Describe the differences of the demodulated vectors between the two received signals **rxQAM16.m** and **rxQAM16N.m**.
- Reconstruct the original voice signal and play it.