Generation Of M-PAM, M-PSK And M-QAM In AWGN Channel

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Abstract

Today, digital modulations have become part and parcel of the present and future communication technologies. Despite the advantages of these schemes, the traditional channel impairments, such as noise, can affect their performance. Moreover, data transmission is mostly done over wireless channels, which are very unpredictable and are characterised by multipath fading effects. This paper presents a short research article that presents a study of digital modulation schemes (M-ary PAM, M-ary PSK and M-ary QAM) using MATLAB under Additive White Gaussian Noise (AWGN). The result shows that, among the three modulation schemes compared, ...(write name) has the best BER performance with minimal energy consumption.

I. Introduction

There are several digital modulation schemes that have been proposed but there is a trade-off between data rate and mismatch of the three basic parameters as phase, frequency and time between the transmitter and receiver. These digital modulation schemes include basic schemes such as MPSK (M-ary Phase Shift Keying), MPAM (M-ary Pulse Amplitude Modulation), MQAM (M-ary Quadrature Amplitude Modulation). M-ary modulation schemes are one of most efficient digital data transmission systems as it achieves better bandwidth efficiency than other modulation techniques and give higher data rate[1] . The objective of this paper is to review the key characteristics and performance of digital modulation schemes. Simulations are used to compare the performance and trade-off of M- ary techniques, including analysis of BER in the presence of Additive White Gaussian Noise (AWGN).

II. THEORETICAL DESCRIPTION

The waveform $s_m(t)$ used to transmit information over the communication channel can be,in general,of any form. However, usually these waveforms are bandpass signals which may differ in amplitude or phase or frequency, or some combination of two or more signal parameters.

A. *M-ARY TECHNIQUES*

In an M-ary signaling scheme, we may send one of M possible signals $s_1(t), s_2(t) \ldots s_m(t)$, during each signaling interval of duration Ts. The symbol duration Ts = nTb, where Tb is the bit duration. In pass-band data transmission these signals are generated by changing the amplitude, phase, frequency of a sinusoidal carrier in M discrete steps thus we have M-ary ASK, M-ary PSK and M-ary QAM digital modulation schemes Different bandwidth efficiency at the expense of power efficiency can be achieved using M-ary modulation schemes[2]

B. *M-ary PAM*

The PAM signals are carrier-modulated band pass signals with lowpass equivalents of the form. $s_{ml} = Re[s_{ml}(t)e^{j2\pi f_c t}]$, where f_c is the carrier frequency[3]. The basis for the bandpass PAM signal given by $s_{ml} = A_m\sqrt{\frac{E_g}{2}}\phi$. The corresponding signal space diagrams for M = 2, M = 4, and M = 8 are shown in fig 1 (a). The bandpass digital PAM is also called amplitude-shift keying (ASK). The symbol error probability is given by,

$$P_e = 2Q\left(\sqrt{\frac{6log_2 M E_{bavg}}{(M^2-1)N_o}}\right) \tag{1}$$

C. *M-ary PSK*

In digital phase modulation, the M signal waveforms are represented as

$$s_{ml} = Re[g(t)e^{\frac{j2\pi(m-1)}{M}}e^{2}\pi f_c t]$$

where m = 1,2,...,M and g(t) is the signal pulse shape and $\theta_m = 2\pi(m-1)/M$ m = 1, 2,..., M is the M possible phases of the carrier that convey the transmitted information. Digital phase modulation is usually called phase-shift keying (PSK). The corresponding signal space diagram is shown in Fig.1 (b). The Probability of error is given by,

$$P_e = 2Q\left(\sqrt{\frac{2\pi^2 log_2 M E_b}{M^2 N_o}}\right) \tag{2}$$

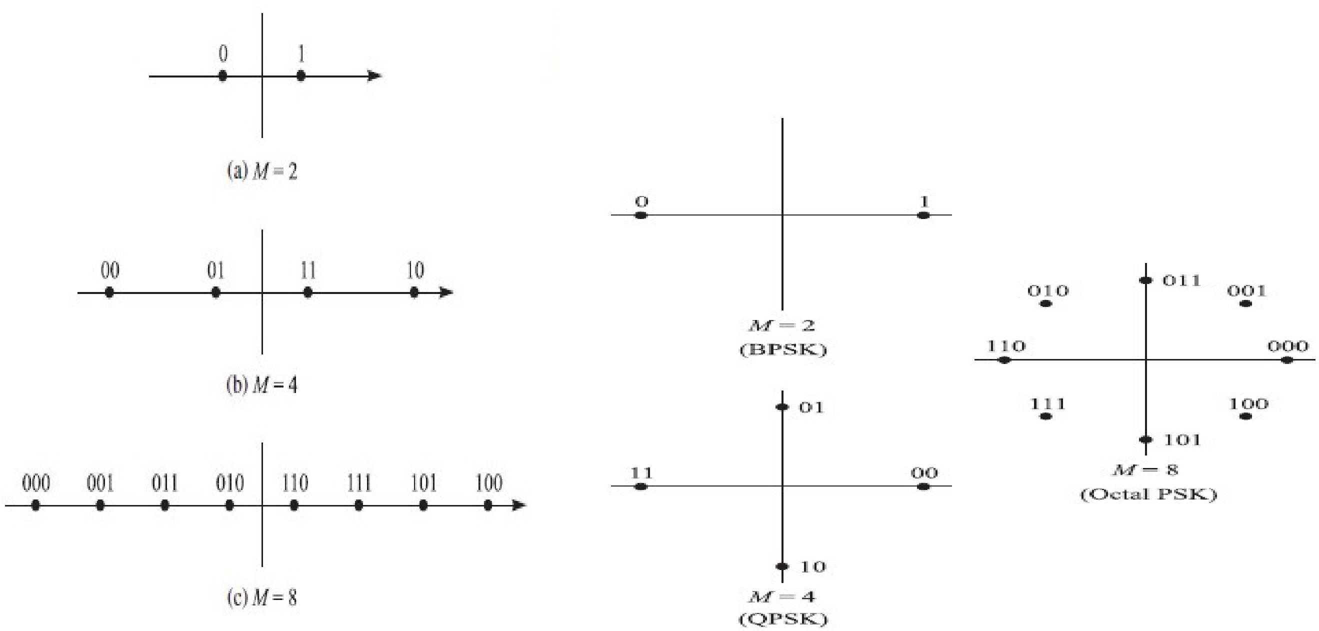
D. *M-ary QAM*

The QAM signal waveform may be viewed as combined amplitude(r_m) and phase(θ_m) modulation. The corresponding signal waveform may be expressed as,

$$s_{ml} = Re[(A_{mi} + jA_{mq})g(t)e^{2\pi f_c t}]$$

where m = 1,2,...,M and $A_m i$ and $A_m q$ are the information bearing signal amplitudes of the quadrature carrier and g_t is the signal pulse. The signal space shown in Fig. 2 The probability of error is given by,

$$P_e = 4Q\left(\sqrt{\frac{3log_2 M E_{bavg}}{(M-1)N_o}}\right) \tag{3}$$



(a) PAM

(b) PSK

Fig. 1. **Signal space diagram for PAM and PSK**

E. *AWGN (Additive white Gaussian noise)*

In communication systems, the most common type of noise added over the channel is the Additive White Gaussian Noise (AWGN). It is additive because the received signal is equal to the transmitted signal plus the noise. It is white because it has a constant power spectral density. It is Gaussian because its probability density function can be accurately modelled to behave like a Gaussian distribution. It is noise because it distorts the received signal. Because the bandwidth of the signal is very less as compare to the bandwidth of the AWGN channel. The higher the variance of the noise, the more is the deviation of the received symbols with respect to the constellation set and, thus, the higher is the probability to demodulate a wrong symbol and make errors[4]

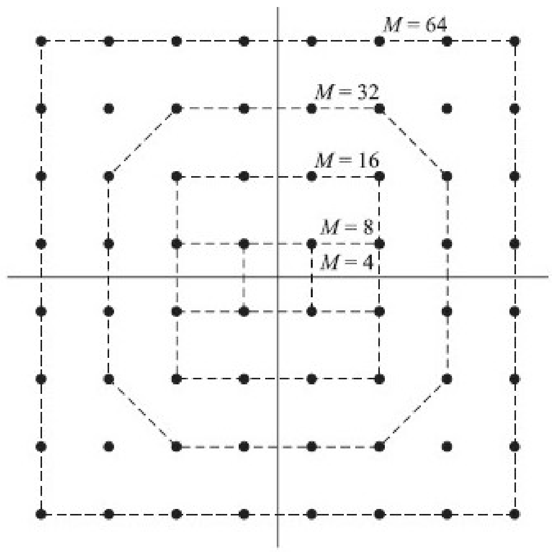


Fig. 2. **Signal space diagram for reactangular QAM**

F. *Bit Error Rate*

Bit Error Rate (BER) is the number of bit errors that occur for a given number of bits transmitted. It is related to the error probability because it is the ratio of bit errors to bits transmitted. The energy per bit is the amount of power in a digital bit for a given amount of time.

III. SIMULATION RESUNTS AND DISCUSSIONS

A. *Performance of M-PAM*

The BER for M-ary PAM for different value of E_b/N_o for the AWGN channel is shown in Fig. 3.

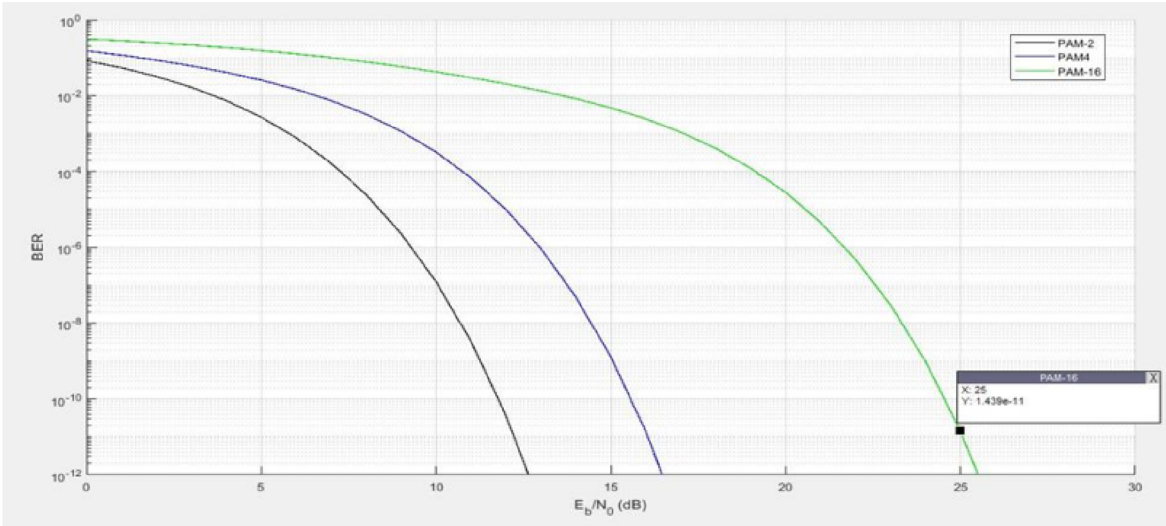


Fig. 3. **BER Vs E_b/No for M-PAM**

B. Performance of M-PSK

The BER for M-ary PSK for different value of E_b/N_o for the AWGN channel is shown in Fig. 4.

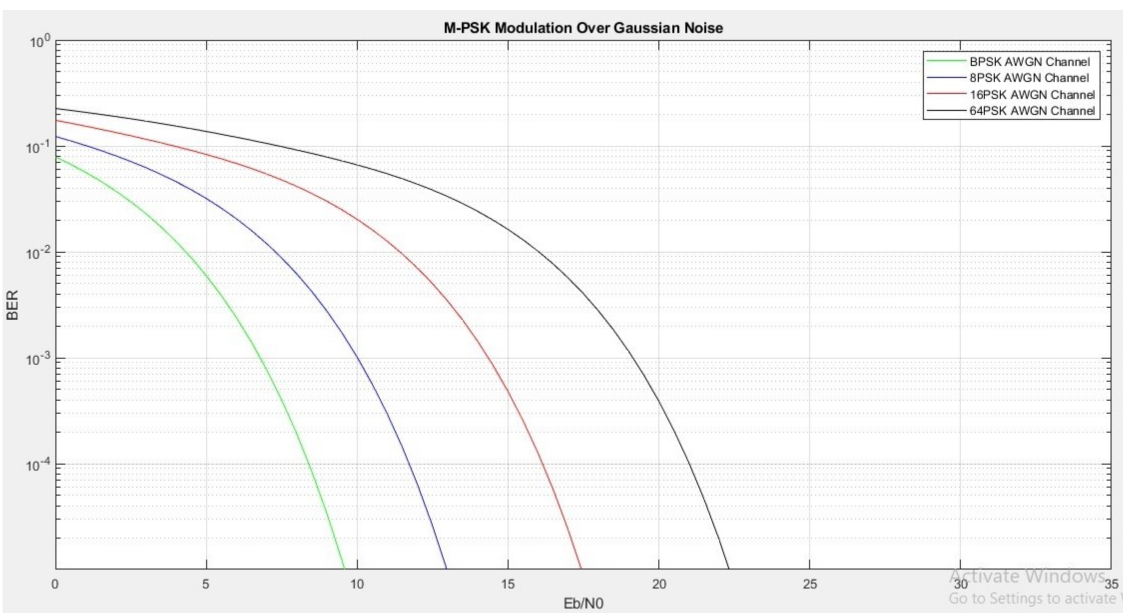


Fig. 4. **BER Vs E_b/No for M-PSK**

C. *Performance of M-QAM*

The BER for M-ary QAM for different value of E_b/N_o for the AWGN channel is shown in Fig. 5.

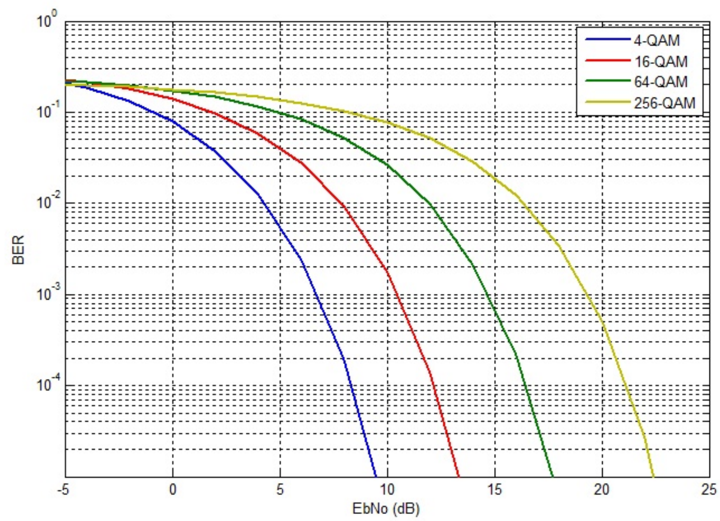


Fig. 5. **BER Vs E_b/No for M-QAM**

IV. Conclusion

From above graph it can be concluded that BER for all systems decrease monotonically with increase in Eb/No. Normally the Bit Error Rate is measured by the distance between two nearest possible signal points in the signal space diagram (constellation diagram) as the distance between two points decreases the possibility of error increases. Hence distance should be as large as Possible. So, probability of BER increases as M increases. In a AWGN channel, for large values of $Eb/No.$, the error probability decreases exponentially with respect to $Eb/No.$.

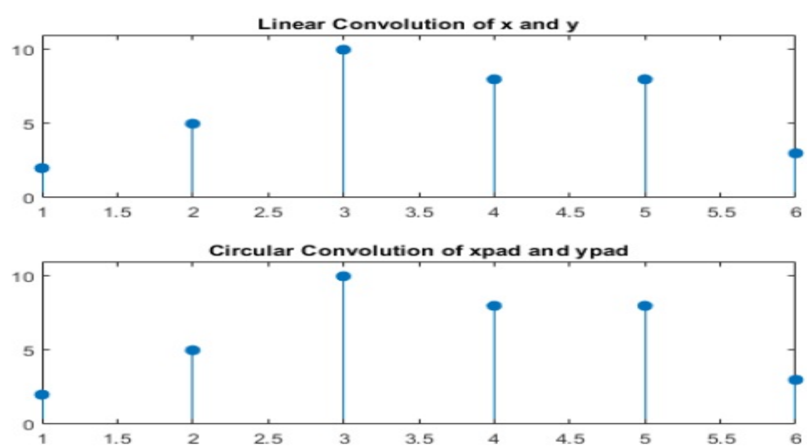
The value of M increases means more number of bits are combined to make a symbol and these bits are packed more closely in signal constellation. From this experiment, it is proven that between the different M-ary modulation schemes experimented with, M-PAM has the lowest bit error rate within the same Eb/No value of the AWGN channel used with the other modulation schemes considered in the study. Between BPSK, QPSK, and M-PSK, BPSK has the lowest bit error rate within the same Eb/No values.

V. LAB SHEET

A. Write a function to perform linear and circular convolutions on two arrays. Compare the result of your function with the result of linear convolution using FFT method.

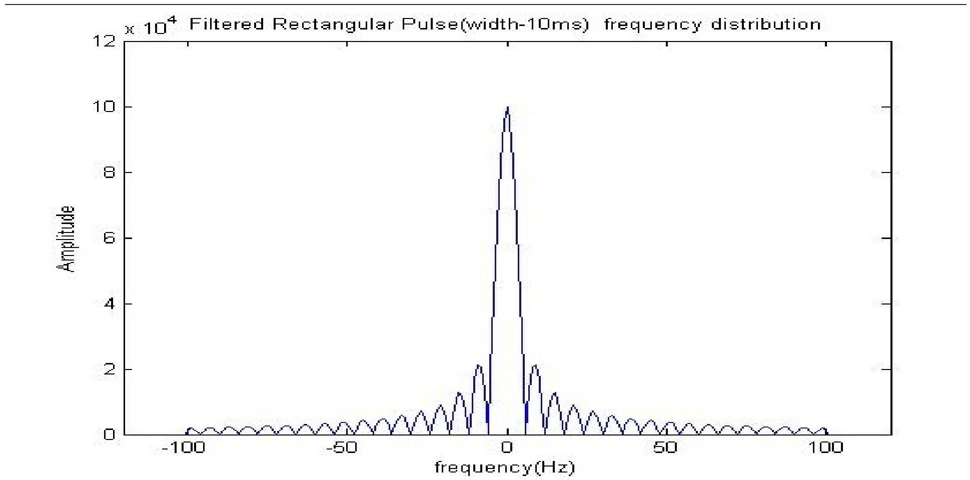
Answer:

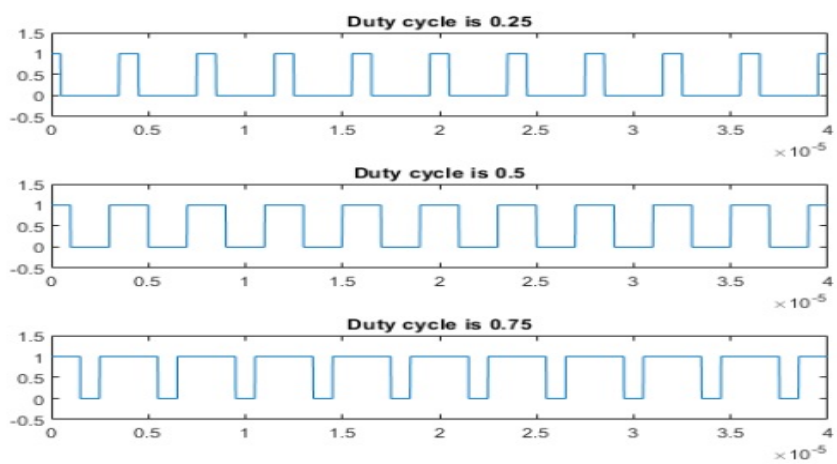
Linear and circular convolution are fundamentally different operations. However, there are conditions under which linear and circular convolution are equivalent. Establishing this equivalence has important implications. For two vectors, x and y , the circular convolution is equal to the inverse discrete Fourier transform (DFT) of the product of the vectors. DFTs. Knowing the conditions under which linear and circular convolution are equivalent allows you to use the DFT to efficiently compute linear convolutions. The linear convolution of an $N - point$ vector,x , and an L-point vector,y, has length $N + L = 1$. For the circular convolution of x and y to be equivalent, you must pad the vectors with zeros to length at least $N + L = 1$ before you take the DFT. After you invert the product of the DFTs, retain only the first $N + L = 1$ elements. Create two vectors , x and y and compute the linear convolution of the two vectors. $x = [2 \quad 1 \quad 2 \quad 1]$; and $y = [1 \quad 2 \quad 3]$, $C_{lin} = Con(x, y)$. The output has length $4 + 3 = 1$. Pad both vectors with zeros to length $4 + 3 = 1$. Obtain the DFT of both vectors, multiply the DFTs, and obtain the inverse DFT of the product. $x_{pad} = [x_{zeros}(1, 6 - length(x))]$; $y_{pad} = [y_{zeros}(1, 6 - length(y))]$; $C_{Circ} = IIFT[FFT(x_{pad}). * FFT(y_{pad})]$; The circular convolution of the zero-padded vectors, x_{pad} and y_{pad} , is equivalent to the linear convolution of x and y you retain all the elements of C_{Circ} because the output has length 4 + 3 = 1. Plot of the output of linear convolution and the inverse of the DFT product to show the equivalence is shown below .



B. Write a function gateduty (N, d) to generate a rectangular pulse of length N and duty cycle d . Generate M periods of this pulse, and compute the power spectral density (PSD). Plot the pulse in both time and frequency. Plot the PSD of the pulse. Write your program so that M, N and d can be taken as user input. Comment on the value of PSD based on the variation in N and d for a single pulse period.

Answer

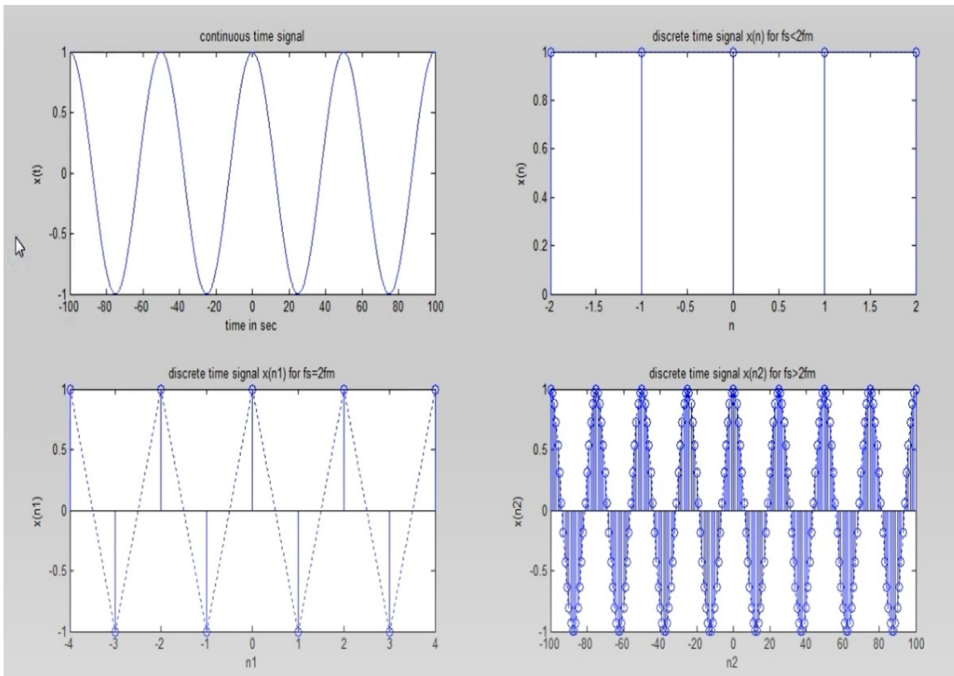




C. Sampling of sinosoidal

Sampling Theorem Statement:

Sampling theorem states that “continues form of a time-variant signal can be represented in the discrete form of a signal with help of samples and the sampled (discrete) signal can be recovered to original form when the sampling signal frequency Fs having the greater frequency value than or equal to the input signal frequency Fm. i.e. $F_s \geq 2F_m$. If the sampling frequency (Fs) equals twice the input signal frequency (Fm), then such a condition is called the Nyquist Criteria for sampling. When sampling frequency equals twice the input signal frequency is known as "Nyquist rate" i.e. $F_s = 2F_m$



REFERENCES

[1] R. Suthar, S. Joshi, and N. Agrawal, "Performance analysis of different m-ary modulation techniques in cellular mobile communication."

[2] S. Haykins, "Digital communication," *disp*, vol. 11, p. 10, 2010.

[3] J. G. Proakis and M. Salehi, *Digital communications*. McGraw-Hill., 2008.

[4] T. S. Rappaport *et al.*, *Wireless communications: principles and practice*. prentice hall PTR New Jersey, 1996, vol. 2.