通訊系統 (II)

國立清華大學電機系暨通訊工程研究所
蔡育仁
台達館 821 室
Tel: 62210
E-mail: yrtsai@ee.nthu.edu.tw

課程要求

- 課程要求
  - Homework: 30 %
  - Midterm Exam: 35 %
  - Final Exam: 35 %
- 教科書：
  - Communication Systems, Simon Haykin **(4th Ed./5th Ed.)** John Wiley & Sons, Inc.
- 講義位置：https://nyquist.ee.nthu.edu.tw/WCS.html
  (Password: CommsysII20250219EE4640)
- 助教時間：每週二13:20~15:10, EECS 605 室
- 助教：TWNTHUEE4640@gmail.com

課程內容

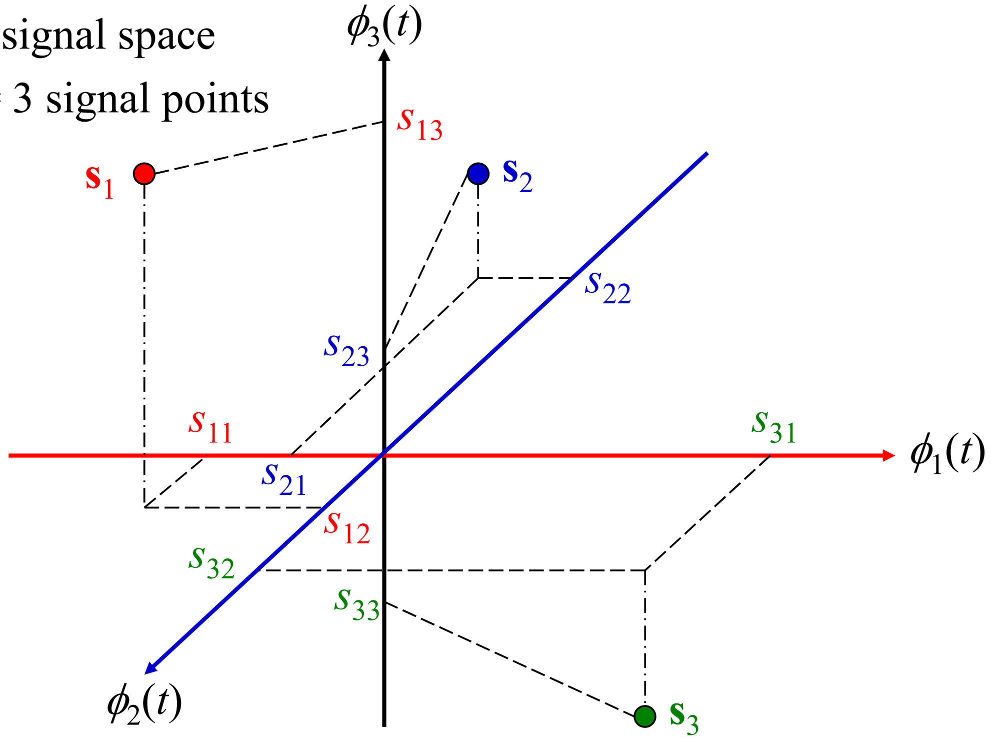
- Preliminaries
- Ch. 1: Signal-Space Analysis
- Ch. 2: Phase-Shift Keying Modulation
- Ch. 3: Hybrid Amplitude/Phase Modulation
- Ch. 4: Frequency-Shift Keying Modulation
- Ch. 5: Detection of Signals with Unknown Phase (Non-coherent Detection)
- Ch. 6: Comparison of Digital Modulation Schemes Using a Single Carrier ← 期中考試
- Ch. 7: Information Theory
- Ch. 8: Multichannel Modulation
- Ch. 9: Error-Control Coding
- Ch. 10: Spread-Spectrum Modulation ← 期末考試

Introductory Courses

- Signals and Systems
  - Signals and Systems
  - Linear Time-Invariant Systems
  - Fourier Analysis
- Probability Theory
  - Probability
  - Statistic
- Communications System I

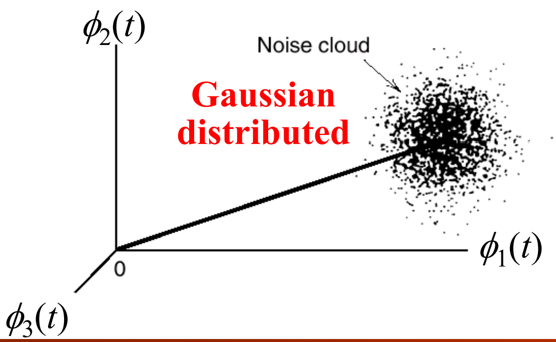
Ch. 1 – Signal-Space Analysis

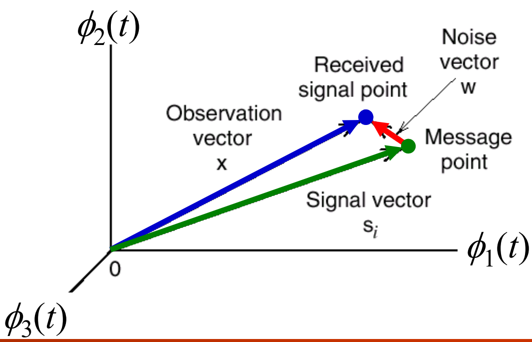
- 3D signal space
- $N = 3$ signal points



Ch. 1 – Signal-Space Analysis (Cont.)

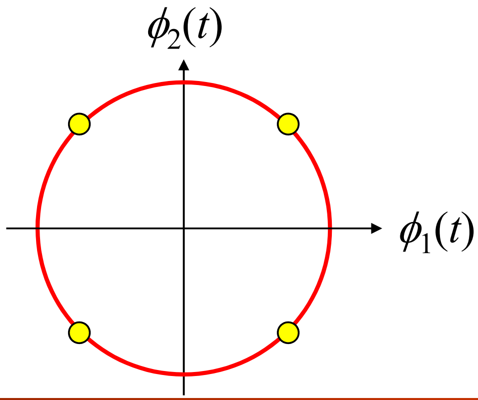
- Signal Detection – **MAP** (**maximum *a posteriori* probability**) and **ML** (**maximum likelihood**) decision rules
- The observation vector **x** (**received signal point**) differs from the transmitted signal vector $\mathbf{s}_i$ by a **random noise vector w**
- Given the observation vector **x**, perform a mapping from **x** to an **estimate** $\hat{m}$ of the transmitted symbol m_i





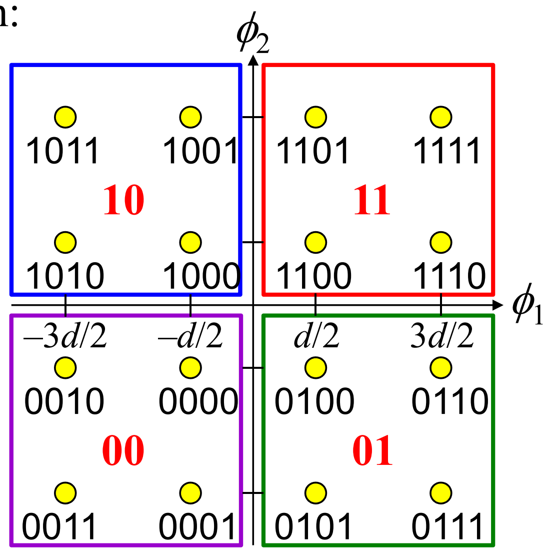
Ch. 2 – Phase-Shift Keying Modulation

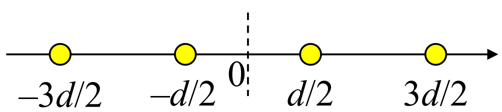
- In an ***M*-ary** PSK modulation scheme, multiple bits are transmitted in a symbol
- The signal are generated by changing the phase of a sinusoidal carrier in ***M* discrete steps**
- In QPSK, the phase of the carrier takes on one of four **equally spaced** values, such as **π/4, 3π/4, 5π/4, and 7π/4**



Ch. 3 – Hybrid Amplitude/Phase Modulation

- *M*-ary **Quadrature Amplitude Modulation** (**QAM**) is a **two-dimensional** generalization of *M*-ary PAM (**Pulse-Amplitude Modulation**)
- Consider a 16-QAM constellation: 4 bits per symbol

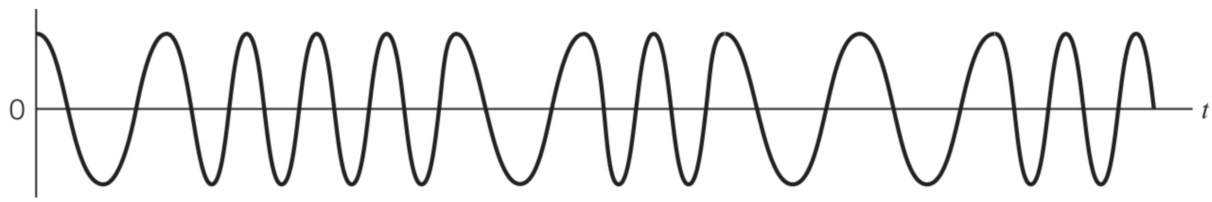




One-dimensional 4-ary PAM constellation

Ch. 4 – Frequency-Shift Keying Modulation

- In an ***M*-ary** FSK modulation scheme, multiple bits are transmitted in a symbol
- The signal are generated by changing the frequency of a sinusoidal carrier in ***M* discrete steps**
- In **binary FSK**, symbols 1 and 0 are distinguished from each other by transmitting one of two sinusoidal waves that **differ in frequency by a fixed amount**



Ch. 5–Detection of Signals with Unknown Phase

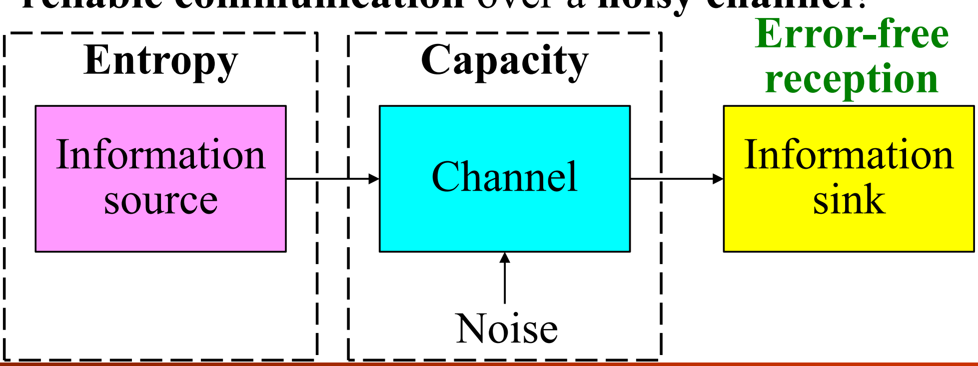
- In previous study, we assume that the receiver is **perfectly synchronized** (in both **frequency** and **phase**) to the transmitter
  - The only **channel impairment** is **AWGN**
- In practice, there is also uncertainty due to the randomness of certain signal parameters; for example, a **time-variant channel**
- The **phase** may change in a way that the receiver cannot follow
  - The receiver **cannot estimate** the received carrier phase
  - The carrier phase may **change too rapidly** for the receiver to **track**
- A digital communication receiver with no provision made for **carrier phase recovery** is said to be **noncoherent**
  - **Noncoherent detection**

Ch. 6–Comparison of Digital Modulation Schemes

- The popular digital modulation schemes are classified into **two categories**, depending on the method of **detection** used at the receiver:
  - **Class I, Coherent detection**:
    - Binary PSK: two symbols, single frequency
    - Binary FSK: two symbols, two frequencies
    - QPSK: four symbols, single frequency—includes the QAM as a special case
    - MSK: four symbols, two frequencies
  - **Class II, Noncoherent detection**:
    - DPSK: two symbols, single frequency
    - Binary FSK: two symbols, two frequencies

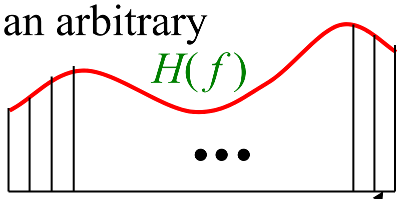
Ch. 7 – Information Theory

- In communications, **information theory** deals with modeling and analysis of a **communication system**
- In particular, it provides answers to two fundamental questions:
  - **Signal Source:** What is the irreducible **complexity**, below which a signal **cannot be compressed**?
  - **Channel:** What is the ultimate **transmission rate** for **reliable communication** over a **noisy channel**?



Ch. 8 – Multichannel Modulation

- Consider a linear **wideband channel** with an arbitrary frequency response $H(f)$.
  - The magnitude response $|H(f)|$ is approximated by a **staircase** function
  - Δf: the width of each **subchannel**



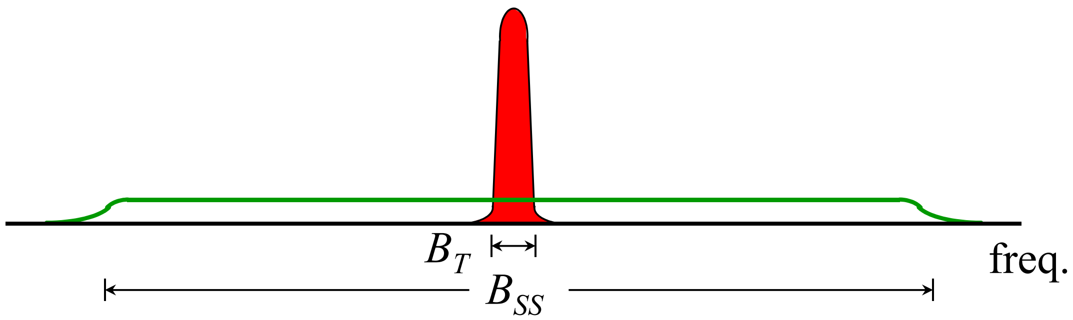
- In each step, the channel may be assumed to operate as an AWGN channel **free from inter-symbol interference**.
- **Power Loading** is to **maximize** the bit rate R through an **optimal sharing** of the total transmit power P between the N subchannels
  - Subject to the total transmit power constraint

Ch. 9 – Error-Control Coding

- **Error-control coding**: At the transmitter, incorporate a fixed number of **redundant bits** into the structure of a **codeword**
- It is feasible to provide **reliable communication** over a noisy channel
  - Provided that **Shannon's code theorem** is satisfied
- In effect, **channel bandwidth** is traded off for **reliability** in communications.
- Another practical motivation for the use of coding is to **reduce the required E_b/N_0** for a fixed BER. This reduction in E_b/N_0 may, in turn, be exploited to
  - **Reduce** the **required transmitted power**
  - **Reduce** the **hardware costs** by requiring a **smaller antenna size** (antenna gain) in the case of **radio communications**

Ch. 10 – Spread-Spectrum Modulation

- **Spread-spectrum** modulation refers to any modulation scheme that produces a spectrum for the transmitted signal **much wider** than the bandwidth of the information being transmitted
- The **demodulation** must be accomplished, in part, by correlating **the received signal** with **a replica of the signal** that is used in the transmitter to spread the information signal



Preliminaries

Probability and Random Variables

Probability

- Probability: Axioms and Definitions
- Conditional Probability and Bayes' rule
- Random Variables
  - Cumulative distribution function (CDF)
  - Probability mass/density function (PMF/PDF)
- Multiple Random Variables
  - Joint CDF and Joint PMF/PDF
  - Conditional PMF/PDF and Marginal PMF/PDF
- Statistical Averages
  - Expectation
  - Moments and Central Moments
- Gaussian (Normal) Distribution

Fourier Theory and Signal Representation

Fourier Transform

- Fourier Series: Periodic signals
- Fourier Transform: Non-periodic deterministic signals
- Properties of Fourier Transform

Random Process, Autocorrelation and Power Spectral Density

Random Process

- Concept of Random Process
  - Stationary and Non-stationary Processes
- Autocorrelation Function
  - Stationary Processes
- Power Spectral Density
  - Definition (Fourier Transform of Autocorrelation Function)
  - Average Total Power

Sinusoidal Signal Representation

Sinusoidal Signal Representation

- Consider a general sinusoidal signal

$$s(t) = A(t)\cos\left[2\pi f_c t + \phi(t)\right]$$

  - $A(t)$ is the time-varying envelope
  - f_c is the carrier frequency
  - $\phi(t)$ is the time-varying phase
- The **time-varying phase** also implies that the **instantaneous frequency** is time-varying

$$\exp(j\theta) = \cos(\theta) + j\sin(\theta)$$

- The signal can also be represented as

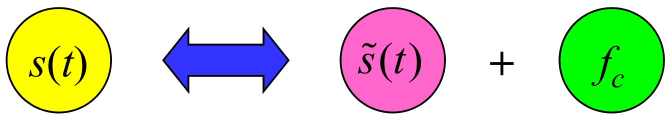
$$\begin{aligned} s(t) &= \mathrm{Re}\left\{A(t)\exp\left[j\left(2\pi f_c t + \phi(t)\right)\right]\right\} \\ &= \mathrm{Re}\left\{A(t)\exp\left[j\phi(t)\right]\exp\left[j2\pi f_c t\right]\right\} \\ &= \mathrm{Re}\left\{\tilde{s}(t)\exp\left[j2\pi f_c t\right]\right\} \end{aligned}$$

  - $\tilde{s}(t)$ is known as the **complex envelope** (a **low-pass** signal)

Sinusoidal Signal Representation (Cont.)

- For a specific **known** carrier frequency f_c, the signal $s(t)$ can be **completely** represented by the **complex envelope** $\tilde{s}(t)$

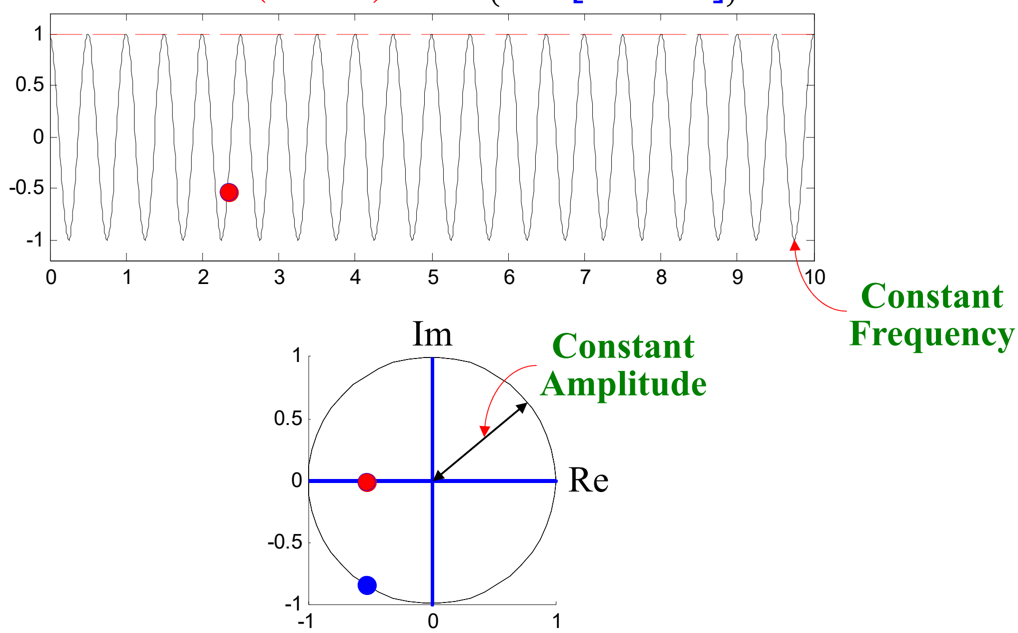
$$
\begin{aligned}
s(t) &= A(t)\cos\left(2\pi f_c t + \phi(t)\right) \\
&= \mathrm{Re}\left\{A(t)\exp\left[j\left(2\pi f_c t + \phi(t)\right)\right]\right\} \\
&= \mathrm{Re}\left\{\tilde{s}(t)\exp\left[j 2\pi f_c t\right]\right\}
\end{aligned}
$$



Constant Envelope and Zero Phase

- When the envelope $A(t)$ is constant ($A(t) = 1$) and $\phi(t) = 0$

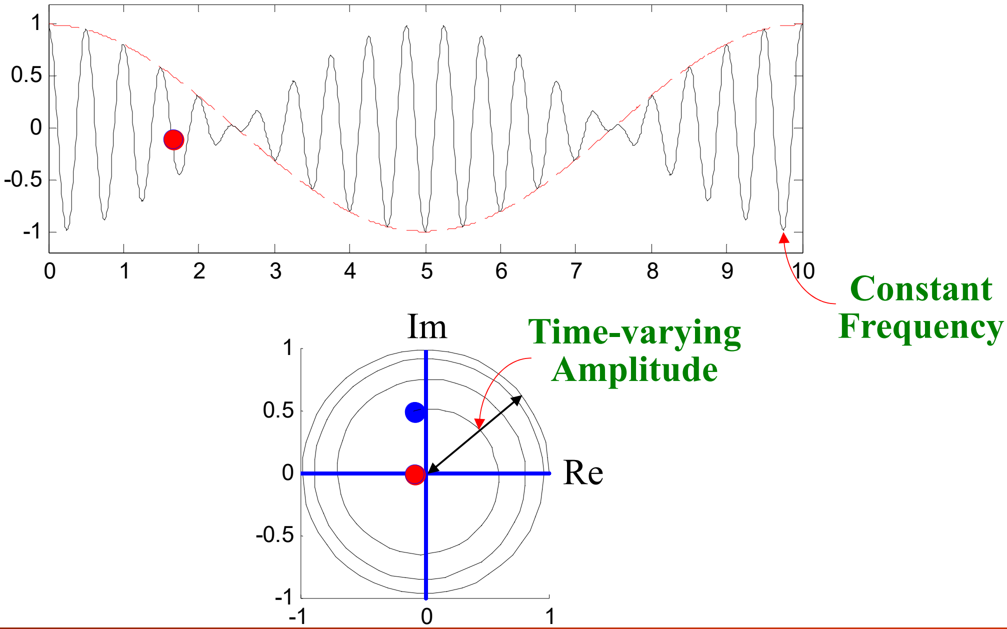
$$s(t) = \cos\left(2\pi f_c t\right) = \mathrm{Re}\left\{\exp\left[j2\pi f_c t\right]\right\}$$



Time-varying Envelope and Zero Phase

- When the envelope $A(t)$ is time-varying and $\phi(t) = 0$

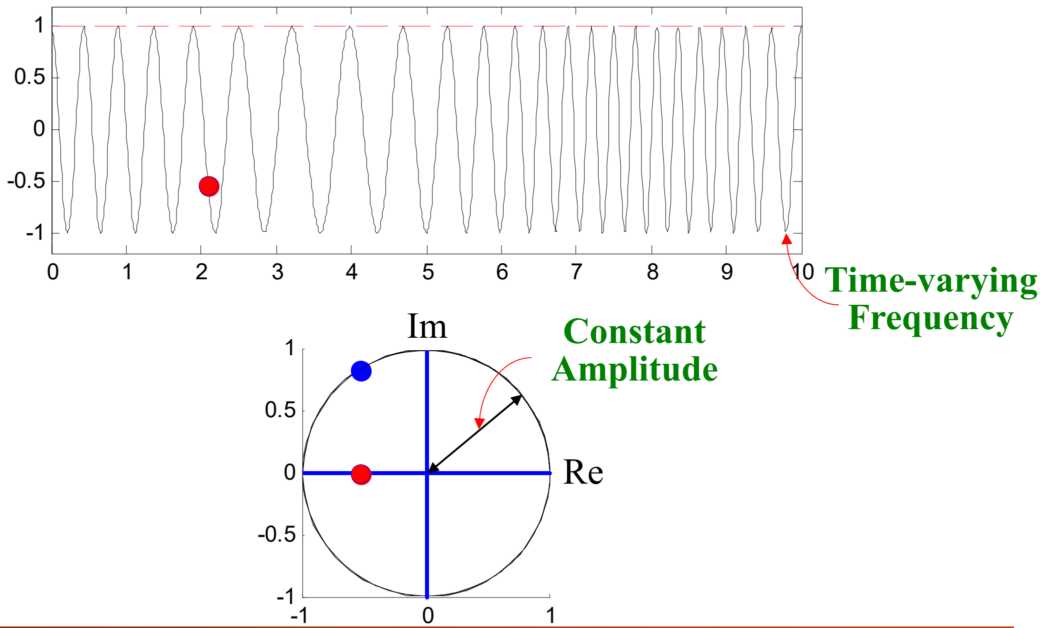
$$s(t) = \cos\left(2\pi f_a t\right)\cos\left(2\pi f_c t\right) = \mathrm{Re}\left\{\cos\left(2\pi f_a t\right)\exp\left[j2\pi f_c t\right]\right\}$$



Constant Envelope and Time-varying Phase

- When the envelope $A(t)$ is constant and $\phi(t)$ is time-varying

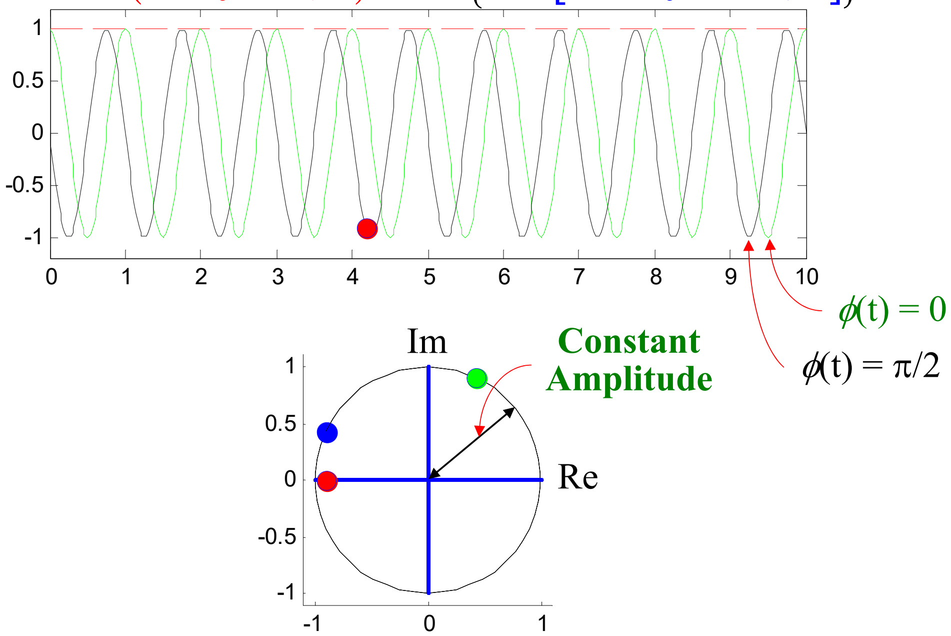
$$s(t) = \cos\left(2\pi f_c t + \phi(t)\right) = \mathrm{Re}\left\{\exp\left[j2\pi f_c t + j\phi(t)\right]\right\}$$



Constant Envelope and Constant Phase

- When both the envelope $A(t)$ and phase $\phi(t)$ (= $\pi/2$) are constant

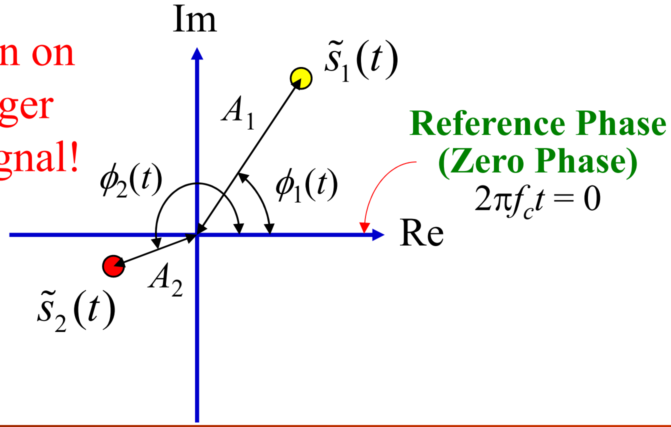
$$s(t) = \cos\left(2\pi f_c t + \pi/2\right) = \mathrm{Re}\left\{\exp\left[j2\pi f_c t + j\pi/2\right]\right\}$$



Constant Envelope and Constant Phase (Cont.)

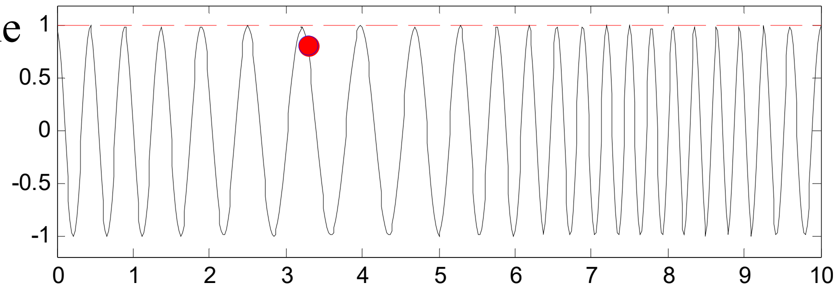
- In general, we can set $\mathbf{2\pi f_c t}$ as the **reference phase** (i.e., the zero phase) $s(t) = A\cos\left(2\pi f_c t + \phi\right)$
- Then, a signal with constant envelope A and constant phase ϕ can be represented as a complex number (a point in the complex-plane) $\tilde{s}(t) = A\exp\left[j\phi\right]$

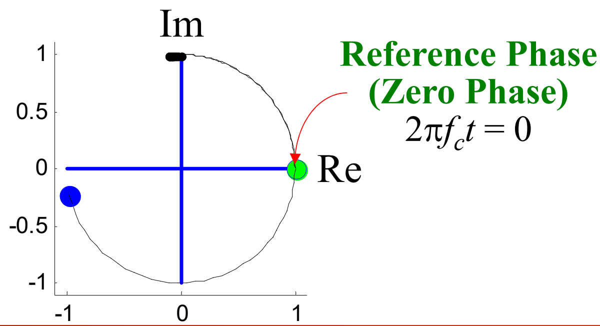
Note that the projection on the **Re** axis is no longer the amplitude of the signal!



Constant Envelope and Time-varying Phase

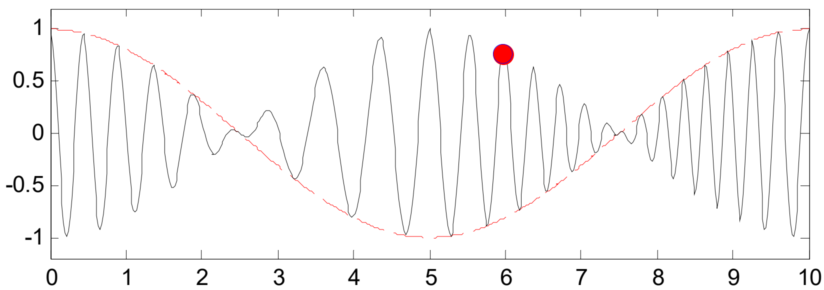
- Similarly, a signal with constant envelope $A(t)$ and time-varying phase $\phi(t)$ can be represented as a time-varying point in the complex-plane

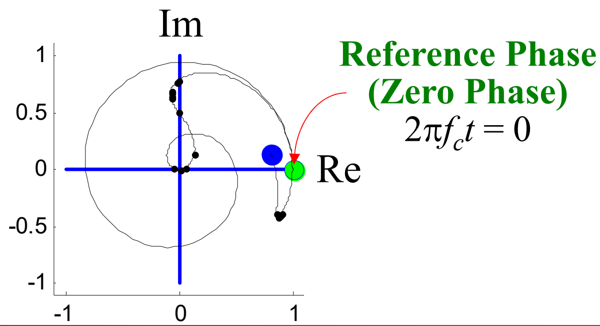




Time-varying Envelope and Phase

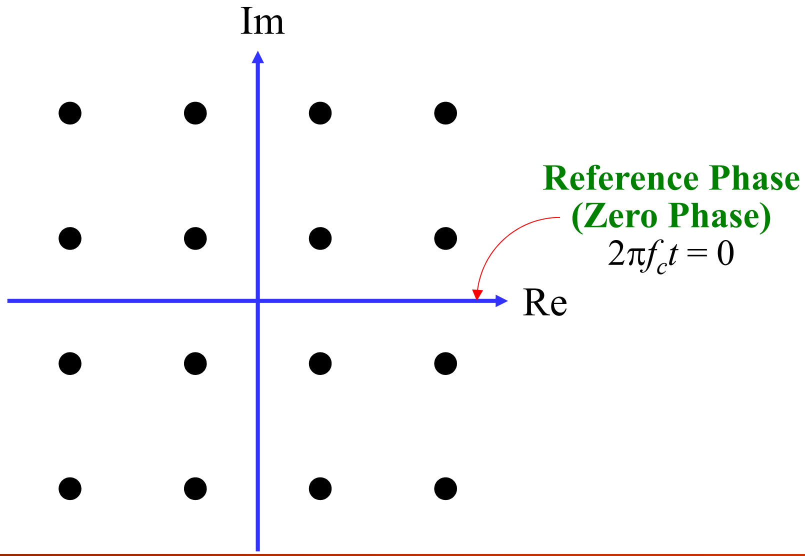
- Similarly, a signal with time-varying envelope $A(t)$ and phase $\phi(t)$ can be represented as a time-varying point in the complex-plane





Representation of 16QAM Signals

- QAM: a kind of **digital modulation** by using **different phases** and/or **different amplitudes** to represent different data
- 16QAM: 16 signal points (complex numbers) representing 4-bit data



Band-pass Signals

- A **band-pass signal** is sinusoidal with approximate frequency f_c and an amplitude varying with time

$$g(t) = a(t)\cos\left[2\pi f_c t + \phi(t)\right]$$

  – where $a(t)$ is the **envelope** and $\phi(t)$ is the **phase** of the signal
- The band-pass signal $g(t) = a(t)\cos\left[2\pi f_c t + \phi(t)\right]$ can be rewritten as

$$g(t) = \mathrm{Re}\left[\underline{a(t)\exp(j\phi(t))}\exp(j2\pi f_c t)\right]$$
$$= \mathrm{Re}\left[\tilde{g}(t)\exp(j2\pi f_c t)\right]$$

  – where $\tilde{g}(t)$ is referred to as the **complex envelope** of the band-pass signal (a **low-pass equivalent** signal)
- The complex envelope can be represented as

$$\tilde{g}(t) = g_I(t) + jg_Q(t)$$

  – where $g_I(t) = a(t)\cos\phi(t), \quad g_Q(t) = a(t)\sin\phi(t)$

dB Representation

Decibel (dB)

- The **decibel** (**dB**) is a relative unit used to express the **ratio** of one value to another on a **logarithmic scale**.
- It can be used to express an **absolute value**. In this case, it expresses the ratio of a value to a **fixed reference value**.
  - A **suffix** indicating the reference value is appended after dB
  - e.g., dBW, dBm, dBV
- The definition of dB is $X_{(\mathrm{dB})} = 10 \times \log_{10}(X)$
- The representation of dB can be used to expression a very large value or a very small (closed to 0) value
  - 100 dB = 10000000000
  - −100 dB = 0.0000000001
  - 2 = 3dB; 3 = 4.771dB; 5 = 7dB

Decibel (dB) (Cont.)

- Someone may question that there seems to be another expression of dB defined as

$$X_{\text{(dB)}} = 20 \times \log_{10}(X)$$

  - **No!** There is only one expression of dB:

$$X_{\text{(dB)}} = 10 \times \log_{10}(X)$$

- In electrical circuits, power dissipation is proportional to the square of voltage or current when the impedance is constant.
  - The power gain level (in dB) is expressed as

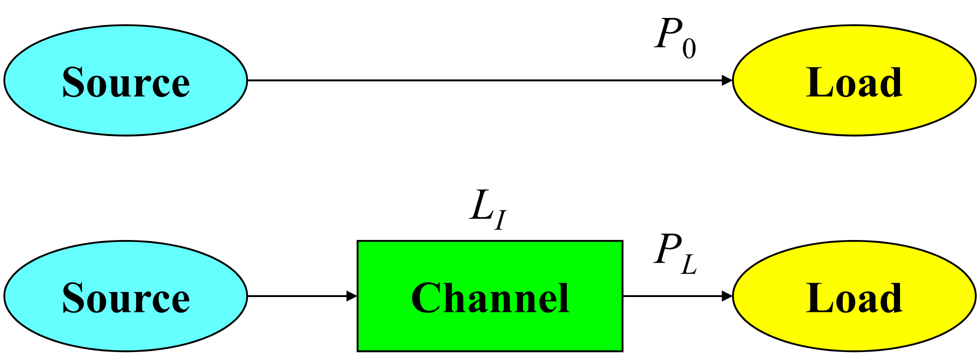
$$G_{\text{(dB)}} = 20 \times \log_{10}(V_{\text{out}}/V_{\text{in}})$$

- However, the expression is due to

$$G_{\text{(dB)}} = 10 \times \log_{10}(V_{\text{out}}^2/V_{\text{in}}^2) = 10 \times \log_{10}(V_{\text{out}}/V_{\text{in}})^2$$
$$= 20 \times \log_{10}(V_{\text{out}}/V_{\text{in}})$$

Insertion Loss (Path Loss)

- P_0: the power delivered to a load when it is connected directly to the source (linear scale, W, mW, …)
- P_L: the power delivered to a load from a source via a **channel**
- Insertion loss $L_I = 10 \log_{10} (P_0 / P_L)$ **dB**
- $x \leftrightarrow y$ dB $\Rightarrow y = 10 \log_{10} x$, e.g., 20 = 13 dB



Insertion Loss (Path Loss) (Cont.)

- For example, if the transmit power is $P_{(dB)} = 0$ dBW (i.e., 1W), the channel bandwidth is $B = 100$ KHz, and the noise power spectral density is $N_0/2$ with $N_0 = -110$ dBW/Hz.
- If the propagation loss of the channel is $L = 40$ dB, the received signal power is

$$P_{r(dB)} = 10 \times \log_{10}(P/L) = P_{(dB)} - L_{(dB)} = -40 \text{ dBW}$$

- The symbol energy is

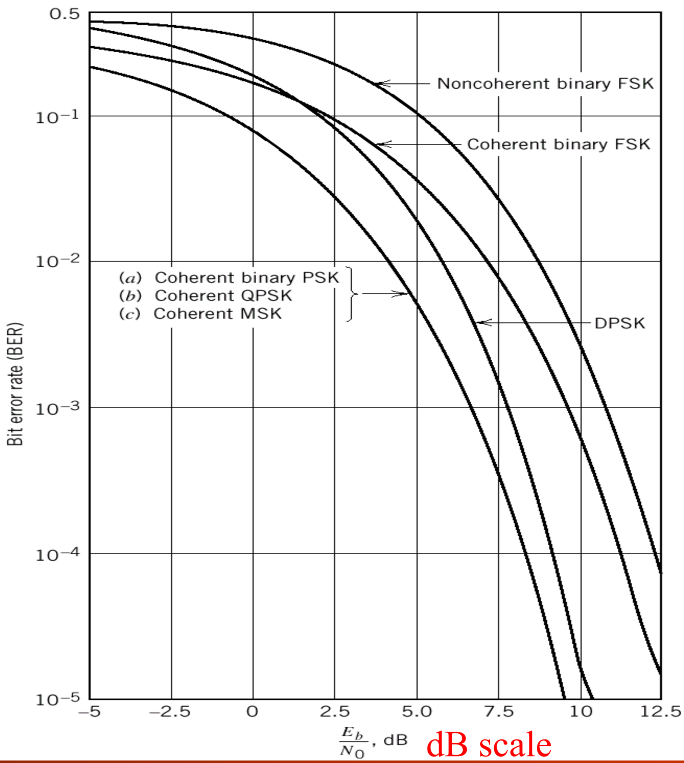
$$E_{(dB)} = 10 \times \log_{10}(P_r \times T) = 10 \times \log_{10}(P_r/B)$$
$$= P_{r(dB)} - B_{(dB)} = -40 - 50 = -90 \text{ dB Joules (W/Hz)}$$

- The received signal SNR is

$$(E/N_0)_{(dB)} = E_{(dB)} - N_{0(dB)} = -90 - (-110) = 20 \text{ dB}$$

Representation of Probability of Error

dB scale



dB scale