~~3~~ 4 Binary Phase-Shift Keying (BPSK)

q. 1(a)

~~*BPSK is perhaps the simplest modulation scheme in common use and we shall analyse it in some detail. The results for other schemes can often be derived from those for BPSK.*~~

~~3.1~~ 4.1 Definition of BPSK:

Let k^{th} data bit $b_k = +1$ or -1 and bit period $= T_b$.

The modulated phasor during the k^{th} bit period is:

$$p_k(t) = b_k\ a_0 e^{j\phi_0} \qquad \text{for } kT_b \le t < (k+1)T_b$$

To apply this for all t, we introduce a time limited pulse:

$$g(t) = \begin{cases} a_0 & \text{for } 0 \le t < T_b \\ 0 & \text{elsewhere.} \end{cases}$$

$$\therefore\ p(t) = e^{j\phi_0} \sum_k b_k g(t - kT_b) \tag{3.1}$$

Note that $g(t)$ is normally a rectangular pulse, but modified forms of BPSK can use other shapes such as half-sine or raised cosine.

~~3.2~~ Power Spectrum for Random Data:

We observe that $p(t)$ is just a constant phasor $e^{j\phi_0}$ multiplied by a polar binary data stream, in which the data impulses have been filtered (convolved) with an impulse response $g(t)$.

Hence we use techniques similar to those in the 3rd-year E5 Baseband Transmission course.

The discrete autocorrelation function (ACF) of the random data stream b_k is:

$$R_{bb}(L) = E\{b_k\ b_{k-L}\} = \begin{cases} 1 & \text{for } L = 0 \\ 0 & \text{elsewhere.} \end{cases}$$

$\therefore$ the autocorrelation function of the stream of data impulses $b(t) = \sum_k b_k \delta(t - kT_b)$ is:

$$C_{bb}(\tau) = \frac{1}{T_b} \sum_L R_{bb}(L) \delta(\tau - LT_b) = \frac{1}{T_b} \delta(\tau)$$

and its power spectrum is the Fourier Transform of this (Wiener-Kintchine Theorem).

$$\therefore\ |B(\omega)|^2 = \int_{-\infty}^{\infty} C_{bb}(\tau) e^{-j\omega\tau} d\tau = \frac{1}{T_b} \int_{-\infty}^{\infty} \delta(\tau) e^{-j\omega\tau} d\tau = \frac{1}{T_b}$$

Since the data impulses $b(t) = \sum_k b_k \delta(t - kT_b)$ are convolved with $g(t)$ in equation 3.1, the Fourier transform of $p(t)$ is given by:

$$P(\omega) = e^{j\phi_0} B(\omega)\ G(\omega)$$

1. (a) (cont.)

$$|P(\omega)|^2 = |B(\omega)|^2 \, |G(\omega)|^2 = \frac{1}{T_b} \, |G(\omega)|^2$$

$$= a_0^2 \, T_b \, \mathrm{sinc}^2\left(\frac{\omega T_b}{2}\right) \quad \text{(from data book)}$$

if $g(t)$ is a rect pulse of ampl. a_0 & duration T_b.

If $f = \frac{1.58 R_b}{2}$, $\omega T_b = 2\pi \cdot \frac{1.58}{2} = \pi \cdot 1.58$

& $\mathrm{sinc}^2\left(\frac{1.58\pi}{2}\right) = 0.0610 = -12.1 \text{ dB}$ (as a power ratio)

Hence a bandwidth of $1.58 R_b$ includes all components down to -12 dB ($\underbrace{\mathrm{sinc}^2(0)}_{=1}$ is the max value & the sidelobes of $\mathrm{sinc}^2(\cdot)$ are more than 12 dB below the main peak.

1. (b) TDMA - Low cost, circuitry may be shared by many signals, allows high bit-rate burst mode. BUT requires central timing sync for ~~base~~ mobile → base station working.

SSMA - Higher cost, but users need not cooperate at all. Less efficient use of spectrum than TDMA. Provides anti-jam and low probability of intercept as additional advantages.

For DS SSMA, coding gain = $\frac{\text{Code rate}}{\text{Message bit rate}}$

Coding gain represents the power gain of a wanted signal (whose code is known at the receiver) over jamming signals and noise, ~~(whose code is not~~ which become spread to the full code bandwidth at the receiver, while the wanted signal is despread back to the ~~n~~ narrow bandwidth corresponding to the message modulation.

1. (c) For TDMA,

channel BW = 300 MHz

For 12 dB bandwidth of $1.58\,R_b$

$$R_b = \frac{300.10^6}{1.58}$$

FEC coding will reduce the user rate to $\frac{2}{3}R_b$.

There is probably approx 5% overhead for sync data for the TDMA multiplexer(s).

$$\text{Hence max no. of users at 56 kb/s} = \frac{\frac{2}{3}\times\frac{300.10^6}{1.58}}{1.05\times 56.10^3}$$

$$\simeq \underline{\underline{2150}}$$

For DS SSMA

$$\text{Code } \cancel{\text{rat}} \text{ chip rate} = \frac{300.10^6}{1.58} = 189.9\text{ MHz} .= R_c$$

[Coding gain for a user at 56 kb/s with rate 2:3 ECC

$$= \frac{189.9.10^6}{3/2\,\cancel{2/3} \times 56.10^3} = 2260$$]

Assuming all users are at equal power at the satellite (path losses likely to be similar for all paths via satellite), interference power $= (n_u - 1)$(wanted signal power)

$$= (n_u - 1).\, E_b .\, 56.10^3 .\, \underset{\uparrow\ \text{ECC code rate}}{\frac{3}{2}}$$

$$N_0 = \frac{\text{interference power}}{R_c} = \frac{(n_u-1)\,E_b .\, 56.10^3 .\, \frac{3}{2}}{189.9.10^6}$$

$$\therefore (n_u - 1) = \frac{189.9.10^6}{\frac{3}{2}.\,56.10^3} . \frac{N_0}{E_b} = 2260/10^{0.4} = 899.$$

$$\therefore n_u = \underline{\underline{900}}$$

1 (d) ~~B~~ QPSK doubles the allowed bit in a given bandwidth compared with BPSK.

Hence for TDMA, $R_b = 2 \times \dfrac{300.10^6}{1.58}$

and the max no. of users would double to ~ 4300.

For DS-SSMA, the code chip rate would double but the power spectrum of the spread signal would remain unchanged & hence ~~the~~ the interference noise PSD per unwanted signal ~~to~~ would not change.

The E_b/N_o required for a QPSK demodulator is the same as that for a BPSK modulator, for a given bit error rate, so the number of users will remain unchanged at ~900.

2 (a) Multi-level modulation transmits several bits per transmitted symbol & is therefore more spectrally efficient than binary methods (bandwidth is usually proportional to symbol rate).
∴ Multi-level gives more bits/sec per Hz of ~~bandw~~ bandwidth.

BUT Multi-level places modulation phasors much closer together and so is more prone to errors in the presence of noise.

QAM distributes phasors uniformly in a 2-D plane, whereas MPSK can only distribute them around the 1-D perimeter of the unit circle.
Hence QAM achieves a wider spacing of phasors for a given no. of levels M than MPSK, & hence achieves better error-rate performance than MPSK – especially if M is large.

2. (b)

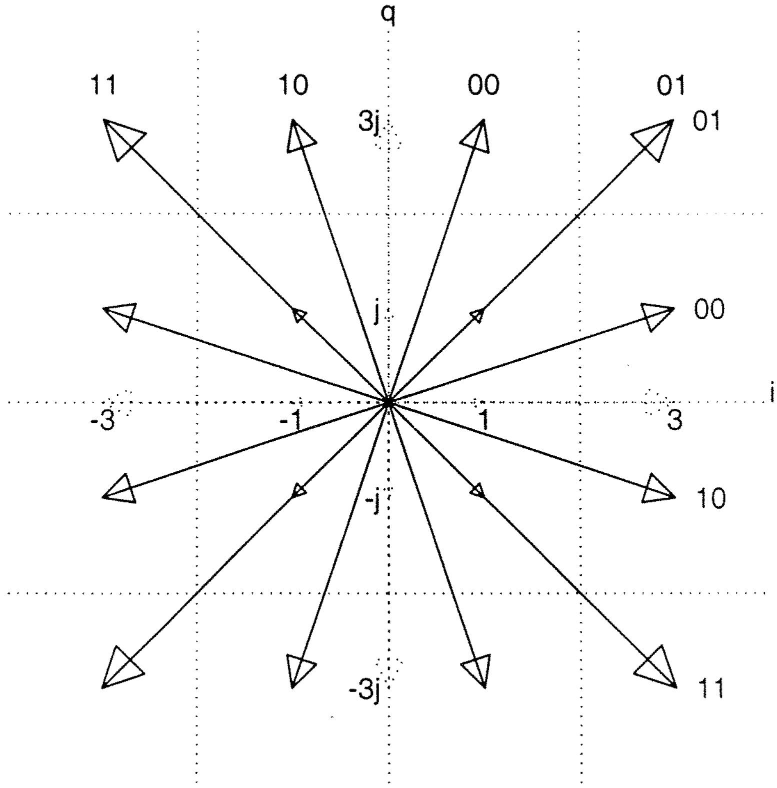

Fig 5.2: 16-QAM phasor diagram

2-bit Gray code used for i & q components to minimise BER due to each symbol error.

2 (c) Prob. of noise causing a boundary crossing $= Q\left(\sqrt{\frac{3E_s}{(M^2-1)N_0}}\right) = P_{bc}$

If there are m bits coded <u>on each component</u> of the M^2 QAM phasor, $M = 2^m$ & $M^2 = 2^{2m}$

~~For 16 QAM~~

Since there are $2m$ bits per symbol, $\underline{\underline{E_s = 2m\,E_b}}$

Considering detection of the horizontal component of the QAM phasor:

$$\text{Prob. of symbol error} = \frac{P_{bc} + (M-2).2P_{bc} + P_{bc}}{M}$$

$$= 2P_{bc}\left(\frac{M-1}{M}\right)$$

since the $M-2$ inner states have 2 boundaries (& hence 2 chances of error) while the 2 outer states have just 1 boundary.

If Gray Coding is used:

$$\text{Prob. of bit error} = \frac{\text{Prob of symbol error}}{m}$$

$$= \frac{2}{m} P_{bc}\left(\frac{M-1}{M}\right)$$

$$\underline{\underline{= \frac{2}{m}\left(\frac{M-1}{M}\right).\,Q\left(\sqrt{\frac{6m\,E_b}{(M^2-1)\,N_0}}\right)}}$$

2(d) For 16-QAM, $M^2 = 16$ & $2m = 4$

For 256-QAM, $M^2 = 256$ & $2m = 8$

∴ For a given symbol rate (or bandwidth) 256-QAM can transmit <u>twice</u> the bit-rate of 16-QAM. <u>OR</u> for a given bit rate, only <u>half the bandwidth</u> is needed by 256-QAM.

For 16-QAM, $\frac{2}{m}\left(\frac{M-1}{M}\right) = \frac{2}{2}\left(\frac{4-1}{4}\right) = \frac{3}{4}$

For 256-QAM, $\quad '' \quad = \frac{2}{4}\left(\frac{16-1}{16}\right) = \frac{15}{32}$

If BER $= 10^{-3}$, $P_{bc} = 10^{-3}/\tfrac{3}{4} = 1.333.10^{-3}$ for 16 QAM

$P_{bc} = 10^{-3}/\tfrac{15}{32} = 2.133.10^{-3}$ for 256-QAM

By trial & error $x \doteq 3.00$ if $Q(x) = 1.333.10^{-3}$

& $x = 2.86$ if $Q(x) = 2.133.10^{-3}$

∴ For 16-QAM: $\frac{E_b}{N_0} = \frac{M^2-1}{6m} \cdot x^2 = \frac{16-1}{12} \cdot (3.00)^2 = 11.25$

$= 10.5$ dB

For 256-QAM: $\frac{E_b}{N_0} = \frac{256-1}{24} \cdot (2.86)^2 = 86.9$

$= \underline{19.4 \text{ dB}}$

Hence 256-QAM requires <u>8.9 dB</u> better SNR (E_b/N_0) to achieve 10^{-3} BER.

Q3 4F5 Digital Communications 2003

You are required to design an ultrasonic indoor location system.

(a) Outline the design of a system which could provide three-dimensional accuracy of up to 3 centimeters in each dimension most of the time [40%]

(b) Comment on techniques for maximizing battery life of the active tags [20%]

(c) How could the location quality of service (LQoS) be improved for those situations where a human in the loop is performing a control function [20%]

(d) How does the accuracy of an ultrasonic system compare to one based on ultrawideband radio [20%]

(a)
Use ultrasonic transmissions from tag and multiple ceiling receivers
Multiple times of flight used to find solution
If solution above ceiling can be ignored need 3 times of flight
Time measurement needs to be at about the 100 microsecond level
Shaping outgoing pulse can help with reflections and time measurement
Receivers have to be surveyed to calibrate the system
Filter outlying solutions

Need to control transmissions to avoid collisions
Random re-transmit interval is possible and simple
Better system control achieved by using radio to nominate a single tag
Hence need radio receiver in each tag (potentially wastes energy)
Need on-air protocol with at least an address field
Could have a master controller or a distributed algorithm
Bluetooth could be one approach to the radio side

(b)
Always on radio receiver is biggest user of energy
Radio receiver switched off as much as possible
But radio receiver left on if high LQoS is required
Also radio receiver switched off tag in deep sleep mode
Very low power RC time out to bring tag out of deep sleep mode
When awaking transmit on radio to announce arrival
Hence need radio transmitter as well as receiver

Maybe a movement detector in tag can trigger transmissions
Transmissions must only occur when required by applications

(c) Server can nominate tags on moving devices more often
Server can nominate other devices nearby to moving devices
Server can take into account state of battery at tag
Server can reduce the number of location requests and extrapolate
Server can build profiles for LQoS by application type and state
Tags not in use by applications are left in deep sleep mode

(d) Ultrawideband radio uses short pulses of about 100 psec duration
A large spectrum is used, for example 3GHz-10GHz
Radio waves travel at about 1 nsec / 30 cms
Accuracy depends on clock (local) synchronisation
In principle precision should be similar to bat
Few multipath problems compared to ultrasonic approach
However unclear what the local effect of metallic objects might be

4

We are interested in supporting mobility in an IP environment.

(a) What is IPv4's most fundamental obstacle to transparent mobile networking? [10%]

(b) Mobile IP has been developed to address this obstacle.

(i) Give a brief overview of the Mobile IP approach, using the correct jargon ("mobile node", "home address", "home agent", "foreign network" etc). [20%]

(ii) What are the functions of the foreign agent? [10%]

(iii) The three principal mechanisms of Mobile IP are: discovering *X*, registering *X* and tunnelling to *X*. After stating what *X* stands for in these definitions, describe these three mechanisms in detail. [30%]

(c) The base Mobile IP specification is affected by routing inefficiencies.

(i) Explain the problem of triangle routing. [10%]

(ii) Describe a solution to it. [10%]

(iii) What are the drawbacks, if any, of this solution? [10%]

Crib

4 ~~1~~ (a)

In IPv4, each node is identified by an address. The network to which the node is connected is obtained by masking off the least significant bits (host portion) of the node's address. Therefore, if the node changes network, it cannot retain the same address. Changing address usually breaks the upper-layer connection (it does for TCP, which accounts for most of the traffic), so transparent mobility is not possible.

4 ~~1~~ (b) (i)

In Mobile IP, each mobile node is assigned two addresses: the home address, which is permanent, and the care-of address, which varies dynamically as the node visits any foreign networks. Nodes wishing to contact the mobile node always write to the home address. There, the home agent passes the packet along to the mobile node, wherever it happens to be.

Whenever the mobile node visits a foreign network, it receives a new care-of address from the network's foreign agent. The home agent must be notified of this in order for it to know where to forward the incoming packets meant for the mobile node.

4 ~~1~~ (b) (ii)

Advertise the availability of the Mobile IP service in that network (seen as a foreign network by potential customers).

Assign care-of addresses to visiting mobile nodes.

Untunnel messages from the home agent and pass them to the visiting mobile node.

4 ~~1~~ (b) (iii)

X is the care-of address.

Discovering the care-of address:

The mobile node visits the foreign network and listens for an "agent advertisement" broadcast from the local foreign agent. (If this isn't forthcoming, the impatient visiting mobile node can request it by broadcasting an "agent solicitation".) The agent advertisement will contain one or more care-of addresses. The mobile node grabs one.

Registering the care-of address:

The mobile node sends its newly-acquired care-of address to the home agent. This constitutes a "binding update", i.e. a request that the home agent update its routing table with a new "binding", which is a triple containing home address, care-of address and registration lifetime. Since this mechanism could be used to steal traffic meant for the mobile node, the mobile node must authenticate itself to the home agent as part of the binding update request. The home agent sends a reply after performing the update.

Tunnelling to the care-of address:

When a correspondent node sends a packet to the mobile node at the home address, the home agent tunnels this packet to the care-of address. This is done by encapsulating the packet in the payload of a new packet sent by the home agent to the care-of address. The foreign agent receives this packet and untunnels it, i.e. removes the outer (tunnel) header and delivers the inner packet to the mobile node identified by the care-of address.

4 (c) (i)

The name "triangle routing" comes from the shape of the round-trip path taken by messages between a fixed and a mobile node. When a fixed correspondent node writes to a mobile node, the IP packet first goes to the home agent (first side of the triangle) and then (second side) to the mobile node. When the mobile node replies, instead, the packet travels directly (third side) from the mobile node to the correspondent node. There is an obvious inefficiency in having to take a detour via the home agent, particularly if the correspondent node and the mobile node are much closer to each other than they are to the home agent.

4 (c) (ii)

The mobile node could send a binding update directly to the correspondent node, like it does with the home agent.

4 (c) (iii)

Unlike the home agent, the correspondent node wouldn't normally keep any bindings: it just uses the raw addresses directly. We must modify the IP code of each correspondent node before we can send it a binding update—whereas the triangle routing solution does not require modifications to correspondent nodes. This makes this solution unsuitable for IPv4 where the deployed base is too large, but acceptable for IPv6 while the specification is not yet frozen.