Example solutions EEE6430 / 443 (2008)

1. a.

Refer to Fig. 1.1. Data streams *A* and *B* are spread by a factor of *6* by XORing with Spreading Code's *A* and *B* respectively to produce Sig *A*+*B*, which modulates a carrier and is transmitted as *ASK*. At the receiver Sig *A*+*B* is demodulated, and then multiplied by Spreading Code *B* (*A*) to recover Data *B* (*A*). The spreading codes associated with the data streams are therefore known at the receiver.

1. b.

Chip rate $R_c = 3.84Mcps$

Data rate $R_d = 12.2kbps$

Processing Gain

$$G_p = 10\,log_{10}(\frac{R_c}{R_d}) = 25dB \qquad (1.1)$$

If after de-spreading E_b is the power density per user bit, and N_o is the interference and noise power density, then for a speech service this ratio $10\,log_{10}(E_b / N_o) \approx 5dB$, which means that the signal power can be up to *20dB* under the interference or thermal noise level, and the *WCDMA* receiver can still detect the signal. For *GSM* the signal level needs to be significantly higher than this.

1. c.

The downlink *DPCH* is *QPSK* modulated and therefore the bit rate is twice the symbol (or baud) rate. For a spreading factor $S_f = 128$ then the symbol rate is

$$R_s = \frac{R_c}{S_f} = \frac{3.84 \times 10^6}{128} = 30kbps \qquad (1.2)$$

and the channel bit rate is

$$R_b = 2 \times R_s = 60kbps \qquad (1.3)$$

Now the *DPDCH* shares the downlink *DPCH* frame with control data. So the actual *DPDCH* data rate depends on the amount of control data within a frame. There will also be some error protection coding which further reduces the *DPDCH* data. Thus, as a pure estimate *(all that's asked for in question)*, assuming the *DPDCH* occupies *50%* of a frame, and further assuming half rate coding, then the user data rate is

$$R_d \approx 0.5 \times 0.5 \times 60 \approx 15kbps \qquad (1.4)$$

The uplink *DPDCH* is transmitted as straight *PSK* and therefore the bit rate equals the symbol rate. It also occupies an entire frame, and thus again assuming half rate coding

$$R_d \approx 1 \times 0.5 \times 30 \approx 15kbps \qquad (1.5)$$

1. d.

The lowest spreading factor is *4*, giving a symbol rate of *960kbps*. Considering the downlink therefore, using *3* parallel codes yields a *DPCH* bit rate of *5.76kbps*. Allowing for some (minimum) control data and assuming half rate coding gives a maximum user data rate of $\sim 2.3Mbps$. In the uplink direction, *6* parallel codes would be required.

2. a.
The gain of N elements over a single element is given by

$$G = 10\log_{10}\left(\frac{P_N(90^o)}{P_1(90^o)}\right)^2 \ dBd \qquad (2.1)$$

but the same power has to be delivered to both the N element and single element arrays. If I is the current through a single element with input resistance R, then for the same power to be delivered to the array

$$I^2 R = (NI_o)^2 \frac{R}{N} \qquad (2.2)$$

hence

$$I = I_o\sqrt{N}$$

From the given array factor

$$P_N(90^o) = I_o N \qquad (2.3)$$

and the gain is therefore

$$G = 10\log_{10}\left(\frac{I_o N}{I}\right)^2 = 10\log_{10}\left(\frac{I_o N}{I_o\sqrt{N}}\right)^2 = 10\log_{10}(N) \ dBd \qquad (2.4)$$

The gain above isotropic is

$$G = 10\log_{10}(N) + 2.15 \ dBi \qquad (2.5)$$

Thus for *11* elements,

$$G = 10\log_{10}(11) + 2.15 = 12.56 dBi \qquad (2.6)$$

Position of first side lobe is given by

$$\frac{kNd}{2}\cos(\theta) = 0.8N\pi\cos(\theta) = \pm\frac{3\pi}{2} \qquad (2.7)$$

so

$$\theta = \cos^{-1}\left(\pm\frac{3}{1.6N}\right) = 80.2^o \ \text{or} \ 99.8^o \qquad (2.8)$$

Height of first side lobe with respect to main lobe is

$$20\log_{10}\left(\frac{\sin(3\pi/2)}{\sin(3\pi/2N)}/N\right) = -13.2db \qquad (2.9)$$

2. b.
(i) A mobile phone user may vary the orientation of the handset during a call, and also signal propagation between the handset and *BTS* may be subject to scattering off buildings etc. Both these factors could alter the polarization of the signal, and therefore the signal level can potentially be improved by having the ability to switch between orthogonal polarizations at the *BTS*, thus providing polarization diversity gain.
(ii) Multi-path propagation could cause fading nulls at a *BTS* antenna, but a second receive antenna a few wavelengths away would probably not experience the same null at the same time. Thus the ability to switch between two such antennas can increase the received signal strength and provide space diversity gain. Alternatively direct and

reflected signals coming in from different directions and received by different antenna ‘beams’ could be combined in a rake receiver to increase the signal strength.
(iii) Multipath reflected signals arriving at the *BTS* antenna at different times could be delay compensated and equalised and summed in a rake receiver to provide time diversity gain.

2. c.

(i) The Handset User (HU) would probably keep altering the orientation of the handset, thereby affecting the signal polarization.

(ii) The HU’s head may come between the phone and *BTS* (if he changed hands for instance), causing absorption of some of the signal in head tissue.

(iii) Although the azimuth beamwidth of a typical *BTS* ‘plank’ antenna is quite broad, some signal variation may be due to the HU walking in/out of the main lobe.

(iv) It is possible the network would instruct the phone to alter its transmit power according to the received signal strength at the *BTS*.

(v) Since this is a tri-cell *BTS*, the HU’s phone would probably be instructed to hand over to the adjacent cell at 120^o sector intervals, thereby causing a frequency change.

2. d.

The *BTS Test* engineering option keeps the handset on the same frequency channel, so that it will not change cells according to the strongest received signal. Hence, if this were invoked there would be no frequency changes, and therefore the signal strength variation would be more extreme as the HU walked out of the sector served by this channel.

3. a.
(i) **Q**uadrature **P**hase **S**hift **K**eying (**QPSK**) involves changing the carrier phase between four states separated by 90^o. Each of the four possible carrier phases therefore represents two bits, *00*, *01*, *10*, *11*, and therefore the bit rate is double the symbol rate. The bit pairs can change in any sequence, and those changes corresponding to a complete phase reversal of the carrier can cause undesirable amplitude modulation effects which may increase the signal bandwidth, i.e. $00 \leftrightarrow 11$, or $01 \leftrightarrow 10$. *QPSK* is used to transmit the *WCDMA* protocol, and bits on the *I* and *Q* channels are phase shifted to prevent them changing simultaneously.

(ii) **F**requency **S**hift **K**eying (**FSK**) involves sending different binary data bits as separate frequencies. Here the bit rate equals the symbol rate.

(iii) **M**inimum **S**hift **K**eying (**MSK**) is used to reduce amplitude variations and hence the bandwidth of an *FSK* signal due to jumps in phase at frequency changes. With *MSK* there is phase continuity between the two frequencies at bit transitions.

(iv) **G**aussian **M**inimum **S**hift **K**eying (**GMSK**) is used in *GSM*, where the base band keying waveform is put through a Gaussian filter. Instead of the oscillator suddenly changing between frequencies therefore, the change is more gradual depending on the characteristics of the Gaussian filter. This further reduces the overall signal bandwidth.

The *bandwidth efficiency* can be defined as $\frac{R}{B}$, where *R* is the bit rate and *B* the required channel bandwidth. This can be expressed as $\frac{1}{BT}$, since the bit period $T = \frac{1}{R}$. In *GSM*, the bit period is $T = 3.69\mu s$ and the *BT* product is *0.3*, defining the bandwidth of the Gaussian filter as $B = 81.3kHz$. Hence the bandwidth efficiency is

$$\frac{1}{BT} = 3.33bps / Hz \qquad (3.1)$$

3. b.
A mobile handset 'bursts' at *4.6ms* intervals. This causes spectral components at $1/(4.6\times10^{-3}) = 217.4Hz$ intervals. Thus, the signal spectrum will be

$$S = \sum_{i=-N}^{N} A_i \cos(2\pi(f + i\times217)t) \qquad (3.2)$$

where *f* is the carrier frequency. Now if this signal induces a voltage in a non-linear medium, such as the output stages of an audio amplifier for instance, then currents will be driven according to S^2, S^3 etc., producing inter-modulation terms such as

$$\cos(2\pi ft)\times\left[\cos(2\pi(f+217)t) + ... + \cos(2\pi(f + N\times217)t)\right] + ... \qquad (3.3)$$

which can be re-written as

$$\frac{1}{2}\begin{bmatrix} cos(2\pi(2f+217)t)+cos(2\pi\times 217t)+... \\ +cos(2\pi(2f+N\times 217)t)+cos(2\pi\times N\times 217t) \end{bmatrix} \quad (3.4)$$

Thus there will be frequency components at $i\times 217\,Hz$ intervals, well within the bandwidth of the audio amplifier, hence the buzzing.

A possible health implication of this is that head/brain tissue is also non-linear, and therefore such sustained low frequency currents may also be induced therein. There has been much debate regarding the health effects of *50Hz* fields, and *217Hz* is not far removed. Humans have only been exposed to analogue handsets or walkie-talkies prior to the advent of *GSM*, which do not cause sustained low frequency inter-modulation products.

3. c.
The reason the uplink is modulated differently to the downlink in *3G* systems, is to ameliorate the problems caused by such on/off handset bursting. Hence the uplink *DPDCH* is carried on the *I* axis and the *DPCCH* on the *Q* axis of the *QPSK* modulation. Thus, during periods of *DTX*, when there is no speech, carrier is maintained via the control *Q* channel. Further, there is no *TDMA* type time slot scheduling to cause regular pulsing of the transmitter. So no buzzing.

4. a.

Refer to Fig. 4.1. In a full rate traffic channel, the speech data is transmitted in the same time slot in consecutive frames, with an *SACCH* control burst being substituted at the centre of the multi-frame. For half rate speech, two traffic channels are interleaved and transmitted in alternate frames, and there are two *SACCH*'s, one for each traffic channel.

4. b.

The *20ms* speech block is coded onto the traffic channel as *456* bits. The bit rate is therefore

$$R_b = \frac{1}{20 \times 10^{-3}} \times 456 = 22.8 kbps \qquad (4.1)$$

The speech data rate after compression by the codec is *13kbps*, and the increase in transmitted data payload is due to error protection bits being added. The *456* bits are sent over eight traffic channel time slot bursts in the following manner:

Bits 0,8…448 comprise the ***even*** *bits of burst N*
Bits 1,9…449 comprise the ***even*** *bits of burst N+1*
Bits 2,10…450 comprise the ***even*** *bits of burst N+2*
Bits 3,11…451 comprise the ***even*** *bits of burst N+3*
Bits 4,12…452 comprise the ***odd*** *bits of burst N+4*
Bits 5,13…453 comprise the ***odd*** *bits of burst N+5*
Bits 6,14…454 comprise the ***odd*** *bits of burst N+6*
Bits 7,15…455 comprise the ***odd*** *bits of burst N+7*

Thus *57* bits of a particular speech block appear in each burst, and these are interleaved with *57* bits from another block. If the *456* bits were just sent in *4* consecutive frames, then if noise knocked out a frame *25%* of consecutive data would be lost, making recovery difficult. But this way, only *12.5%* of data from a particular speech block is lost, and additionally it is not consecutive data, so the chance of recovery is improved.

4. c.

Circuit **S**witched **D**ata (**CSD**) involves sending data synchronously in a particular time slot. There is little scope for dynamic resource allocation, and the link remains at full capacity irrespective of the amount of data being transmitted. Speech data is essentially just replaced with computer data in a standard 'phone call' to a modem. Data rates depend on compression protocol being used but are typically *~9kbps*.

High **S**peed **C**ircuit **S**witched **D**ata (**HSCSD**) does the same thing but with more time slots, so the data rate is higher, typically up to *~28kbps*.

General **P**acket **R**adio **S**ystem (**GPRS**) uses internet ip-type packet scheduling to dynamically load data onto available time slots, and data rates are typically up to *~50kbps*. *GPRS* simply adds a packet switched network to the existing circuit switched core network.

Enhanced **G**eneral **P**acket **R**adio **S**ystem (**EGPRS**) upgrades the air interface while using the same GPRS packet switched core network. It does this by using 8-level

Phase **S**hift **K**eying (8**PSK**) instead of **F**requency **S**hift **K**eying, thereby increasing the bit rate by a factor of three.

4. d.

Voice **o**ver **IP (VoIP**) technology sends speech as internet type data. Thus a mobile running a *VoIP* application (such as Fring or Skype) could make internet based calls by sending speech as packet data through a *GPRS* connection for example.