# TSKS03 Wireless Systems

## Solutions for the exam 2013-01-07

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**1**
The available channel is split up in time and frequency, into slots. Each such slot can be allocated to a user. The basic idea is that the user that needs access to the channel, and has the best channel in that particular time-frequency slot gets that slot. This is based on an estimation of the SNR for each user and each slot.

*Comments not needed for getting full credit:* This aims at maximizing the total throughput per cell. Effectively, this also means that users near the edge of the cell essentially never would get access to the channel. The result is that the cell radius shrinks. To counteract this effect, the scheduling can be done slightly different, where each connected user is given at least some minimum part of the channel, which then slightly reduces the resulting total throughput.

**2**
A data fragment is protected by a CRC code. If the CRC check is not OK, then a retransmission is needed. If the channel is noisy, than the risk of such a retransmission is relatively high. In that case, the whole fragment has to be retransmitted even if only a few bits are erroneous, since the CRC check does not reveal the error locations. By sending data in smaller fragments, such retransmissions will be shorter, and the overall fraction of data that has to be retransmited may be reduced.

**3**
This has to do with how the retransmisson is done in HARQ, and how those retransmissions are combined in the receiver.

Chase combining is based on retransmitting exactly the same codeword (information and parity bits) as in the first transmission. The two – or possibly more – received signals are added together and then decoded.

In incremental redundancy, not the same bits are transmitted. Instead, in each transmission, a new part of a long codeword is transmitted, thus increasing the redundancy of the effective codeword in each retransmission. A new decoding attempt is done after each retransmission, based on all received bits. The result after a retransmission is a codeword in a code with larger length and minimum distance than after the previous transmission.

**4**
*The question was not formulated as intended. The second last sentence was supposed to be "Still, there are mobile phone systems using TDD that give the users the impression of full duplex communication". The answer below is for the intended formulation. The grading of the task is of course generous because of this mistake.*

Full duplex means that two communicating nodes can continuously communicate in both directions simultaneously. In half duplex, communication is possible in both directions, but not simultaneously.

In FDD (Frequency-Division Duplex), the two directions use different frequency bands. Thus it is possible to provide communication in both directions simultaneously, since the two frequency bands can be separated using filters.

In TDD (Time-Division Duplex), the two directions take turns using the same frequency band. It is thus by construction impossible to truly provide communication in both directions simultaneously.

Mobile phone systems based on TDD use the fact that the human ear accepts some delay. Sound can therefore be recorded during one period and sent during another. Therefore, it is possible to use TDD with short enough (a few tens of milliseconds) intervals in each direction, and still give the user the impression of full-duplex communication.

**5**
We start with the problem.

This has primarily to do with the uplink. Ideally, subchannels used by different users should be orthogonal. Each subchannel would then be possible to filter out individually. In that case, different users would not interfere with each other. In practice, due to oscillator uncertainty and timing uncertainty, and sometimes also design choices, the subchannels are not completely orthogonal, and thus users do cause interference to each other.

The other part of the problem is called the near-far effect, and is due to the fact that different users experience different channels. This is to a large extent due to different distances between the different users on one hand and the basestation on the other, and that is the reason for the name. It is also due to multi-path propagation and shadowing. If all users use the same sender power, then since they experience different channels, the corresponding received powers will be different. Since the channels are not completely orthogonal, users with good channels will cause unacceptable interference to users with bad channels.

The solution to the problem is power control, which aims at obtaining the same received power from all users at the basestation, by adjusting each user's sender power. This is done in the following way. The basestation estimates the received power for each user, and individually orders each user to increase or decrease its sender power.

The result is twofold: Each received user signal experiences approximately equally much interference. Also, each mobile phone uses approximately as little power as needed for the situation, thus increasing the battery life of the phone compared to not using power control.

**6**

An isotropic antenna is a theoretical concept. It is an imagined antenna that radiates equally much in all directions. In free space, the result would be that the received power depends on two things, the distance between sender and receiver antenna on one hand and the type of receiver antenna on the other, provided that the receiver antenna is oriented the same way relative to the sender antenna.

Consider a sphere of radius $r$ with its center at the sender antenna. Since the signal propagates through free space, no power is lost. Since the sender antenna radiates equally much in all directions, and no power is lost, the sent power is distributed uniformly over the surface of this sphere, i.e. the power density is constant on that surface. More precisely, the power density on the surface of the sphere is the sent power divided by the surface area of the sphere. The important thing here is that the surface area is proportional to $r^2$.

The type of receiver antenna results in a scalar constant multiplied with the power density on the surface. This, however is in fact an approximation that is good only for large $r$. Obviously, it cannot be true for very small $r$, since it would in that case be possible to recieve more power than was sent, and that is completely impossible by the energy principle. The physical size of the receiver antenna has to be significantly smaller than $r$ for this approximation to be good.

**7**

This is about multiple access, i.e. methods for sharing a common channel among several users.

**DS-CDMA** - Code Division Multiple Access
The narrow-banded signal that we wish to send is multiplied by a spreading sequence consisting of pseudo-random $\pm 1$. The result is that the signal is spread over frequencies. Different users occupy the same frequency band at he same time by using different spreading sequences and the cross correllation between these sequences is low. In the receiver, the signal is once again multiplied by the appropriate spreading sequence, and we regain the original signal, while all other users signals are still spread and constitute a small noise to our signal.
Exemple of an application: Third generation mobile telephony.

**FDMA** - Frequency Division Multiple Access
Every user is given its own sub-band of the available frequency band, such that no sub-bands overlap.
Exemple of an application: Several variations of the first generation mobile phone systems, e.g. NMT.

**TDMA** - Time Division Multiple Access
Every user is given all of the available frequency band during his own time interval in a repeated sequence of intervals, such that no two user intervals overlap.
Exemple of an application: In GSM, a combination of frequency hopping and TDMA is used, or more correctly: In GSM, the available frequency band is splitted into sub-bands. Those sub-bands are then splitted into eight sub-channels by TDMA, and the standard allows for frequency hopping after each block of eight intervals.

**8**

We asked for three methods. Here is a bunch of methods that we discussed during the lectures. Any three of them are enough to score points.

*Perceptive Quality Measures:*

**A-B discrimination test**
People listen to both the original sound and the result after encoding and decoding. They are asked to identify the original. The result is then a percentage of correct identifying.

**Mean opinion score**
People listen to encoded/decoded sound and rates the quality from 1 (bad) to 5 (excellent). After averaging, this gives a quality measure in that scale.

**Diagnostic rhyme test**
People listen to encoded/decoded sound of similar words, and are asked to identify one of them. The result is a percentage of correct identifying.

*Objective Quality Measures:*

**Signal-to-noise ratio (SNR)**
Power of useful signal divided by power of distorsion. A high value is supposed to be good.

**Segmental SNR**
As SNR above, but the powers are averaged over short intervals.

**Perceptually weighted SNR**
A variant of SNR, where distorsion in weak frequency bands is amplified. Weak as in where the useful signal is weak.

**9**

You do not need to explain your answers in this task. All that is needed is a true or a false. However, a short explanation or comment is given here for some claims.

**a**. **False**. It is true for lossless source coding. There is no real lower bound for lossy source coding.

**b**. **False**. Gold sequences are pseudo noise sequencies.

**c**. **True**.

**d**. **True**.

**e**. **False**. Puncturing is used to adjust the rate and error correcting capability of an error control code.

**f**. **True**.