University of Agder
Department of Information and Communication Technology

EXAM

Course code:	**IKT 444**
Course title:	**Mobile Communication Networks**
Date:	Tuesday, 6th December 2016
Duration:	09:00 – 13:00
Number of pages including front page:	8
Remedy:	Calculator
Notes:	*The questions are given in English only, but you can write your answers either in English or in Norwegian*

Teacher: **Frank Y. Li**

Part 1: Multiple Choice Questions (20%)

For each question below, there may be one or several correct answers. You get 1 point if you mark a correct answer alternative. You also get 1 point if you leave the wrong answer alternative unmarked. Otherwise you get 0 point.

There are 7 questions and 20 answer alternatives in this part. You do not get any point if you mark all 20 answer alternatives as correct, or leave all 20 answer alternatives unmarked.

Mark an answer alternative as correct by filling in the check-box ☐ with an 'X' (☒), or leave it blank to unselect a wrong answer alternative. Draw an 'O' around the check-box (⮾) to invalidate an already marked answer.

1.1 Fundamentals on cellular networks and services (3%)

Which statement(s) below is/are correct with respect to the fundamental concepts in cellular networks and for providing services in mobile communication networks?

☐ Mobile communication in a region does not explicitly mean the existence of cells in order to cover that region

☐ The shape of a cell in real-life cellular networks is either hexagonal or circular but the cell size may vary from macro-cells to pico-cells

☐ Data services can also be provided based on CS (Circuit Switched) technologies, e.g., ISDN (Integrated Services Digital Network)

1.2 Core network evolution from 1G to 3G/4G (3%)

Which statement(s) below is/are correct regarding the core network evolution from 1G to 4G?

☐ Although 1G is an analog cellular network, its core network is based on digital telephone network

☐ In the 2G GSM-only core network, no PS (Packet Switched) domain exits

☐ In the 3G and 4G networks, there is always a PCU (Packet Control Unit) which classifies traffic types so that voice or data traffic goes to the CS or PS domain separately

1.3 Capacity and number of users in a cell (3%)

Which statement(s) below is/are correct regarding the concepts of capacity and the number of users supported in a cell?

☐ According to the UMTS standard, the number of users served by a CDMA (Code Division Multiple Access) cell is the same regardless of the types of ongoing services

☐ Since more users can be supported with a larger number of timeslots in a frame, it is beneficial to design a TDMA (Time Division Multiple Access) frame structure with 16 timeslots than with 8 timeslots (given that there are a fixed number of channels in a cell)

☐ In the 1G FDMA based cellular network, the number of users served by a cell is decided by the number of duplex channels available in the cell

1.4 The principles of SIP/IMS (3%)

Which of the following statement(s) is/are correct with respect to the principles of the SIP (Session Initiation Protocol)/IMS (IP Multimedia Subsystem) architecture?

☐ SIP works only when both ends for a connection are IP based

☐ SIP itself alone is not sufficient to establish a multimedia session

☐ CSCF (Call Session Control Function) in the IMS does not support voice traffic

1.5 How are higher data rates achieved in GPRS and HSPA? (3%)

Which statement(s) below is/are correct regarding the basic principles of GPRS and HSPA?

☐ The achieved data rate for one user over a GPRS link is variable, depending on the quality of the channel, even if this user is always occupying the same number of timeslots

☐ Following the similar technique as used in GPRS, a user may occupy multiple orthogonal CDMA codes in HSPA for his/her transmission but not more than five codes

☐ In HSPA, the coding rate could be 1 if the channel quality if excellent (error-free)

1.6 Reliability and availability in communication networks (3%)

Which one(s) of the following statement(s) is/are correct with respect to the concept of reliability and availability in communication networks?

☐ MTBF (Mean Time Between Failures) is a dependability metric which applies only to reparable systems

☐ An end user cares more about the reliability than the availability of the communication network he/she has subscribed to

☐ A communication network becomes more reliable when more components are added, in parallel but not in series, to the network topology

1.7 Handover and roaming in cellular networks (2%)

Which statement(s) below is/are correct regarding how handover and roaming are performed in cellular networks?

☐ While a roaming mobile station is in a visiting network, it still keeps its original IMSI (International Mobile Subscriber Identity) number

☐ During the handover procedure in 2G/3G cellular networks, a mobile phone may measure the signal strengths from both the currently-connected and the target base stations but it is connected only to one of them

Part 2: Essay Questions (80%)

Write your answers to the following questions on a separate sheet of paper.

Question 2: Fundamental Knowledge on Communication Networks (20%)

2.1 Signaling in digital telephone networks (6%)

What are in-band signaling and out-of-band signaling respectively? Give one example for each of them. Discuss briefly one benefit and one disadvantage of employing common channel signaling in digital telephone networks.

2.2 Delay and QoS (Quality of Service) in communication networks (8%)

In a typical mobile or wireless communication network, which four types of delays form the main components of the end-to-end delay?

For QoS provisioning in communication networks, what are delay-tolerant and loss-tolerant services respectively? Give one example for each category.

2.3 Understanding the Erlang-B formula (6%)

The Erlang-B formula as shown below is widely used to calculate the blocking probability for incoming calls in a loss system, where *Pb* is the blocking probability, *c* is the number of trunks in the network and *a* is the offered traffic load.

$$P_b = \frac{a^c / c!}{\sum_{k=0}^{c} a^k / k!}$$

2.3.1 Given that $a=0.15$ Erlang and keep c as a variable. How many minimal number of trunks do we need in order to guarantee a blocking probability of $Pb < 10^{-5}$? (3%)

2.3.2 Re-consider this network as an IP-based network. Assume that the incoming IP flows still follow the same assumptions so that we can calculate blocking probability using the same formula. List one technique which can reduce *Pb* and explain briefly how it works considering the co-existence of various types of traffic in an IP network. (3%)

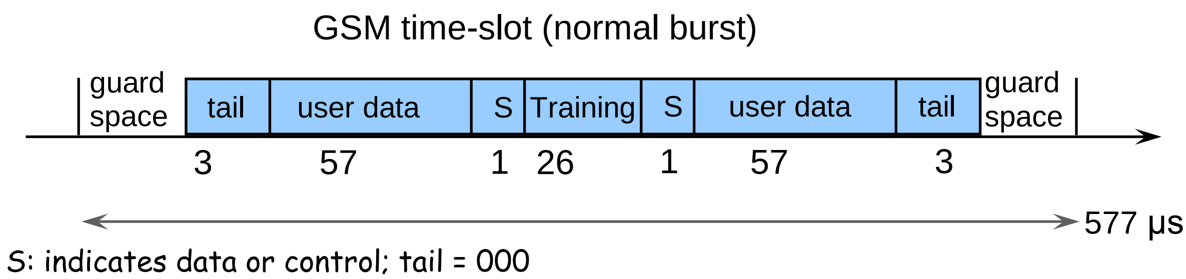
Question 3: 1G, 2G, 3G and 4G Radio Access Networks (30%)

3.1 1G and the concept of cells (4%)

What is the main idea behind the concept of cells in mobile communication networks?
Discuss briefly the pros and cons of this concept.

3.2 TDMA principle and GSM radio access (10%)

3.2.1 According to the GSM standard, a TDMA frame is divided into 8 timeslots. The following figure illustrates the frame structure for a normal burst within a GSM timeslot.

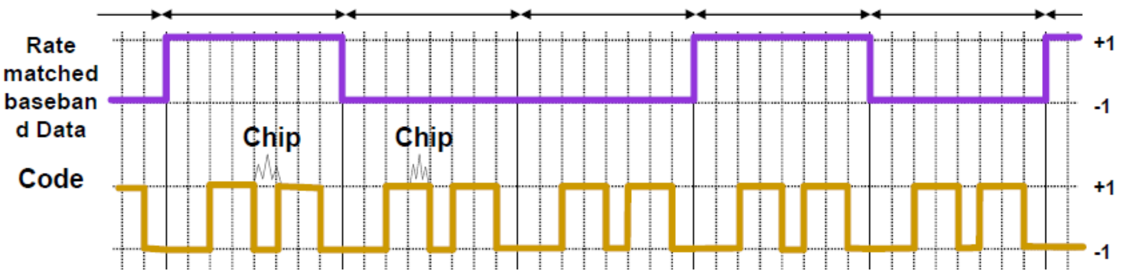


What is the achieved transmission data rate over the radio channel between a mobile user and the base station (i.e., channel data rate for each user)? Given that the encoded source bit rate for voice traffic is 22.8 kbps, how do we achieve the same transmission data rate as you calculated above (Hint: consider the burst formatting procedure shown in the figure)? (6%)

3.2.2 Assume that we re-design a TDMA system following the same parameters as used in GSM but the number of timeslots per frame is doubled to 16 timeslots. Considering that total channel capacity is still 270.8 kbps, can this system satisfy the source bit rate requirement for voice traffic as 22.8 kbps? If not, how can we make it possible? (4%)

3.3 CDMA principle and UMTS radio access (8%)

3.3.1 SF (Spreading Factor) is a basic concept in CDMA. What is SF used for? What is the obtained SF according to the following figure? If a UMTS user is going to transmit at a higher data rate, will the allocated SF to this user be increased or decreased and why? (6%)



3.3.2 According to the UMTS FDD frame structure, a frame is divided into 15 timeslots and multiple users are transmitting in the same timeslot in a synchronous manner. What is the benefit of using synchronous CDMA? (2%)

3.4 OFDM principle and LTE/LTE-A radio access (8%)

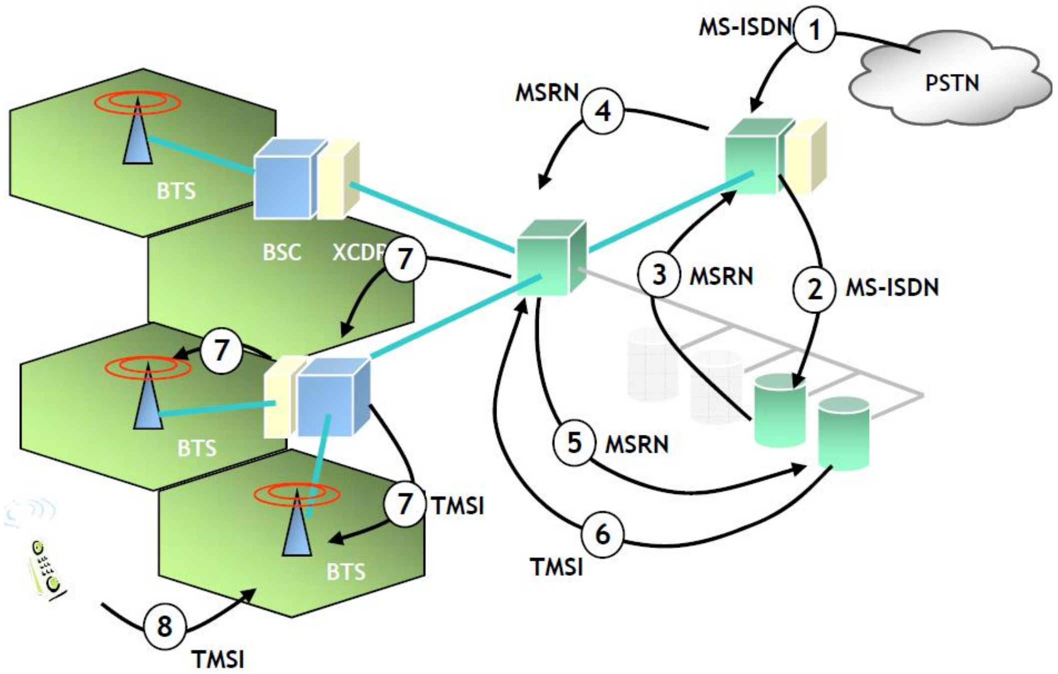
3.4.1 OFDMA (Orthogonal Frequency Division Multiplexing Access) is adopted as the fundamental medium access mechanism for LTE/LTE-A. Given the fact that a sub-channel or tone in OFDM has very narrow bandwidth, how can OFDM achieve very high transmission data rate? From its principle, explain how frequency-selective fading is combatted in OFDMA. (4%)

3.4.2 Relay is one of the main techniques adopted in LTE-A in order to further achieve higher data rate. From the basic principles of radio propagation and Shannon capacity, explain how higher data rate is achieved by relaying data transmission between a mobile phone and its associated base station. (4%)

Question 4: GSM/GPRS/UMTS/LTE Core Networks and Core Network Evolution (30%)

4.1 GSM and GPRS/UMTS core networks (16%)

4.1.1 The following figure illustrates how a mobile-terminated call for a roaming user is performed in a GSM network. Explain respectively what MS-ISDN (Mobile Subscriber ISDN number), MSRN (Mobile Subscriber Roaming Number), and TMSI (Temporary Mobile Subscriber Identity) are. Based on this figure, describe briefly how this roaming user is reached by a PSTN call initiator. (6%)



4.1.2 The above figure illustrates a purely CS network. Is it possible to provide data services in such a network? If yes, give one example and explain briefly how such a data service is provided. (4%)

4.1.3 Assume that the above network is evolved to support IP services but only the core network part is upgraded to support the IP protocol stack. Which two major entities are introduced for this core network evolution? Explain the main steps and protocols adopted in order to provide Internet access for UEs in this network. (6%)

4.2 IP-in-IP and voice traffic in 3G/4G networks (14%)

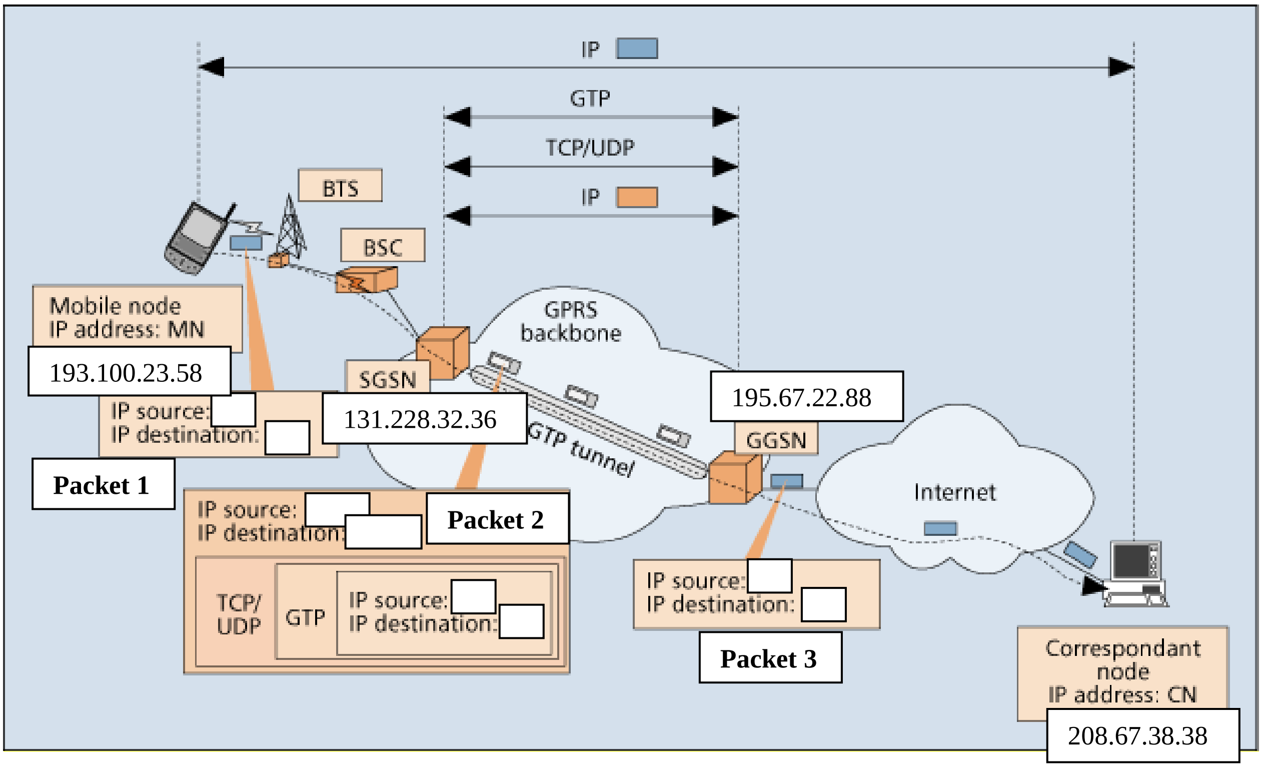
4.2.1 IP-in-IP may be needed for carrying inter-domain traffic in 3G/4G networks. Assume that the mobile node (MN) in the following figure has obtained a temporally IPv4 address as IP_MN = 193.100.23.58 and the IP address of the corresponding node (CN) is IP_CN = 208.67.38.38. The other known IP addresses are marked in the figure, as IP_SGSN = 131.228.32.36 and IP_GGSN = 195.67.22.88. (8%)

Fill out the IP source and destination addresses used for the three IP packets illustrated in the figure in order to establish a connection between the MN and the CN.

Packet 1: Source address = ________________, Destination address = _________________

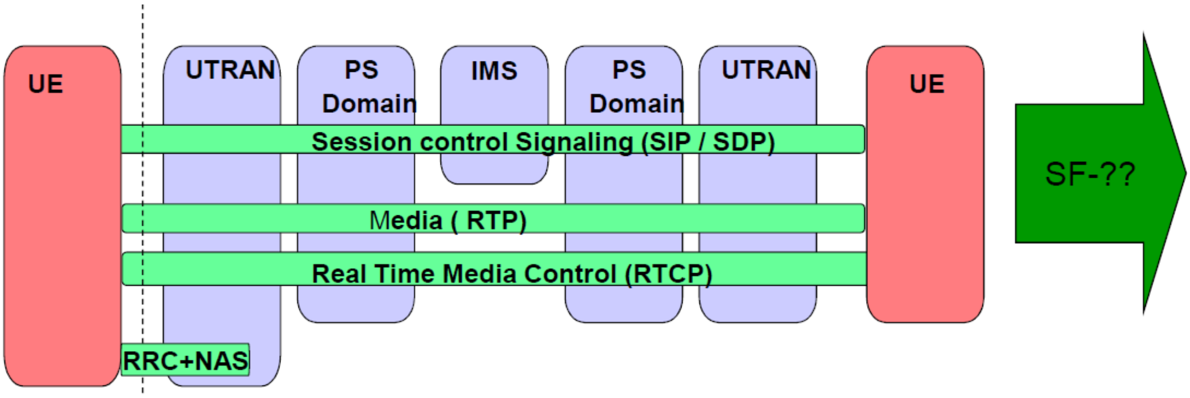
Packet 2: Source address = ________________, Destination address = _________________

Packet 3: Source address = ________________, Destination address = _________________



Assume further that the obtained IP address for the MN is IP_MN = 10.10.20.30.
Which one(s) of the above six addressed would be changed and why?

4.2.2 The following figure illustrates how voice services can be provided in IMS based core networks. Why is there no CS domain in this figure? Is it true that the SIP session is always end-to-end regardless of the type of connection a UE has with this network (explain why or why not)? Assume that the radio access part of this network is LTE/LTE-A, what is the spreading factor used for voice traffic? (6%)



Candidate number:

Answer Sheet for Part I: Multiple Choice Questions

(You may answer the questions from Part I on this page or directly on pages 2-3)

1.1	**1.2**	**1.3**	**1.4**
☐	☐	☐	☐
☐	☐	☐	☐
☐	☐	☐	☐

1.5	**1.6**	**1.7**
☐	☐	☐
☐	☐	☐
☐	☐	