Norges teknisk-naturvitenskapelige universitet
Institutt for telematikk

EKSAMENSOPPGAVE I TTM4130 – EXAM IN TTM4130 Solution
TTM4130 - Tjenesteintelligens og mobilitet
TTM4130 – Service intelligence and mobility

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Tlf.:	97 72 72 27
Eksamensdato / Date:	21. mai 2010 / 21st May 2010
Eksamenstid /Time:	15.00 - 19.00 / 3pm - 7pm
Vekttall / Credits :	7,5 SP
Examination aids:	**D:** No written and handwritten examination support materials. A specified, simple calculator is permitted.
Tillatte hjelpemidler/	**D:** Ingen trykte eller håndskrevne hjelpemidler tillatt Bestemt, enkel kalkulator tillatt
Språkform / language:	**English** is the authoritative language (Bokmål finnes) **Answer can be in nynorsk, bokmål or English**
Number of pages in English:	6 (pages 2 - 12)
Antall sider bokmål:	6 (sidene **Error! Bookmark not defined. - Error! Bookmark not defined.**)
Antall sider nynorsk:	0
Appendix (in English):	4 (pages **Error! Bookmark not defined. - Error! Bookmark not defined.**)
Sheets for drawings (in English):	2 sheets which may be handed in (pages **Error! Bookmark not defined.-Error! Bookmark not defined.**)
Sensurdato[1]:	**14. juni 2010 / 14th June 2010**

[1] Merk! Eksamenskontoret vil ikke kunne svare på slike telefoner.

You should **start by reading through all the material** and then decide which sequence you want to use when answering the exercises. In some places there are some dependencies that are listed, but all other places the exercises are independent.

2 sheets with diagrams to be filled in by the candidate may be used to save some drawing and are attached at the end. It is optional to use these sheets.

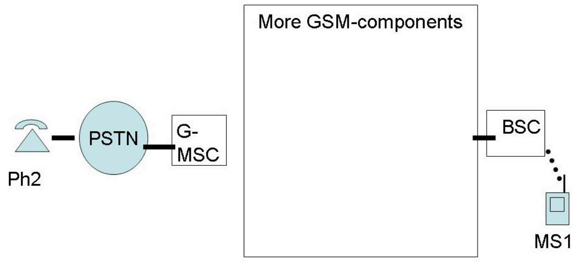
Short answers are requested, UNLESS otherwise stated.

Make your own reasonable assumptions if the text is unclear.

A **collaboration diagram** (or communication diagram) is a diagram with boxes and arrows showing a call flow in a semi-formal way. The arrows need not be placed as in an MSC, hence sequence numbers are attached to the arrows. (The "Sip trapezoid" is one example of such a diagram)

1 Exercise 1. (16 %) On GSM

Take as the starting point the following figure:



Figur 1 Illustration of GSM - - > PSTN and PSTN - - > GSM calls
NB Similar figure exists on an extra sheet on page Error! Bookmark not defined. **. Use this sheet if you like (recommended).**

Assume for simplicity that MS1 is stationary and stays connected to one BSC for as long as this exam lasts. This BSC is in a foreign domain (i.e., MS1 is roaming).

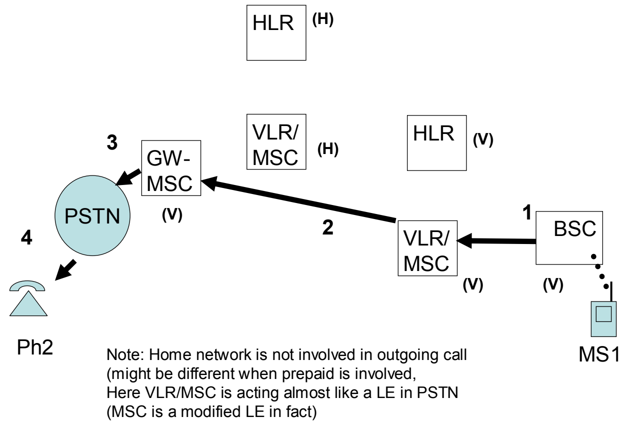
Use the same level of details in your answer as used in figures in appendix **Error! Reference source not found.** and **Error! Reference source not found.** (Establish or similar).

a) (6 %)
Assume that MS1 is already registered with its own (home) HLR, and that the human using MS1 has activated CLIR (Calling Line Id. Restriction / hidden A-number)
Explain what data is held by HLR, VLR and MSC at this point in time.
Answer:
2p HLR holds ref. to VLR
2p VLR holds TMSI and a ref. back to HLR (in order to perform loc.updates)
2p MSC holds call rel. Subscription data rel. to CLIR (and more), (this will ensure that Establish is sent with correctCLI-bit)
(An answer treating VLR/MSC as one entity will also be fully honoured)

b) (6 %)
Draw a collaboration diagram for the case that MS1 places a call to Ph2.
Indicate the right domain (H or V) for each GSM component.
Indicate how the caller's activation of CLIR impacts the parameters in the various

messages.
You may use the extra sheet on page Error! Bookmark not defined. **to save some drawing.**

Answer:



1 a variant of call setup
2 Establish (to: dialled-number)
p-bit is set by the serving MSC before the message 2 is send towards GW-MSC
(MSCs communicates eitehr directly or some network in between here)
3 establish is sent over a domain border
(optional: clid may be removed if new domain is not trusted, or sent with P-bit 1
4 alert (no CLId sent) (hidden A-number)

minus for showing GW-MSC at home or involving HLR in call setup and if the statment 'no CLId sent' in Alert is missing.

A handmade figure is OK, and (HLR and MSCs) not used need not be shown at all.

c) (4 %)
Draw a collaboration diagram for the case that Ph2 places a call to MS1.
You shall also indicate the right domain (H or V) for each GSM component.
You may use the extra sheet on page Error! Bookmark not defined. **to save some drawing.**

Solution for 1 b is identical to material in append. **Error! Reference source not found.**!
but add H and V as follows:
B's HLR@H and B's MSC@V (sometimes even called V-MSC)

2 Exercise 2 (20 %) On GSM and IN

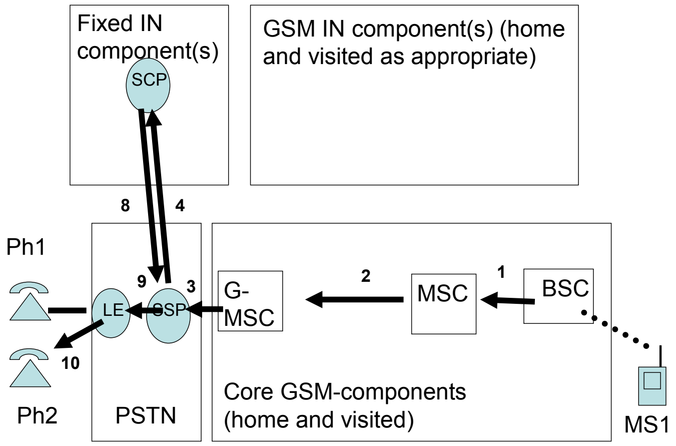
It might be useful to solve exercise 1 before this one.

a) **(**6 %)
Read the appendix **Error! Reference source not found.** about Italian Letizia visiting Norway.
Show a collaboration diagram or call flow or explain in text how the call is set up.
You need only show the messages from right to left, i.e., you may stop when Ph2 is

receiving Alert.
You shall show how IN is involved, but you need not show details of the playing of the announcements via IP. You may use IN-1, IN-2 and similar as you find appropriate.

You may use the following figure to guide your drawing. You may **use the extra sheet on page** Error! Bookmark not defined. (or make your own drawing).

Solution for 2a:



1, 2, and 3: setup or establish (to: 8xxxxxxx)
4: (no name given in lecture) handle call via IN (number: 8xxxxxxx must be sent 'up')
5-7 (IN-IP voice menu not requested in reality more messages than 3)
8 (no name given in lecture) call continue (number: 73735678 'down')

9 Establish (to 73735678)
10: alert
(stop here according to details req. / of course alerting etc. will follow as well)

b) (14 %)
Read the material provided in Appendix **Error! Reference source not found.**.
You may assume that the calls taking place during the swindling are initiated and terminated in Italy, by a service similar to Future-TellingAS, here called Futura-Italia.
It might be useful to answer 2a) before you answer this task.

Write a small fact box explaining what IN and telemarket [Norw.: Teletorg] is. Use **approximately 150-250 words**. Make efforts to write in a way suitable for a newspaper article writing about the case. Pay attention to those parts of IN which is of relevance to the money laundry (i.e. the money aspects are of special interest).

For the particular task (2b) we will judge you based on technical correctness, but also on language, length, and relevance of the technical information.

Solution for 2b:

Teletorg/telemarket allows the charging for a call to be free for the caller (like 800-numbers) or charged at extra (premium) rate (820-829 services). Such premium rate telemarket services can be used by a lawyer or future teller and other companies in order to charge 5, 10 or 29 nok/min. for phone calls. The bill will be handled by the telco offering the service. The telco will charge the enduser (those dialling the number) via the ordinary phone bill, whether that is paperbased or via prepaid. The money (minus a fee) will be sent to the company operating this 820-number (such as Futura-Italia)

lill 5/20/10 4:07 PM
Comment [1]: Here the language can be further improved

In this case the swindlers are establishing such an 820 service (say Futura-Italia) in a country like Italy by establishing a relation to the fixed operator in that country (e.g. Ital-Fixed-Tel). They are also obtaining SIMcards from a company like TalkLess in the name of someone/person or company (like Inno) in Norway. From Italian GSM networks they dial the 82o-number with Norwegian SIM card . Due to roaming agreements common in GSM this is possible. The idea is that the money shall be paid from Ital-Fixed-Tel to Futura-Italia before someone gets suspicious. In this case a company Inno in Norway and TalkLess in Norway is involved as well as ItalTel and Ital-Mob-tel in Italy. This setup will cause some delay and hide the setup. **(226 words)**

Max. 6 points for technically correct and relevant information

Up to 8 points for language and relevance

Minus for students describing

- Outgoing calls involving IN on the *originating* side: This case is about a non-IN subscriber calling to a company with an IN-number. the details of the incoming/terminating side is of less importance
- Handling of announcements and routing of the incoming call to IN (like a pizza-ordering service): This case is about a non-IN subscriber calling to a company with an IN-number. The charging (not the IP/voice menues etc.) is what is important. Routing based on time of day etc. is irrelevant, the swindler does not need such advanded setup like that discussed for the pizza example.
- Handling of IN in GSM (Camel): In this case only fixed IN is involved
- Handling of routing of calls *to* a roaming GSMsubscriber: The case is about an *outgoing call* from a roaming GSM subscriber
- Other irrelevant technical info, even though it is correct. I.e. 0p for just listing all you know about IN&GSM

The fact box on SIM from the newspaper would not obtain many points on the exam if the task was to write a fact box on SIM card or GSM relevant to the case. They describe the case of an *incoming* call to GSM. They almost say that routing in GSM is based on the serial number (?!). And they are unclear about how the ‘phone number’ is related to SIM and to the phone itself: “The subscriber’s phone number is normally not stored on the card, so that one can move the SIM card between mobile phones.”+ “[HLR] checked to find which SIM card number belongs to the respective phone number” (Maybe they believe the phone number stored on the phone and the SIM ID + phone number is sent to HLR during registration??)

I am not sure whether that phone number is stored on the SIM or not, but this is NOT relevant to the swindling case. The important thing is that you can buy a phone separately (without stealing any id), and then buy SIM card separately (with the help of a stolen ‘id’).

In fact I am not sure how much 'id' you need to present to buy a SIM card on-line. Filling in name of CTO and street address on the web can hardly be said to be 'stealing an id'. But picking up a parcel from another persons/companies mailbox is probably illegal. Ordering things in other persons name is probably illegal, but if this is done without requireing any Id, then who is responsible?

For the student corcerned with privacy: Opening other persons snail mail (skriftlig Meddelelse) is illegal for ordinary persons. (This is a really old law I believe. I am not a lawyer, but I think this is now covered here in Nowegian law: http://www.lovdata.no/all/hl-19020522-010.html#146)

However not necessarily illegal for authorities:. "Bestemmelsene om postkontroll i 1915-loven har praktisk betydning. Mistankekravet i 1915-loven er forskjellig fra bestemmelsene i straffeprosessloven §§ 211 og 212. Etter 1915-loven gis mistenkte ikke underretning om postkontroll" (Militære snakker om medlesning av brev (epost), artig språkbruk!) se for øvrig http://folk.uio.no/jonhaug/security/personvern/lovoversikt.html for an overview of Norw. laws for privacy.

3 Exercise 3. (27 %) On SIP

It might help you to think in the global case here, taking into account network delay, congestion and packet loss (as you find suitable).

(DNS and Location server are removed for simplicity, you need not reintroduce them.)

a) (4 %)
Explain the header in SIP ensuring that all responses for a SIP METHOD follow the same path back as the METHOD followed on its way forward.
Explain briefly how it works.

Answer:

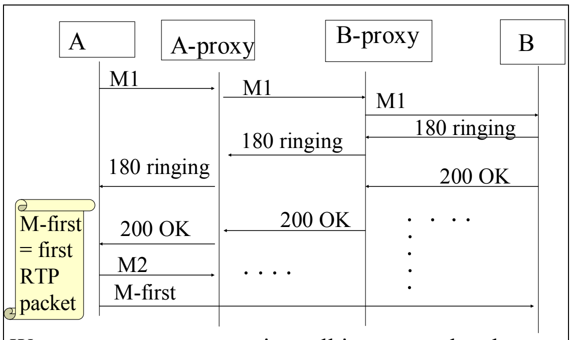
The header dealing with this is Via. It acts like a 'stack' of addresses. Building up as the METHOD is send, and popping off as the responses are sent back.
2p for Via
2 p for 'stack' or a similar explanation

b) (4 %)
Explain two alternative paths ACK (and BYE and more) can be sent according to SIP (as defined by IETF).
Explain which of the alternatives that is *not* according to the specification in IMS.

Answer:
2p ACK may be sent directly between the endpoints, future BYE and re-INVITEs may also be send directly)
(optional info: This is preferred by IETF according to end-end-principle)
2p: such E2E signalling is not allowed in IMS, IMS will have all signalling via X-CSCF (P, S etc.)
(Optional info: since IMS assumes that netw. entities like P-CSCF is involved in QoS and charging / Lab assigment used ACK and BYE via proxies even in en IETF SIP setting)

Note: the use of the header *record-route* was not asked for, and need not be present in the answer (some student asked for record-route before the exam, and the answer was posted to all students on 20th (day before the exam)

For the rest of exercise 3 we will look into the establishment of a one-way media stream. This case is chosen to make it easier for you. In case you are unfamiliar with one-way streams, don't panic! You may then use the more familiar case of a two-way voice call.



We assume a one way voice call is proposed and accepted. Figure illustrates parts of the signalling and first RTP packet

Figur 2 A partial MSC. A is a shorthand for UA-A and other entities on the terminal (similar for B). M1 and M2 are placeholders for real method names in SIP. The path followed by the signalling is the same as you used in the lab.

You shall assume for the rest of exercise 3 that all signalling goes via the proxies.

c) (4 %)
Complete the MSC in the case that everything works normal and that no messages are lost. Write the real names of SIP methods instead of M1 and M2. You shall not include the lookup in DNS etc, but you shall fill in the missing ACK's. Pay attention so that you draw the missing ACK messages in the right sequence. You **may start your own drawing with the first 200 OK** message and showing a total of 6 signalling messages (in either direction).

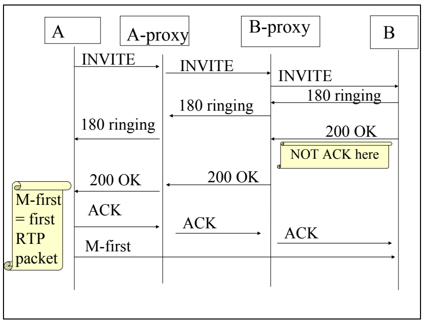
Answer:

1p for M1= INVITE
1p for M2 = ACK
2p for ACK'ene correctly places (in time *after* the first ACK),

In particular not sending ACK from Pr-B imediately when it received the 200 OK, this is a difference between 400 responses and 200 responses. Acking immediately on a 200 OK will destroy the possibility to send ACK directly between the endpoints.)

lillk 5/26/10 11:52 AM
Deleted: 2p

lillk 5/26/10 11:52 AM
Deleted: 2p



d) (4 %)
Explain why B must be prepared to receive M-first before it receives ACK.
List at least 2 different reasons for this.

Answer
2p per correct reason max 4p (no extra for knowing all 3)
1) different paths at udp level (ACK follows a longer path)
2) even two messages between same X and Y may be 'swopped' at IP-level and upd will not correct this
3) ACK may be lost due to netw. congestion (or at the radio link)

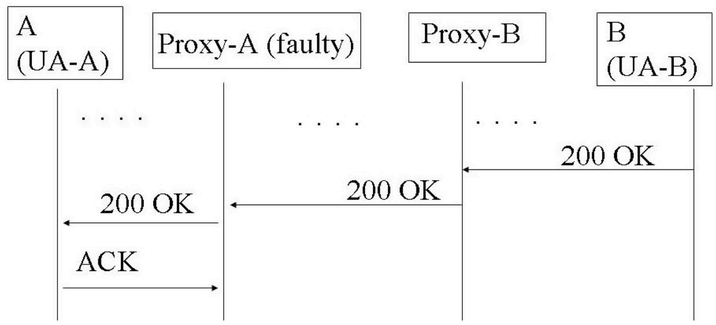
e) (6 %)
Illustrate what happen when A 'hangs up' (by pressing the red button). You shall assume that A stops sending RTP packets before it sends the relevant METHOD towards B (via proxies).
Show the signalling, but show also the (human) user interaction and how the RTP-stream stops. Chose MSC, collaboration diagram or text (or a combination).
In case you have chosen to use a two-way media stream instead: Make your own assumptions.

Answer: (drawing is preferred)
☺ - - red - -> UA (human intercation pressing 'red')
UA - -self-arrow stop media - -> UA (not strictly needed to show this)
A - - M-last -> B
A- - BYE (via PrA and further via PrB) --> to B
200 OK back same path as BYE

One should note that the last RTP-packet cannot be sent before the human har pressed red (as the stream is sending 'continiously' until asked to stop)

f) (5 %)
Assume now that Proxy-A is faulty. It behaves as it shall on all messages, except on ACK. Instead Proxy-A behaves as follows: When it receives an ACK (in any direction) it will simply ignore it (and hence not send it further), and stay in the same state as before. NB: UA-B is not aware of the faulty Proxy-A. UA-B and Proxy-B behaves as it should.

Explain what will happen further and why.



Figur 3 Proxy-A is faulty and is just ignoring all ACKs.
The other entities behave according to the SIP specification.

answer (no detailed distribution of points will be given / will bejudged by the sensor):

The bolded elements should be present
UA-B will **resend 200 OK** (because it may think that its own 200 was lost, ar that the ACK is lost due to congestion)

UA-A will respond also to this new 200 OK as before by **sending ACK again** to Pr-A (which is still faulty) (details of Cseq etc. is not required)

After a while **Human A or human B may hang up**.

If A 'hang up' then a BYE will be send, and A is expecting an OK back. (in case OK back does not arrive, A shall still release all resourses)

If B 'hang up': BYE can be used

Or **(machines) UA-A and UA-B or Pr.B may have a timer fired and quit the session.** (the word quit is chosen here as details of BYE/CANCEL is not required)

Additional info (not required) UA-A may issue a CANCEL in this case, **check details** of Pr-B and UA-B whether they shall use CANCEL or BYE.

From RFC3261:
```
The CANCEL request, as the name implies, is used to cancel a previous
   request sent by a client.  Specifically, it asks the UAS to cease
   processing the request and to generate an error response to that
   request.  CANCEL has no effect on a request to which a UAS has
   already given a final response.  Because of this, it is most useful
   to CANCEL requests to which it can take a server long time to
   respond.  For this reason, CANCEL is best for INVITE requests, which
   can take a long time to generate a response.  In that usage, a UAS
   that receives a CANCEL request for an INVITE, but has not yet sent a
   final response, would "stop ringing", and then respond to the INVITE
   with a specific error response (a 487).
```

This means in particular that if UA-A happens to send CANCEL then UA-B shall ignore it, as UA-B has already sent the final response (200 OK) (as messages may omfares / pass by eachother uncoordinated due to 'mixed initiative', it may happen that A sendt CANCEL and B sendt 200OK 'almost at the same time'.

UA-A has already sent the ACK, so UA-A should rather send a BYE at this stage.

UA-B may also send a BYE.

The statement above is not clear on whether UA-B can CANCEL a request initiated from UA-A at all. It should anyway only take place before 200 OK is sent. The case describes above is for the case that the CANCEL comes toward an entity acting as a UAS (who received the previous METHOD)

Proxies can CANCEL:

```
A stateful proxy MAY generate CANCEL requests for pending INVITE
 client transactions based on the period specified in the INVITE's
 Expires header field elapsing.  However, this is generally
 unnecessary since the endpoints involved will take care of signaling
 the end of the transaction.
```

It is a bit unclear whether proxies are allowed to cancel a req. after 200 OK from B is sent further back from the proxy. (while Pr is waiting for the ACK.

Notice the following principle explained by Audestad in the ATN-book p. 145:

- *Fault recovery:* Each process must be able to recover from faults on its own. If a process fails, all processes pending must be able to return to a safe state independent of the faulty process.(….) must be able to terminate the call and stop processes (…)

We should note the wording "on its own here". We may also note that the statement from the RFC (above): "this is generally unnecessary since the endpoints involved will take care of signaling the end of the transaction." is **NOT in accordance with the principle by Audestad.** Here the RFC asks a stateful proxy to rely on other entities to take actions. No wonder why IETF does not like stateful proxies….In fact this may explain the issue many of you experienced in the lab for the case when UA-A was (faulty configured) and not sending ACK (at least not to the right Pr-A). You experienced no time-out from pr-A or Pr-B to this.

4 Exercise 4 (12 %) NGN, IMS and IP-telephony

These days Telio is running advertisements for video telephony (see appendix **Error! Reference source not found.**).
Assume that you have NGCo as your broadband provider / ISP in your home.

a) (2 %)
NGN defines a horizontal layering with two major parts (planes or layers).
List the name of the bottom plane/layer
List the name of the plane /layer on top (several names are in use, list one)

Answer:

1p: transport plane, 1p: service or application plane/layer

b) (4 %)
Explain why the arrangement with Telio and NGCo is not fully in agreement with NGN**.**
Explain what business relationship there must be between NGCo and Telio in order for your video telephony service to work according to the promises from Telio.

Answer:

b1 2p: NGN assumes a QoS-aware transport network, but Telio assumes BE (best effort)

b2: 2p: None!

(Optional info: Telio explicitly says that the enduser is responsible for having enough capacity/speed/bandwidth and hence Telio is not having any bus.rel with ISP/transp.netw like NGCo. There is not any QoS mechanism on the telio phones to reserve bandwith ('on demand') either (again because Telio assumes (a sufficient) BE netowork.)

c) (6 %)
Another trend in NGN and IMS is the separation between access and core network.
List three possible access technologies that can be used in IMS.
List at least one fixed access technology and one radio based access technology.

Answer: 2p per correct answer but a max. of 6

1) IP over fiber

2) IP over ADSL

3) GPRS

4) WLAN (optional : in the form of I-WLAN)

Answers indicating guessing like listing GSM, DECT etc. with will be punished

Listing GSM may be honoured if explicitly stating teh VCC is asumed (not otherwise)

5 Exercise 5 (7 %) IMS

2nd and 3rd ed. of IMS book differs a bit on the description of the interconnection between IMS networks, as they are referring to different releases of IMS. As explained by wikipedia: " From Release 7 onwards this "entry point" function is removed from the I-CSCF and is now part of the *Interconnection Border Control Function* (IBCF)"

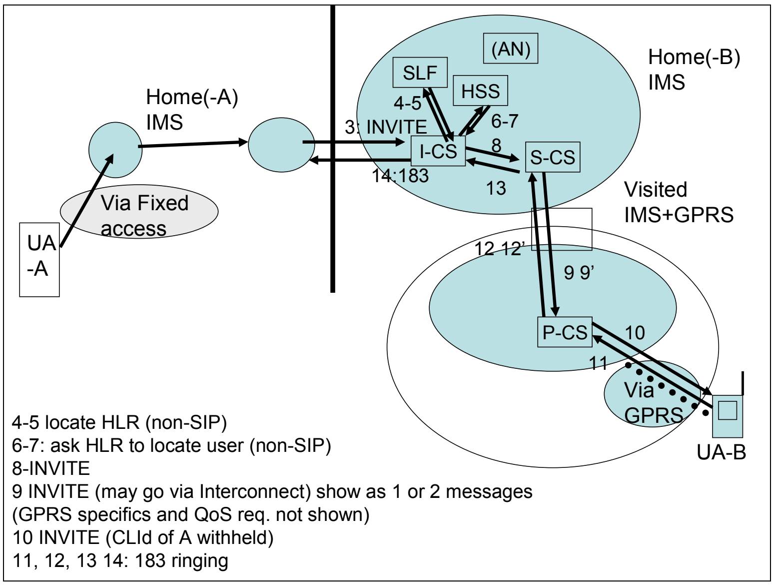
In this task you need *not* focus on the details of the I-CSCF/IBCF.

Assume that Bob with 'phone number' sip:abc@4g.org is registered at home as a roaming user. He has UA-B as his only registered endpoint, where all calls shall be delivered. You shall assume that the call signalling passes through the S-CSCF in the home network of A as illustrated below. (In other world: It is not an emergency call.)

You may use the sheet on page Error! Bookmark not defined.

a) (5 %)
Focus shall be on the terminating side i.e. between the messages number 3 (INVITE) and number Z (183 ringing).
Make a collaboration diagram or similar illustrating an incoming call to Bob.
Illustrate various CSCF, HSS etc. as you find appropriate, and place them in the right domain (H or V) on the terminating side.

Answer
AP: Sjekk: skal 4-5 med på INVITE (er med på reg.)



Figur 4 **Answering figure** Interconnect is not shown in detail, but indicated with the 'blank box' (in reality each domain will have an ICBF) No use of ICBF is also OK. Optional to show AN, even more to show the messages to/from the AN

b) (2 %)
Which messages are SIP messages (and which are not)?

Answer: Lookup to SLF and to HSS are non-SIP, INVITE, 183 are SIP messages (METHOD and response)

(Optional: SIP follows same path back, but of course the lookup to HLR is not part of the path back)

6 Exersice 6 (18 %) On GPRS (General packet radio service)

It might be that the candidate will prefer to answer a)-d) in a different sequence, which is of course allowed.

In GPRS we have the two procedures 'Attach' and 'Activate PDP context'. Attach results in an active network attachment (refer to the slogan 'Always on').

a) (6 %)
Explain class A, B and C in GPRS. Pay special attention in explaining the differences in network attachment between class B and class C.

Answer:

* Class A terminal, which supports simultaneous CS and PS traffic;

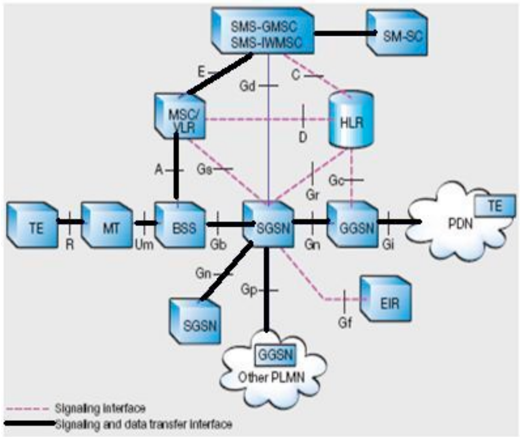
* Class B terminal, which supports either CS or PS traffic (sim.network attachment) but does not support both kinds of traffic simultaneously;

* Class C terminal, which is attached either as a packet-switched or circuit-switched terminal

2p for class A correct

2p for class B correct

2 p for class C correct, BUT only when the difference to class B is clear, like stating that .(i.e. in order to switch between CS and PS for class C a detach and a new attch must be performed, this will involve the HLR)

	**Legend** (acronyms from GSM like TE, BSC, MSC, VLR, HLR, and SMS are assumed known) BSS: Base Station Subsystem (an evolved BSC) GGSN: Gateway GPRS Support Node PDN: Packet Data Network PDP: Packet Data Protocol SGSN: Serving GPRS Support Node TE: Terminal equipment (MT can be ignored) dotted line: Signalling interface (i/f) Fat line: Signalling and data transfer i/f

Figur 5 The reference figure for GPRS from 1999 where GPRS and GSM is used together

b) (3 %)
Explain briefly the main steps in Attach (as of 1999, or a later release).
List the main components involved (ignore EIR and AUC; AUC is not shown on Figur 5)

Answer (the bolded elements in an answer should all be present for 3 p)

Attach is sent via **BSS** and **SGSN** to **HLR** indicating terminal capabilities /multislot and more. HLR authenticates and **returns the service profile into SGSN** and VLR. (Optional: Service profile incl. QoS subscrip. data and subscr.rel data like APN-names. Optional: After this procedure the GPRS terminal is ’on’ refer to ’always on’.)

NB: No GGSN is involved here. No IP-address is given **end sufficient answer**

minus if students say that GGSN is involved or state that IP address is obtained at this stage

It is not required that fig.5 fra ericsson (= **Error! Reference source not found.**) is given

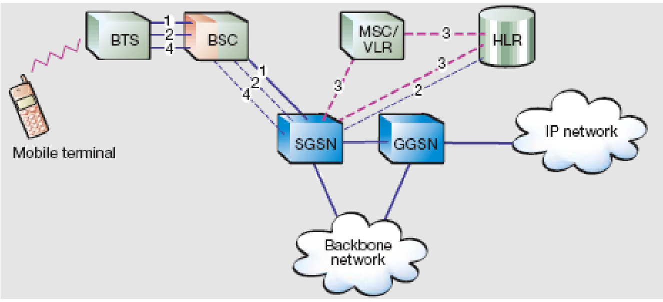


Figure 5
GPRS attach.

Figure 1 (= fig. 5 from E/// paper)

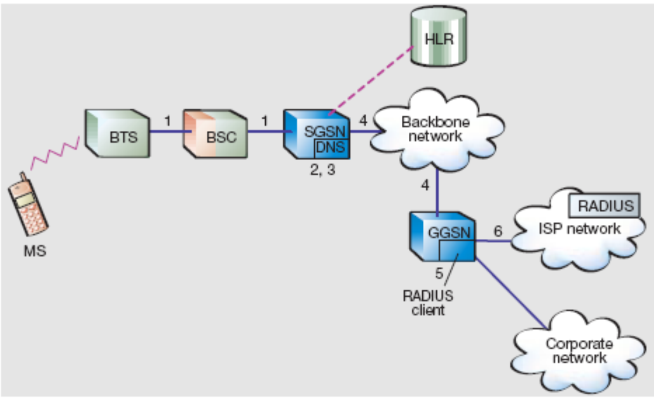
1. The mobile terminal requests that it be attached to the network. The terminal's request, which is sent to the SGSN, indicates its multi-slot capabilities, the ciphering algorithms it supports, and whether it wants to attach to a packet-switched service, a circuit switched service, or to both.
2. Authentication is made between the terminal and the HLR.
3. Subscriber data from the HLR is inserted into the SGSN (and the MSC/VLR) Comment: The sequence of these 3 staps marked 3 is a bit unclear her, but this is not important)
4. The SGSN informs the terminal that it is attached to the network.

c) (3 %)
After Attach a separate procedure ”Activate PDP context” takes place.
Explain the main steps in ”Activate PDP context” (as of 1999, or a later release).You may assume that the terminal wants to establish communication to a PDN (e.g. to www.mycomp.com).

Answer: (all bolded elements should be present for 3p)

Answer: A PDP context is **binding the SGSN to the relevant GGSN**. An id (APN,/ the name is not important) is used for this.(optional: this id (APN) is mapped to GGSN via DNS lookup internally in SGSN) . A tunnel (GTN) is established between the SGSN and the GGSN. Each PDP-context has **associated QoS-parameters**. The terminal **gets an IP-address at this point** (optional: from the GGSN). After this communicaton can take place.
(end sufficient answer,)

From E/// material:



1. The mobile terminal requests PDP contextactivation.

2. The SGSN validates the request based on subscription information received from the HLR during GPRS attach.

3. The APN is sent to the domain name server (DNS) in the SGSN to find the IP address of the relevant GGSN.

4. A logical connection is created between theSGSN and the GGSN (GTP tunnel).

5. The GGSN assigns a dynamic IP address to the mobile terminal from the range of IPaddresses allocated to the PLMN or externally,from a remote authentication dial-inuser service (RADIUS) server (a fixed IPaddress from the HLR could also be used). A RADIUS client is included in the GGSN to support password authentication protocol (PAP) and challenge handshake authentication protocol (CHAP) authentication to external networks with RADIUS servers. At this stage, communication between the user and the external packet data network can commence.

d) (4 %)
Explain some differences between HLR as depicted here and HSS as in IMS.
List one major difference regarding the protocol stack.
List one major difference regarding subscriber data for voice.

Answer:

d1: 2p HLR (as of 1999 in GPRS and in GSM) uses SS7
while HSS in IMS is IP-based

d2: 2p HLR at H sends subscr.data to VLR/MSC in visited domain
while in IMS the HSS sends these data to the S-CSCF which is in (same) home domain. (Optional: This is a difference between home and visited call control, see emergence chapt. In IMS book.)

1p if domain differences are not highlighted but HLR<--> VLR/MSC and HSS <--> S-CSCF is otherwise correct

e) (2 %)
List entities from Figur 5 that no longer exists in the new 'all IP architecture' with IMS as described in the IMS book. (I.e., you shall not look into VCC (voice call continuity); you shall consider only the IP-part of IMS.)

answer:

1 p: VLR/MSC is out, as they relate to CS.

1p: HLR in no longer present (wikipedia lists the HLR (for GPRS) as part of the HSS). IMS-book does not show SGSN and GGSN and hence does not show the look up at home for the GPRS-part.(i.e. does not show whether this goes to HLR (as was the case in 1999) or to an HSS). Audestad book (used in ATN4105) is showing a fig. rather similar to E// fig. as of 1999, i.e. does show HLR and no HSS.

We must discuss how to honour those stating that HLR will still be present and used for GPRS. **(but this will only concern 1 p)**

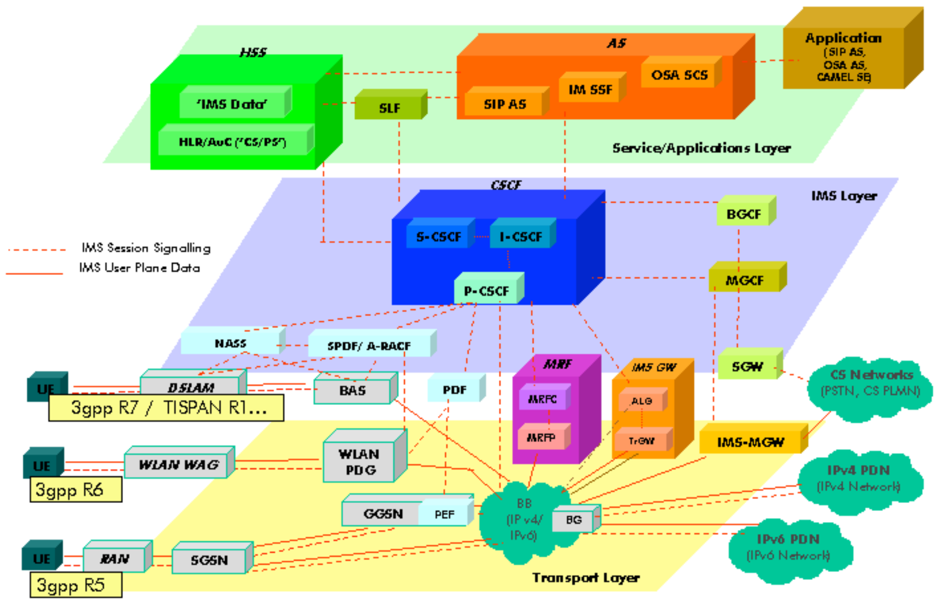


Figure 2 From http://en.wikipedia.org/wiki/IP_Multimedia_Subsystem

From the forum for students I was writing (as an answer to a GSM/HLR question):
"see also: http://en.wikipedia.org/wiki/IP_Multimedia_Subsystem (the colorful figure)
Note: This figure on wikipedia show HLR/CS as part of HSS in IMS which is a bit unusual"

Optionally we may honour answers listing SMS or similar My original plan was to honour 1 p for VLR/MSV and 1p for HLR **Appendix etc. (not given in the solution)**