Norges teknisk-naturvitenskapelige universitet
Institutt for telematikk

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

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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** English is the master text (authorative) (Norwegian text is for information only) Answer can be in or English, nynorsk or bokmål
Number of pages in English:	5 (pages 2-6)
Antall sider bokmål:	5 (sidene 7-11)
Antall sider nynorsk:	0
Appendix (in English):	4 (**Error! Bookmark not defined.- Error! Bookmark**

not defined. with text and figures)

Sheets for drawings	2 sheets which may be handed in
Sensurdato:	**17. juni 2011**

You should **start by reading through all the material** and then decide which sequence you want to use when answering the exercises.

The appendix may contain useful information. Make your own assumptions if the text is unclear, or if information is missing.

At the very back you will find some sheets with diagrams, which you are asked to use in some of the tasks.

Exercise 1 (10%) On emergency calls in IMS and in PSTN

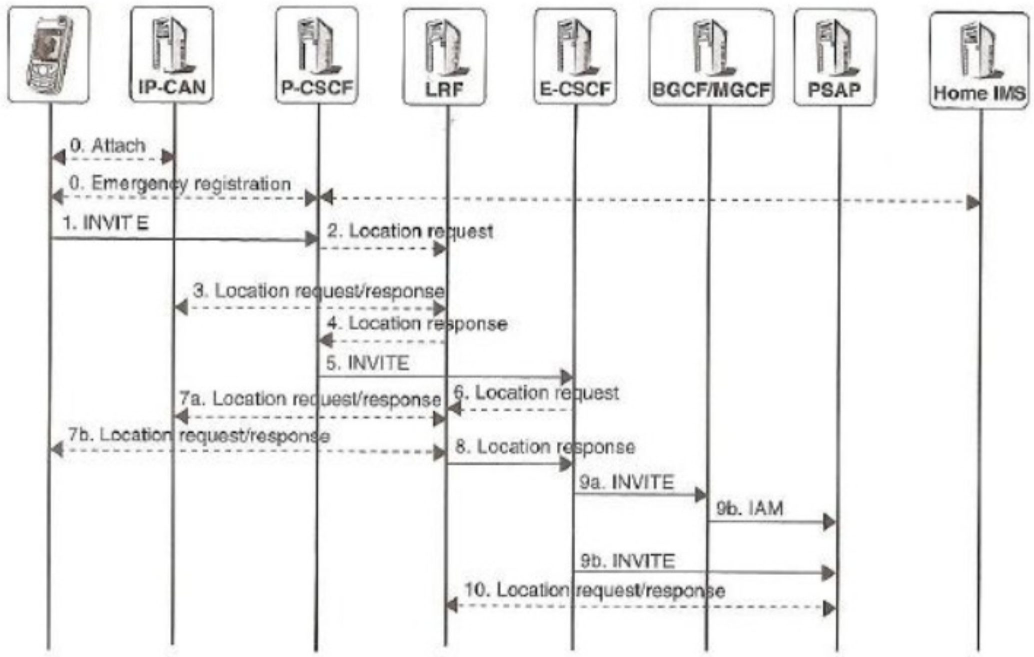


Figure 1 IMS emergency calls according to the IMS-book.
(The informal notation with a b indicates alternative choices and dotted lines are optional messages.)

a) (4%) Figure 1 shows a caller on IMS initiating an emergency call. If we chose to follow 9b in Figure 1 we will end up in a PSAP (emergency centre) connected to IMS. Explain in this case how location information from a GPS-unit on the handheld device on the left, can be brought to PSAP *without* using the messages marked as 6 and 10.

Answer: Loc. Info can be carried in the INVITE by a new header. Optional info: In draft http://tools.ietf.org/html/draft-ietf-sip-location-conveyance-04 this new SIP header is called Geolocation and discussed for use with emergency calls. The IMS-spec. probably contains this header (or a similar one)

For the rest of exercise 1 you shall look at an emergency call where both the caller and the emergency centre are connected to PSTN.

b) (3%) Explain how an emergency call is routed in PSTN. You may assume that 113 is dialled (this is the medical emergency number in Norway).

Answer: The Local exch. (LE) rewrites the number 113 to a full routable number to the PSAP (like 73737373 which is placed in CdPN), call is then further routed based on this. Additional info: In the simplest case all subscriber in same LE have the same medical PSAP-number, but in rural areas it may be that different number series for customers (different villages) may be rewritten to different routable numbers

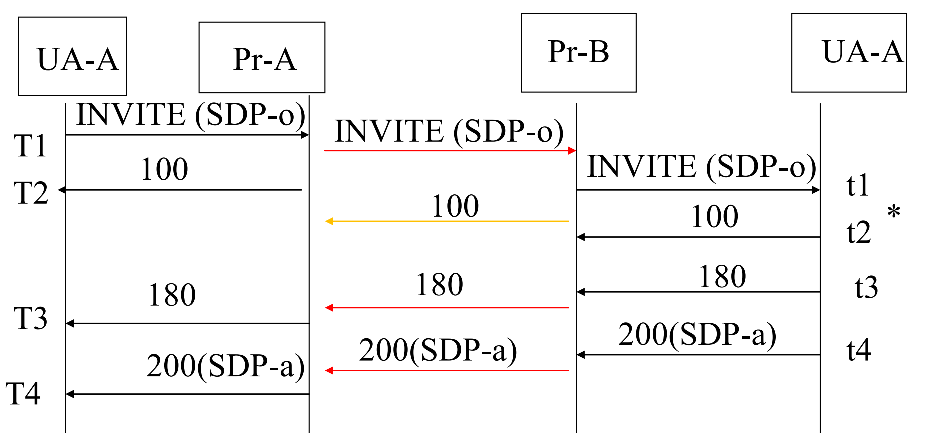
c) **(3%)** Explain how the emergency centre find out the location of the caller is the PSTN case.

Answer: CLId is always delivered to the PSAP (optional: even when caller subscribes to CLIR; in which case this restriction is overruled). After receiving the CLId the number is looked up in a database, where (static) street addresses are stored for each number in PSTN

Exercise 2 (38%) About SIP and SDP (almost as experienced in the lab)

Your tasks in this exercise are to work with some data which could have been real data captured via Wireshark in the lab. The setup, the preconditions and the test case are described in appendix. ch. **Error! Reference source not found.** page **Error! Bookmark not defined.**, and the results as seen in Wireshark are shown on page **Error! Bookmark not defined.**.

a) **(6%)** Look at the results from test run 1 given in append. ch. **Error! Reference source not found.** page **Error! Bookmark not defined.**. Use your knowledge of SIP as defined in RFC3261 to draw one MSC showing the *whole* set of signalling messages being exchanged within the given timeframe between all 4 involved entities (2 proxies and 2 UAs). (You need not show DNS). Use the messages from the Wireshark observations, but add messages on the entities where no Wireshark observation is given. Your MSC shall contain 4 vertical 'timelines', one for each of UA-A, UA-B, Proxy-A and Proxy-B. **Use** Error! Reference source not found. **on the drawing sheet on page** Error! Bookmark not defined. **for your answer.**



Optional info (not required in an answer)
Red arrows means mandatory messages (MUST have happened during the test for the rest to make sense). Orange 100 is optional. We do not know whether orange 100 is sent before of after the INVITE to UA-B
In theory red 180/200 may have swopped in time (and swopped back again before arriving av UA-A). Note that is is a bit strange that 100 is sent from UA-B, normally one goes straight to 180 on the UA, but this 100 MUST be given since my fake wireshark diagram showed this message

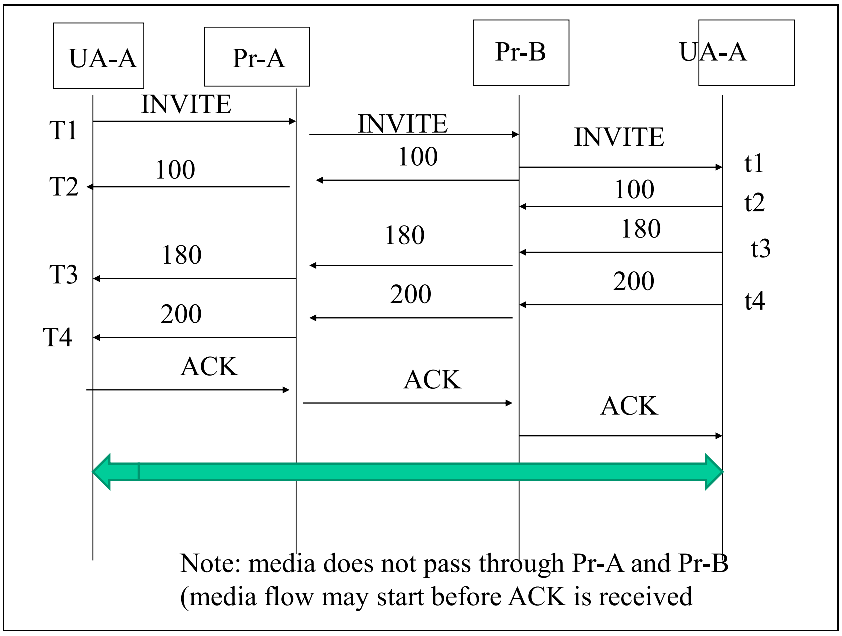
Figur 1 Answer to 2a, 2c (the *) and 2 d (the SDP-parts)

Actually I should check that 100 from UA-B is formally allowed according to RFC3261. It was a mistake (cut&paste-error), but it complies to the METHOD-respons(es) mechanism inherited from http. However, since UA-B is the final recipient sending 100 and then immediately 180 does not make much sense, but it can be allowed anyway.

Students are also allowed to make own assumptions, and to assume that I did a Cut&Paste-error is allowed. (Only if that assumption is listed can the student delete the 100 message from UA-B)

b) **(5%)** Complete the call setup, as it shall be when everything works fine and according to the specification. Illustrate the media flows and where the signalling is going.

Answer: (Here the time T1,T2 ..T1 etc may be removed).

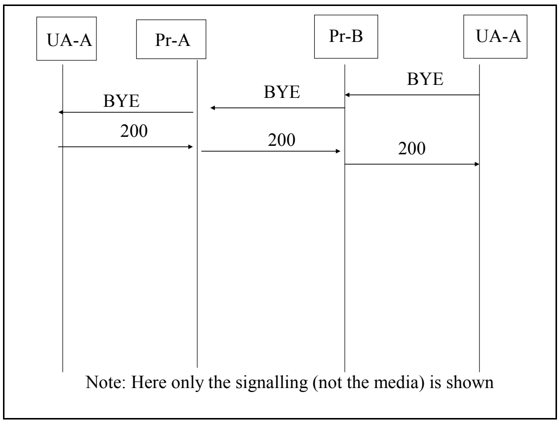


Figur 2

Additional info: Note that we cannot tell from the observed Wireshark the time-sequence on Pr-A and Pr-B. Impossible to tell whether Pr-A first answer 100 and then forwards the INVITE or vice versa, hence both variants are honoured. I comment that two options exists is an additional plus.

c) **(4%)** Illustrate in the same MSC or in a separate figure what happen when after 2 min. of conversation the called party decides to "hang up" (i.e. Bob is pressing the red button on his user interface).

Answer:



d) **(4%)** Which message(/s) from test run 1 are containing the *SDP offer*, and which message(/s) are containing the *SDP answer*? (The terms offer and answer is used according to the terminology in RFC3264) (You may refer to your answer a)

Answer: See (SDP-o) for the SDP offer and (SDP-a) for the SDP answer in a)

e) **(2%)** Give the value of Content-Type in the message 200 OK (the final response to the INVITE) (see **Error! Reference source not found.**, page **Error! Bookmark not defined.** if needed).

Answer: Content-Type=sdp. Apart from the headers (which contains useful info), the content provided in 200 OK is SDP-a (which is of content type sdp)

f) **(8%)** Due to some error in the open source software used in the lab, the outcome of the test is that ACK is not sent from UA-A. (I.e. the SW did not behave according to RFC3261). Instead the following happened: At a later time (t5) the message ”Status: 200 OK with session description” is sent again from UA-B. You have 3 subtasks:

f1) Explain what the value of CSeq shall be in this message. You may assume that the value of CSeq in the INVITE sent from UA-A at time T1 is 1.

Answer: According to RFC3261: “The CSeq number is incremented for each new request within a dialog and is a traditional sequence number.”. But then one needs to know what is “a new request”. 200 is an answer to a request. Note that a CSeq is given with a method name (example: “CSeq: 314159 INVITE). In our case this answer is resent. RFC3261 further states “The CSeq header field of the response MUST equal the CSeq field of the request.” In short the answer is 1 (or in more general terms: “same CSeq as the method it is a response to” (if later in the call/dialogue UA-A will send a new INVITE (to modify som media) then the CSeq shall be increased) Those doing the java-part of task 4 should have a small advantage here, but I asked you all to look at CSeq values in task 1.

f2) Explain how this 200 OK message will be handled by the proxies.

Answer: This 200 shall be forwarded further on (optional supplementary info: i.e. according to the value in via-field). In other words: Proxies must be able to handle resending, (since the message may have been lost at a later stage on its return path)

f3) Explain briefly what *may* happen further. You may describe what *actually* happened in the lab, or what *should* happen if all entities comply with the principle of *Fault recovery* from ATN-book . *It is important that you write in your answer which of the two cases you describe*.

Answer1: What happened: On some SW the 2-way media was active even when ACK was missing, and this went on ‘for ever’. Further: resending of 200 took place as well, and also this seemed to be ‘for ever’. (Untill the call is finished by pressing red button /hanging up.

Answer2: What may have happened: Both UA-B and the proxies may have some timeout mechanism and return to the safe state (say idle-call). Optional answer: CANCEL or BYE may be used for this. NB: It is not possible for UA-B to start sending 486 or similar, since UA-B has already sent a final response 200, and a final response cannot change value. Suggesting use of 486 is wrong, but suggestion of CANCEL or BYE is not required 8especially not which of them that is correct)

g) **(2%)** What is the meaning of t= 0 0 in SDP? (You observed this in the lab)

Answer: ‘Ignore’ and use the times determined by the sequence /timing in the signalling protocol surrounding the SDP offer-answer, in this case the SIP messages (i.e. the t parameter is mandatory for historical reasons, indicating start and stop of multicast sessions etc., and the use of 0 0 means ignore, since it is not allowed just to skip this entity)

h) **(4%)** UA-B is starting to send RTP-data as soon as possible. (This is sometimes called "early media")

Your task is to explain what is the earliest possible time (relative to the timeline for UA-B) that UA-B may send media packets according to RFC3264 (the offer-answer model). You may mark this time with a * on an MSC or explain otherwise.

Answer: See * on answer to a) or say "just after t1 (the arrival of the INVITE)" Optional info: At this stage it may make sense to play "waiting music", (but I did not ask when it makes sense to the humans), I asked what RFC3264 says about the matter:

"Once the offerer has sent the offer, it MUST be prepared to receive media for any recvonly streams described by that offer. It MUST be prepared to send and receive media for any sendrecv streams in the offer, and send media for any sendonly streams in the offer (of course, it cannot actually send until the peer provides an answer with the needed address and port information). In the case of RTP, even though it may receive media before the answer arrives, it will not be able to send RTCP receiver reports until the answer arrives." (Citation from RFC3264). BUT note that UA-B cannot send media before it knows the port number to be used, which happen when the INVITE arrives at UA-B. This places the * just after t1.

i) **(3%) In a second test run with the same configuration and set-up you experienced a different test result, given by** Error! Reference source not found. **appendix ch.** Error! Reference source not found. **on page** Error! Bookmark not defined.**.**

Your task is to explain how this could happen, and why this is still in accordance with RFC3261 (SIP).

Answer: SIP is most often sent over UDP (at least so in our lab) and then messages may swop the sequence (messages may also be lost) The way RFC3261 (and IETF in general) draw their call flow has as an *implicit assumption* that lost messages and change in sequence must be catered for. This means that such resending is automatically to be compliant with the same call flow. And the same is true for swopped messages 8which is the case here) One example: UA-A at T2 is 'waiting for 180-ringing', but should be prepared to receive also later messages as long as they make sense (200 is a typical example which makes sense). The later 180 Ringing shall then just be ignored.

Exercise 3. (22%) About Sip-as-in-IMS and SIP-as-in-IETF

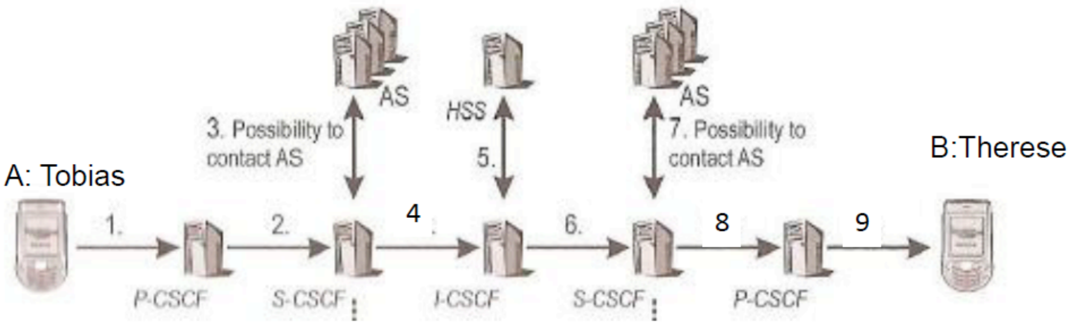


Figure 2 IMS Call setup sketch (from IMS book figure 2.7, slightly edited). Note that domain boarders are not shown here. In real life IBCF (i.e. border control functional units) may exist between the domains, but they are removed for simplicity.

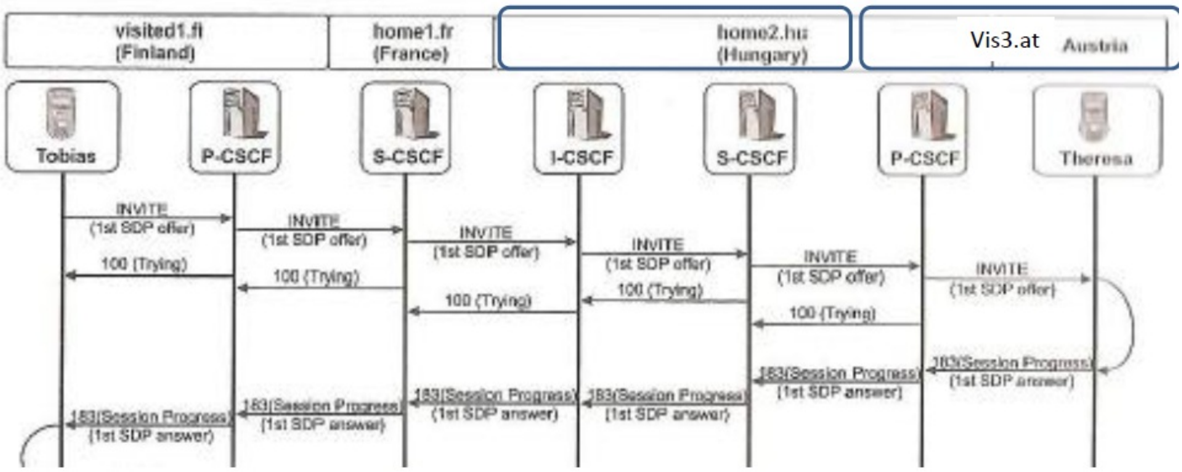


Figure 3 Based on IMS-book figure 12.1 showing parts of the call flow in the case of resource reservation in IMS. (The figure is slightly modified: Now Theresa is attached to a P-CSCF in Austria). Note that in this figure no AS is involved, but in the general case two AS may be involved (similar to what is shown in Figure 2). (HSSes are not shown)

Additional info added in the solution: The original figure from IMS book (12.1) assumes that Theresa is roaming with GPRS (IP-connectivity) home to Hungary, and hence showed that the P-CSCF as the first IMS entity was in Hungary. (My version of the figure above corresponds to left part of fig. 2.1. which should be more familiar to you. The original fig. 12.1 corresponds to the right part of fig. 2.1 (Theresa will then be using the mechanism shown in the left part of the right part of this figure….) BTW: Using IP-connectivity to the home, will cause complications for emergency calls, see fig. 3.32 where the local IP-CAN is involved). The right part of fig. 2.1 is assumed if the roaming-to/visited network does not support ISM, naturally in this case the visited network does not support IMS-emergency either.

a) **(4%)** Figure 3 illustrates the case when Tobias is roaming (to Finland) and Therese is roaming (to Austria). Your task is to illustrate the domain boarders on Figure 2 for 3 cases. **Use the drawing sheet on page** Error! Bookmark not defined. **to give your answer.**

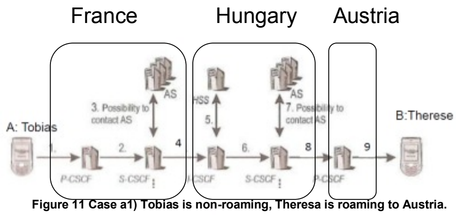


Figure 11 Case a1) Tobias is non-roaming, Theresa is roaming to Austria.

Short notation:

(.fr.) .fr .hu .at

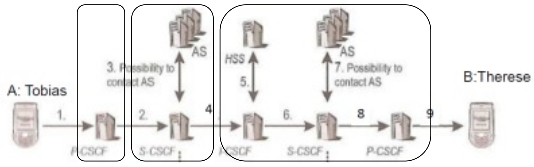


Figure 12 Case a2) Tobias is roaming to Finland, Theresa is non-roaming.

.fi .fr .hu (.hu)

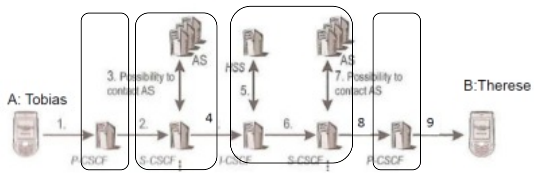


Figure 13 Case a3) Tobias is roaming to Finland, Theresa is roaming to Austria

.fi .fr .hu .at

Comment: IMS book shows The terminal as part of the Visited network. I prefer to say that endpoint is an own Domain (not important in the answer)

Answer:

b) **(4%)** Indicate at least one interface in IMS where SIP is not used. Which protocol is used on this interface?

Answer: Most 'natural' answer: Between I-CSCF and HSS Diameter is used. Additional info: If LSF is used before I-CSCF locates the HSS, then Diameter is the protocol used. Also some i/f towards charging/billing and resource reservation etc are non-SIP as well. The list of ref-points (ch. 2.3 offers even more answers).

c) **(4%)** In IMS the P-CSCF has several roles, such as doing compression of SIP over the radio interface and handle security association. List at least two more roles/tasks which is the responsibility of the P-CSCF in IMS.

Answer: * Detect Emergency calls, * handle QoS and * handle charging

d) **(6%)** Assume that an originating baguette-service (similar to the originating pizza-service from your lab assignment) shall be implemented in IMS using an AS (see Figure 2). The task is to rewrite my-baguette@... to the SIP URI for the baguette-kitchen that is the current favourite of Tobias[1]. There are two AS in Figure 2.
d1) Explain which of the AS in Figure 2 that is the natural place to implement this service.

Answer: The AS connected to the home network of Tobias, i.e. AS connected to the left S-CSCF. Additional info: In this case the name of the favourite baguette shop of Tobias may reside inside the HSS; or in the AS. (Or the AS may be separated into service control and service data: SDF and SCF in a similar way as in IN)

d2) Explain some problems or issues that may occur in case Tobias is roaming to Norway.

Answer: This is a service making sense only in a certain area, since there is a physical good that shall be delivered. Dialling 02323 (Oslo Taxi) does not make much sense while you are in Trondheim or when you are roaming to France. (Even if oslo taxi

[1] Of course this shall only be done in case the caller is in fact Tobias, as other customers may have other baguette favourites.

had an ordinary routable number like 23232323 dialling +4723232323 does not make much sense in Paris, you will end up talking to a taxi-driver in Oslo, but it does not make sense for him to pick you up). In a similar way Tobias may reach a baguette-kitchen in Paris, but having a baguette brought to Norway may make less sense. (Other sensible answers may be credited as well)

e) **(4%)** In the lab assignment you only used one SIP-proxy, and the terminating pizza service was implemented directly on this proxy. Show a sketch of how the terminating pizza-service may be located in an AS in the IMS system. You should indicate clearly what type of CSCF this AS shall be connected to, and which domain this CSCF shall be located in. (You may look at Figure 2 for support).

Answer: The AS connected to the home network of the baguette company (which will be the terminating side in this case, i.e. AS connected to the right-most S-CSCF.

Additional comments: If a baguette-kitchen like "Euro-baguette" wants to have kitchens all over Europe, they may choose a global IMS-provider (like Vodaphone or similar), or they may choose Small-IMS-Norway as their IMS-provider, as long as basic ISM roaming agreements are in place to the rest of Europe. In such a setting the (fixed) baguette kitchen in paris will be considered as a roaming IMS-customer having Small-IMS-Norway as their home network). This is a bit similar to when a company on the internet uses a web-server located in another country.

Exercise 4 (20%) On GSM and mobility

- Carla is a mobile subscriber of GSM in Norway having a Norwegian SIM card
- Carla's GSM-phone-number in Norway is 90 90 90 90.
- Carla's subscription is having IMSI number with the value IMSI-C.
- N1-fix is a fixed subscriber in Norway, on an old-fashioned phone not having an electronic address book, and not having any means to send the '+' sign.
- N2 is an MSISDN-number in Sweden, N2 is currently 'at home' (i.e. non-roaming)
- Let MCC-C be the mobile country code part of IMSI-C; MNC-C be the mobile network code of IMSI-C; and MSIN-C be the mobile subscriber identification number part of IMSI-C.

Additional info which *may* be useful: Country code for Norway is 47 and for Sweden it is 46. The international prefix in Norway is 00, so a fixed subscribed in Norway dialling a fixed subscriber in Stockholm, Sweden will have to dial 00 46 8 xxxxxxx (area code 8).

a) **(3%)** Assume that Carla is visiting Sweden and having the mobile phone on (i.e. Carla is roaming and already registered in Sweden with her GSM phone).

What is the phone number that a user Arne on the fixed device with number N1-fix in Norway should dial to establish a voice call to Carla?

Answer: The name context of the caller (N1-fix) is Norway, and the callee is within the same name context, so **90 90 90 90 is an answer** (or 00 47 90 90 90 90, which is legal, but which may be a bit 'suspicious' (telco may assume some fraud is going on). I dialled 0-00-47-97 etc. from my ntnu office and reached my own mobile.

b) **(3%)** Assume that Carla is still in Sweden. What is the phone number a user Anna on the mobile device having phone number N2 in Sweden must dial to establish a voice call to Carla?

Answer: The name context of the caller (N2) is Sweden, since Anna is currelty at home in Sweden, and the callee is NOT within the same name context, so 00 47 or

+47 must be added. Answer is +47 90 90 90 90 or 00 47 90909090 (00/+ means: out of Sweden (to international level) and then 47 signifies Norway).

Note that the fact that Carla is in Sweden does not matter! However, if Anna was in Norway she could have dialled 90909090 (just as N1-fixed in Norway)

NB: Carla does not have a permanent MSRN in Sweden, and N2 will definitely not be able to dial such an MSNR. (the MSRN will indicate Sweden as country code, but if N2 gets to know such a number, then Carlas location is revealed (at a high level but still…). This is agains all privacy rules in GSM.

c) **(6%)** Explain the role of IMSI, MSISDN and MSRN as names and identities in GSM regarding mobility. You should cover both *registration phase* and the case of an *incoming call* to this mobile user**.** You may use free text of your own, or you may rely on any of the figures in appendix ch. **Error! Reference source not found.** (page **Error! Bookmark not defined.**) to support your explanations.

Answer: Når noe ringer til en mobil: (To be translated)

The short answer is: Based on MSISDN (/diractyry/dialled number) the right contry is identifies and the call will end up in an MSC which can figure out which HLR to ask in order to route the call further to the actual location of the GSM-subscriber/terminal. So MSISDN is used to route call to an MSC and after a lookup in HLR an MSRN is returned. The call is then further routed based on this MSRN.

During registration the IMSI is used, MSC is able to route to the HRL based on the country code and the MNC-part of the IMSI. Inside the HLR the identity-part of the IMSI is used to identify the customer, perform security etc (no details requested for security)

d) **(4%)** Carla is now activating "call forwarding no reply" (CFNR) to her parents phone number +47 22123456, and then switching off her phone for the evening.

Task: Explain *briefly* what happen when Carla switches her phone *on* again to register. You need not show all details, but you should indicate some *location related data or identities* which are held by HLR and VLR/MSC respectively after the registration is finished.

Answer: HLR holds a pointer to VLR and the VLR holds a pointer to the HLR. [In addition the VLR holds a TMSI-value, which is shared with the MS.]

e) **(4%)** The VLR (or the combined VLR/MSC) will also keep some data related to supplementary services, and these data will also be sent during the registration phase. (see "service para" in **Error! Reference source not found.** on page **Error! Bookmark not defined.** in the appendix).

Task: Give *two* examples of supplementary services related to voice calls, and for each of them indicate what kind of "service para" they may need. At *least one* of your service examples must be related to an *originating* call.

You shall stick to the standardized supplementary services, and you *need not* look into CAMEL. You shall *not* consider SMS and GPRS, *only* voice.

Answer: There are several services one may list here:

Originating service examples: (CLIR is tha 'natural answer' based on the lectured material)

Example 1: CLIR. CLIR needs data like "on/off" in order so set the P-bit correctly but not any other parameters

Example 2: barring services (sperr-tjenester). Like disallow all outgoing calls or disallow all 8xx-numbers except those that are for free, or similar… (OCB Outgoing Call Barring is listed in ETSI teddi)

Barring services need some info on which categories of calls that shall be barred (how this data look is outside of the scope)

Sperr noen nr serier slike som 829 etc. / sperr alle utgående tjenester/sperr alle utgående samtaler til utlandet (It is a bit unclear how many OCB-categories what is defined, ant to what extent they are all standardized.)

Term. Service examples: There are several services one may list here (CFNR is 'natural choice')

Example 1: CFNR . parameters for CFNR will be on/off and forwarding-to-number for CFNR (optionally how long the timer shall be). My new phone allow a GUI for me to set this parameter, BUT my network does not allow me to choose this. It will be up to the visited network whether it allows enduser customization of this timer in CFNR or not)

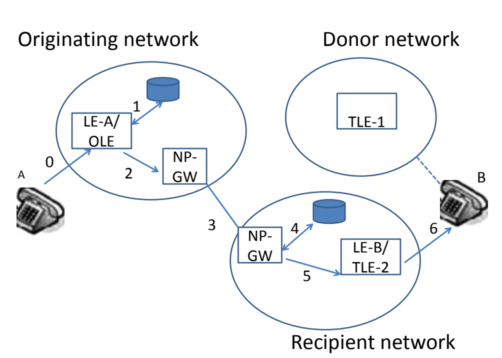
Example 2: CFOut of coverage (formal name: SS-CFNRc Call Forwarding on Not Reachable Supplementary Service)

Example 3 [optionally: CLIP, but in GSM this service seems to be part of the standard, and not being a 'supplement').CLIP needs only on/off as service-para]

Exercise 5 (10%) On number portability (NP) in Norway

Norwegian Post and Telecommunications Authority www.npt.no (norsk: Post- og teletilsynet, PT) defines 3 types of portability.

The procedure AcQ (meaning All call Query) is defined by ntp (PT) as the standard procedure in PSTN for service-provider portability. See Figure 4.



Further details of step 3 is explained in appendix ch. **Error! Reference source not found.** page **Error! Bookmark not defined.**.

Figure 4 AcQ procedures as defined by npt (PT) for service provider portability.

a) **(4%)** Explain the concept *donor network* and *recipient network* for service provider portability. You may use an example to illustrate the concepts.

Answer: Donor network is the network where the customer belonged (had his subscription) before the NP took place. And recipient network is the new network where the customer currently (after NP) has his subscription.

b) **(4%)** *Service provider portability* makes sense for both PSTN networks and GSM networks.

Task: Give a definition of *Service provider portability* which works as intended for both PSTN and GSM.

Answer: To keep the same phone number (E.164 number/directory number) when you are changing service provider (changing where you have your subscription/gets your bills from).
Additional information ("flisespikkeri): This definition works as long as the customer keep the technology type. I.e. a customer on PSTN may choose another PSTN op, and a customer on GSM may choose another GSM-operator. BUT may NOT port a PSTN number/subscription to a GSM-subscription or vice versa.

ETSI Teddi offers this short definition:

- ability for a customer (subscriber) to change service provider while retaining the same number (TIPHON doc.numbers)

ETSI also offers the following definition (which covers the details/flisespikkeri):

Service Provider Portability for Non geographic Numbers (NGNP) Service Provider Portability for NGNP is a service that enables customers to resign their subscription with a Service Provider (Donor) and to contract another subscription with another Service Provider (Recipient) without changing their Non-geographic Number, and without changing the nature of the service offered.

Here we note that the terms "without changing the nature of the service offered" which may imply that customer must stay within the same type of system (such as inside GSM)

c) **(2%)** In GSM in Norway *Service provider portability* is common. Which of the numbers MSISDN and IMSI are unchanged, after the subscriber Carla is asking for service provider porting from a GSM-operator with mobile network code MNC-1, to another GSM-operator with MNC-2. Justify your answer.

Answer: By definition of NP the E.164 number/directory number (i.e. the MSISDN-number) is kept unchanged. One can also deduce that the IMSI-number MUST change since the MNC-part of the IMSI is changed from MNC-1 to MNC-2. (You will get a new SIM-card as well), and for this reason the other is unchanged since the text says that one is unchanged. Candidates did not need to know any details of NP in GSM to answer this task (such details has not been lectured, but some students has asked about this, as it is very common in Norway)

[Additonal info: For GSM network service provider portability also works for so-called "virtual mobile operators" (VMOs), but since we have not looked into the details of VMOs you shall not worry about this. For this task you may assume that each GSM-op has a complete network with BS, BSC, VLR/MSC, GW-MSC and HLR. You shall assume that all mobile subscribers are attached to the access network of their home network as long as they are located in Norway (i.e. the usual non-roaming case as lectured)]

Additional info on GSM exercise 4c)

Case1 caller er selv mobil og tilknytta en MSC. Denne vil analysere MSISDN-nummeret.

Case 1 subtilfelle 1: I en del tilfeller vil den selv vite at dette er et mobilnr, og vil være i stand til å slå opp i rett HLR (eller rute til en annen MSC som kjenner til dette i detalj (adressen til HLRen trengs jo) . Anropet fortsetter da med de meldingene som er etter GW-MSC i figurene i appendix.

Case1 1 subtilfelle 2) er at MSC'ene i mobilnettet ikke forstår at dette er et mobilnr, f.esk. fordi det starter med +39 (noe) og de kjenner ikke til den interne nrplanen i landet med landsprefix 39. De ruter det derfor til fastnettet og det hele fortsetter som om anropet hele tiden var en anrop fra fastnettet.

Case 2: Fra fastnettet På et tidspkt. i b-nr analysen basert på MSISDN kommer man til et pkt i fastnettet som 'forstår' (via sin bnr analyse) at dette er er et GSM mobilnummer, og som peker til en GW-MSC. GW-MSC vil så forespørre en HLR. (uten nr-portablilitet kan vi anta at den forespør den rette HLR direkte, som vist i figurene i append. Uten å bry oss om DB for NP etc)

Etter at HLR er forespurt: da benyttes i alle tilfellene MSRN for videre B-nr analyse**.** Svaret den får fra HLR er et MSRN, et rutbart nummer til den MSCen der abb er lokaliset (dette kan være i et annet land om man roamer), og videre bnr analyse vil bringe oss dit. (der er det en paging-procedyre med Temp.number etc på radia-delen med det er det ikke spurt etter her)

NB: Vi har ikke gjennomgått NP for GSM, så de behøver ikke tenke på prosedyrer som AcQ eller de andre prosedyrene for NP i fastnettet (som jo vil ha sine analogier i GSM-nettet).

From the Q&A material on It's learning the following is listed (the question was almost the one given on exam "How does routing to a mobile GSM phone take place?" I hope you all read it!

The MSISDN categories follow the international ISDN number plan and therefore have the following structure.

- **Country Code (CC) :** Up to 3 decimal places.
- **National Destination Code (NDC):** Typically 2-3 decimal places.
- **Subscriber Number (SN):** Maximum 10 decimal places.

NDC correspons to 'area code' but as we all know the numbering structure in Norway is flat (8 digits, no area code in use)

All MSCs in the country will know NDCs in its own country (no need to know this in Norway, but in the general case)

All MSCs in other countries will route based on the country code (not reading further digits)

In practice this means routing to PSTN via transit nw and to the correct country, where it will end up in a GW-MSC, which will recognize this number as a GSM number (based on the fact that 73'series in fixed Trondelag, while 90'series and 40'series in Norw. are for GSM)

In this way "tromboning" may happen, a Norw GSM subscr. (+47 40404040) in beach in Greece calling a Norw. friend (+47 90909090) also on the beach will have the call routed based on analysis of +47 only (routing to int-exch. in PSTN), then to Norw. and only when arriving in Norw. the 90'part of the number is read and analyzed. (90 will indicate GSM-number inside Norway) After this an MSRN indicating Greece will be received.

Greece- - transit - -> Norw
Greece < - -transit - - Norw,

And in worst case your friend on (+47 90909090 activates CFNR back to Norwegian voice mail box.
Greece- - - -> Norw