

**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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|--------------------------------------|---|
| Faglig kontakt under eksamen: | Lill Kristiansen / Md. Qasim Khan |
| Tlf.: | 97 72 72 27 |
| Eksamensdato / Date: | 25 mai 2019 / May 25th 2011 |
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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 (authoritative) (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 (12- 15 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

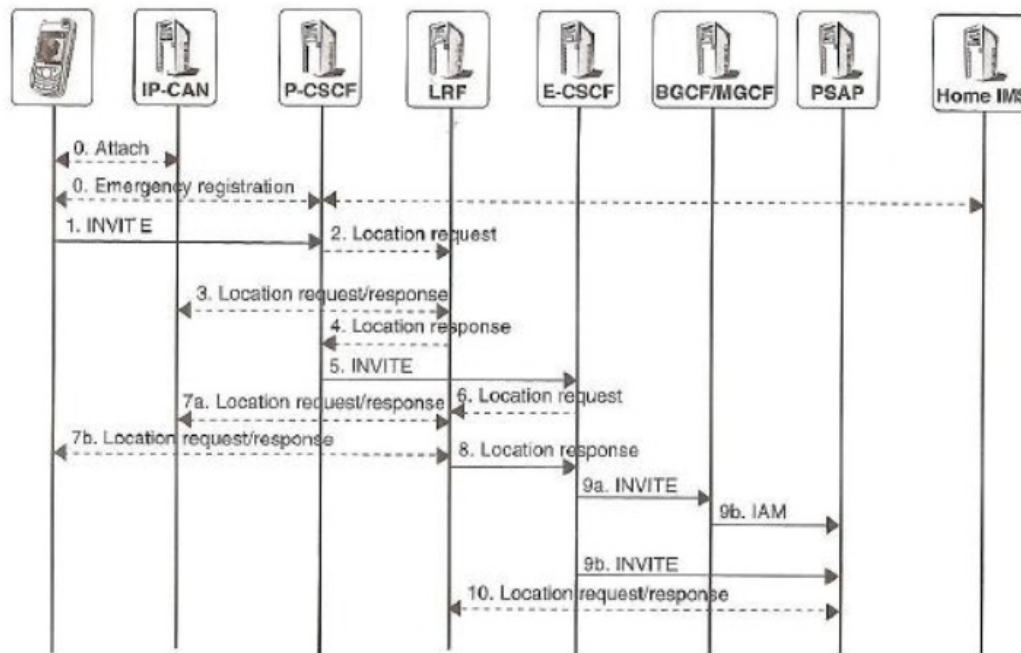


Figure 1 IMS emergency calls according to the IMS-book.

(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.

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

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. 2 page 12, and the results as seen in Wireshark are shown on page 13.

- a) (6%) Look at the results from test run 1 given in append. ch. 2.4 page 13. 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 Figure 9 on the drawing sheet on page 16 for your answer.**
- 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.
- 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).
- 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)
- e) (2%) Give the value of Content-Type in the message 200 OK (the final response to the INVITE) (see Table 3, page 13 if needed).
- 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.
 - f2) **Explain** how this 200 OK message will be handled by the proxies.
 - 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.*
- g) (2%) What is the meaning of t= 0 0 in SDP? (You observed this in the lab)
- 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.

- i) (3%) In a second test run with the same configuration and set-up you experienced a different test result, given by Table 4 appendix ch. 2.5 on page 13.

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

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

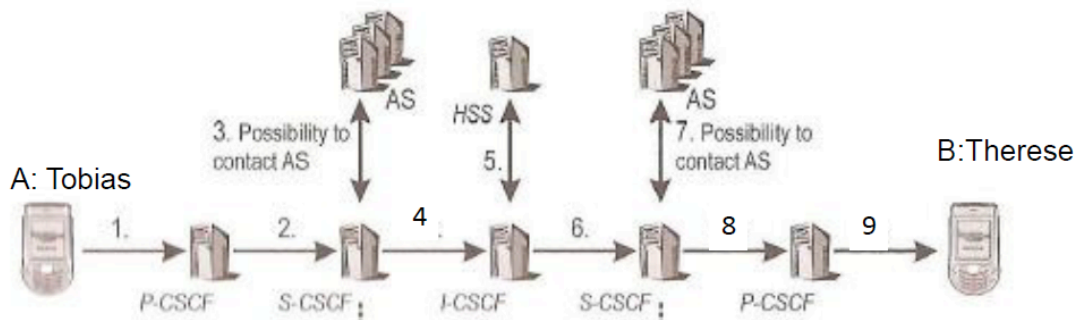


Figure 2 IMS Call setup sketch (from IMS book figure 2.7, slightly edited). Note that domain borders are not shown here. In real life IBCF (i.e. border control functional units) may exist between the domains, but they are removed for simplicity.

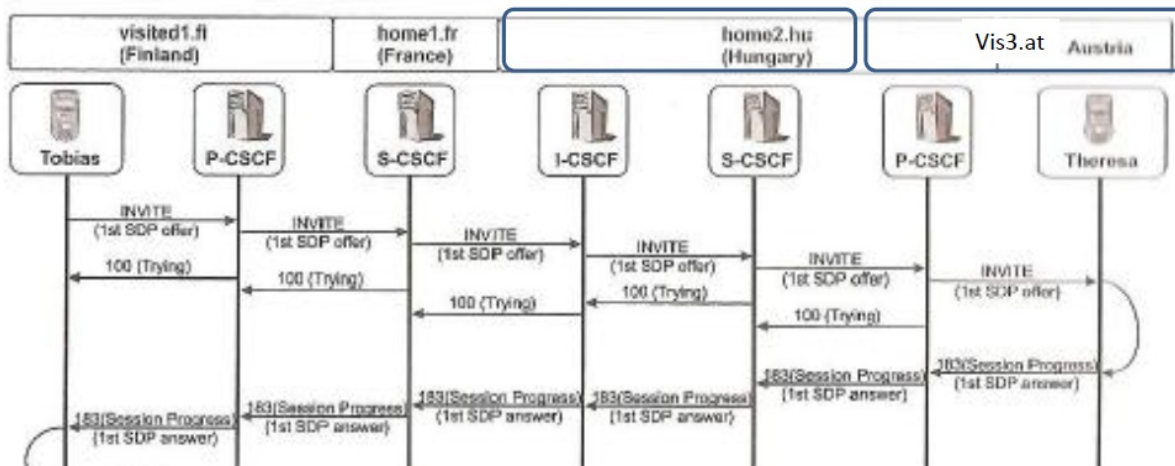


Figure 3 Based on IMS-book figure 12.1 showing parts of the call flow in the case of resource reservation in IMS. (The figure is slightly modified: Now Theresa is attached to a P-CSCF in Austria). Note that in this figure no AS is involved, but in the general case two AS may be involved (similar to what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

- 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 borders on Figure 2 for 3 cases. Use the drawing sheet on page 17 to give your answer.
- b) (4%) Indicate at least one interface in IMS where SIP is not used. Which protocol is used on this interface?
- 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.

- 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 Tobias¹. 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. **d2) Explain** some problems or issues that *may* occur in case Tobias is roaming to Norway.
- 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).

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 subscriber in Norway dialling a fixed subscriber in Stockholm, Sweden will have to dial 00 46 8 xxxxxxxx (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?

- 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?
- 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. 4 (page 15) to support your explanations.
- 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.

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

- 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 Figure 8 on page 15 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.

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 npt (PT) as the standard procedure in PSTN for service-provider portability. See Figure 4.

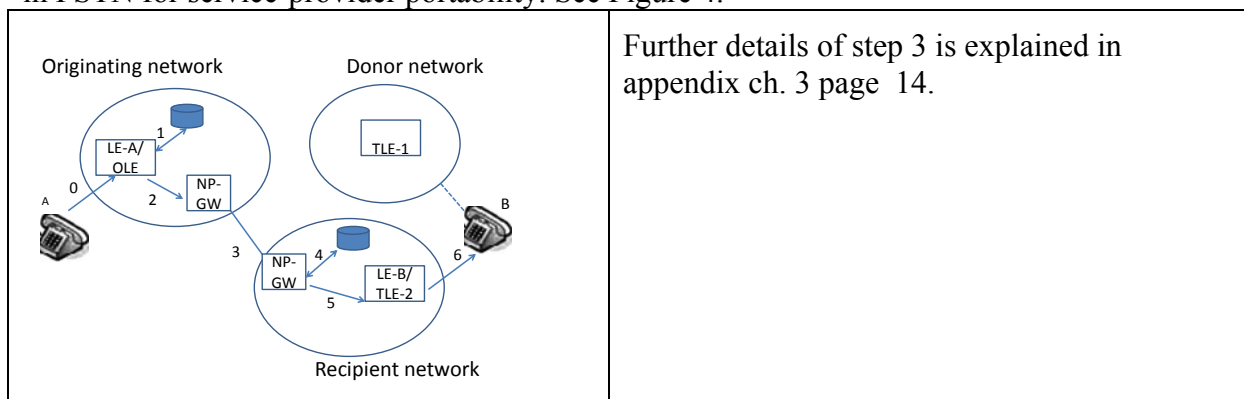


Figure 4 AcQ procedures as defined by npt (PT) for service provider portability.

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

- 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.

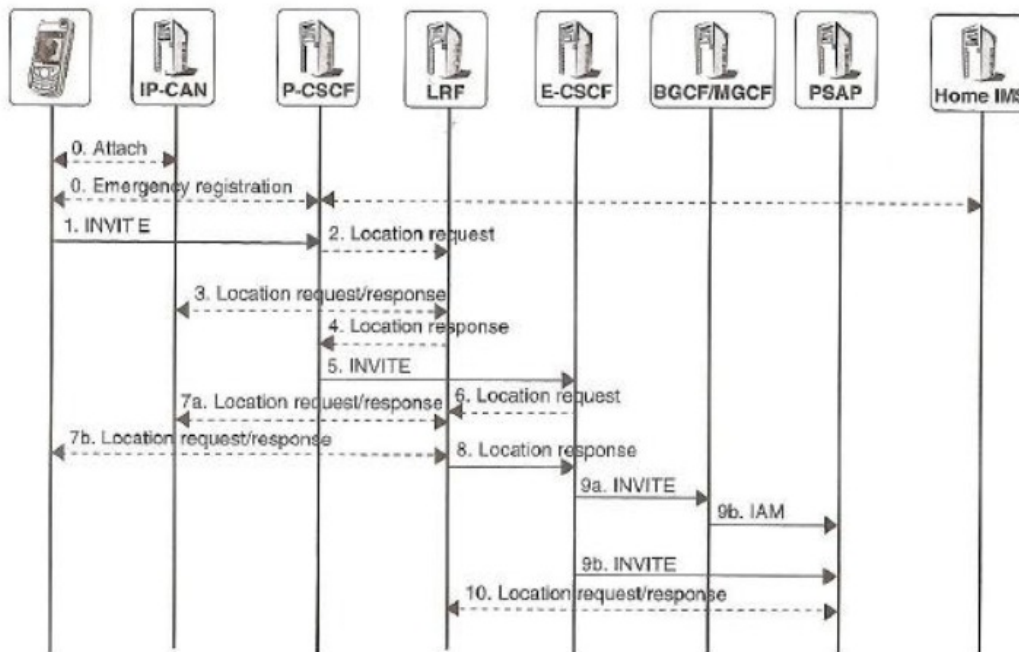
[Additional info: For GSM network service provider portability also works for so-called “virtual mobile operators” (VMOs), but since we have not looked into the details of VMOs you shall not worry about this. For this task you may assume that each GSM-op has a complete network with BS, BSC, VLR/MSC, 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)]

Du bør starte med å **lese gjennom alt materialet** og så bestemme deg for den rekkefølgen du ønsker å besvare oppgavene i.

Appendikset kan inneholde nyttig informasjon. Gjør egne antakelser hvis teksten er uklar eller informasjon mangler.

Helt bakerst finner du noen ark med diagrammer, som du er bedt om å bruke i noen av oppgavene.

Oppgave 1 (10%) Om nødanrop i IMS og i PSTN



Figur 1 IMS nødanrop i hht IMS-boka.

(Den uformelle notasjonen med a og b betyr alternative valg, og alle stiplede linjer er valgfrie/ optionelle)

- a) (4%) Figur 1 viser et utgående anrop fra IMS der det initieres et nødanrop. Hvis vi velger å følge 9b in Figur 1 så ender vi opp i en en PSAP (nødansentral) som er tilknyttet IMS. Forklar for dette tilfellet hvordan lokasjonsinformasjon fra en GPS-enhet på den håndholdte enheten til venstre kan bringes til PSAP *uten* at meldingene merket med 6 og 10 brukes.

Før resten av oppgave 1 skal du studere nødanrop der både innringer ("caller") og nødsentralen er tilknyttet PSTN.

- b) (3%) Forklar hvordan et nødanrop er rutet i PSTN. Du kan anta at 113 er ringt (dette er det medisiske nød-nummeret i Norge).
- c) (3%) Forklar hvordan nødsentralen finner ut lokasjonen til innringeren ("caller") i dette PSTN-tilfellet.

Oppgave 2 (38%) Om SIP og SDP (omtrant slik det var i labben)

Din jobb i denne oppgaven er å arbeide med noen data som kunne vært ekte data fanget med bruk av Wireshark i labben. Oppsettet, forbetninger og testen er alle beskrevet i appendikset kap. 2 page 12, og resultatene slik de ser ut i Wireshark er vist på side 13.

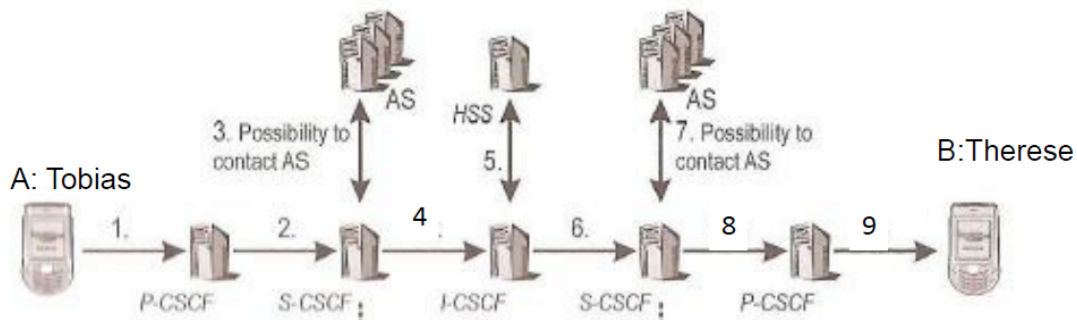
- a) (6%) Se på testkjøring-1, og resultatene gitt i appendiks kap. 2.4 side 13. Du skal bruke din kunnskap om SIP slik det er definert i RFC3261 til å tegne ett meldings-sekvens-diagram (MSC) som viser *hele* signalleringen, men alle meldinger utvekslet i det gitte tidsrommet mellom de 4 involverte enhetene (2 proksier og 2 brukere UA'er). (Du trenger ikke å vise DNS). Bruk meldingene fra Wireshark-observasjonene og suppler med andre meldinger der det ikke er noen Wireshark-observasjoner. Ditt MSC skal inneholde 4 vertikale "tidslinjer", en for hver av UA-A, UA-B, Proxy-A og Proxy-B. **Nyttigjør deg av tegnearket på side 17 når du svarer på denne oppgava.**
- b) (5%) Fullfør samtaleoppsettet, slik det skal bli når alt virker bra og i henhold til spesifikasjonen. Illustrér uformelt mediestrømmen og hvor signaleringen går.
- c) (4%) Illustrer i det samme MSC eller i en separate figur hva som skjer etter 2 minutters samtale når Bob bestemmer seg for å "legge på" (dvs. når Bob trykker rød knapp på sitt brukergrensesnitt).
- d) (4%) Hvilke(/n) melding(/er) fra testkjøring-1 inneholder SDP *tilbud (offer)*, og hvilke(/n) melding(/er) inneholder SDP *svare (answer)*? (Termene *offer/answer* er i henhold til terminologien i RFC3264) (Du kan henvise til deloppgave a) her)
- e) (2%) Angi verdien på Content-Type i meldingen 200 OK (det endelige svaret (final response) på INVITE-meldingen). (Se evt. Table 3 s. 13 ved behov.)
- f) (8%) På grunn av en feil i den åpne kildekoden som var brukt i labben, så var resultatet av testen at ACK ikke blir sendt fra UA-A. (dvs. at programvaren ikke oppførte seg i hht. RFC3261). Isteden skjedde følgende: På et seinere tidspunkt (t5) blir meldingen "Status: 200 OK with session description" sendt om igjen fra UA-B. Du har nå 3 deloppgaver:
 - f1) **Forklar** hva verdien på CSeq skal være i denne meldingen. Du kan anta at verdien på CSeq i INVITE sendt fra UA-A ved tid T1 er 1.
 - f2) **Forklar** hvordan denne 200 OK meldingen vil bli håndtert av proksiene.
 - f3) **Forklar** kort hva som siden kan skje. Du kan beskrive hva som *faktisk* hendte i labben, eller hva som *burde* skje videre hvis alle enhetene følger prinsippet om feil-oppretting (*Fault recovery*) fra ATN-boka. *Det er viktig at du skriver i svaret ditt hvilket av de to tilfellene du beskriver.*
- g) (2%) Hva betyr $t=0$ i de SDP meldingene du observerte i labben?
- h) (4%) UA-B starter å sende RTP-data så snart som mulig. (Dette kalles noen ganger "tidlig media"/"early media")

Din oppgave er å forklare hva som er tidligste mulige tid (relativ tid på tidslinjen til UA-B) at UA-B kan sende media pakker i hht RFC3264 (the offer-answer model). Du kan markere dette med en * på et MSC eller på annen måte.

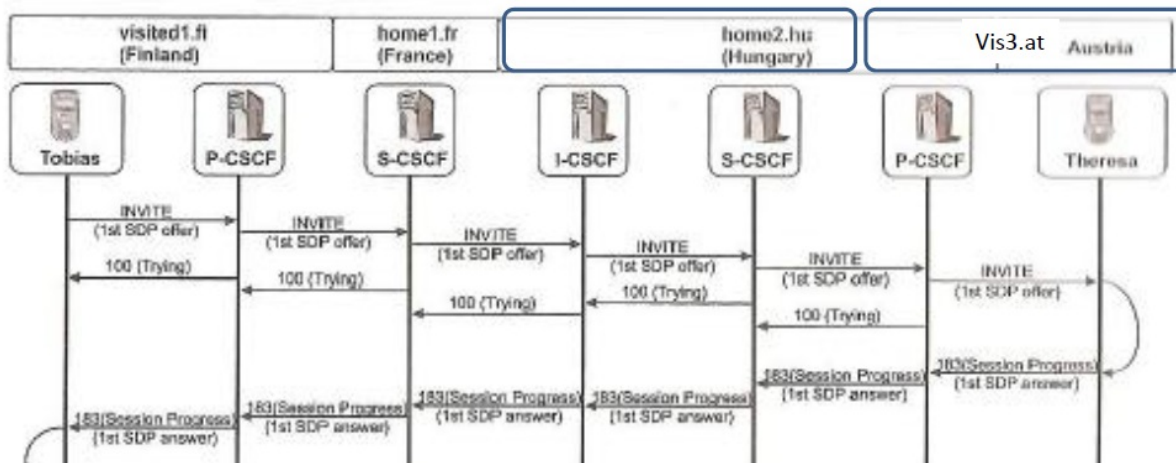
- i) (3%) I en testkjøring nummer to med samme oppsett og konfigurasjon opplever du at resultatene på PC-1 (UA-A) er gitt ved Table 4 i appendikset, (kap2.5 side 13).

Din oppgave er å forklare hvordan dette kunne skjer, og hvorfor dette resultatet også er i henhold til RFC3261 (SIP).

Oppgave 3. (22%) Om Sip-som-i-IMS and SIP-som-i-IETF



Figur 2 IMS skisse av samtaleoppsett (fra IMS boka figur 2.7, lettere endret). Merk at domenegrenser ikke er vist her. I virkeligheten vil IBCF (border control function enheter) kunne eksistere mellom domeneene, men for enkelhets skyld er de ikke vist.



Figur 3 Basert på IMS-boka figur 12.1 viser vi deler av samtaleoppsettet i det tilfellet at det er ressursreservering i IMS. (Figure er lettere modifisert: Nå er Theresa tilknyttet en P-CSCF in Østerrike). Merk at i denne figuren vises ingen AS, men i det generelle tilfellet kan to AS være involvert (i likhet med det som vises i Figur 2). (HSSer vises ikke)

- a) (4%) Figur 2 illustrerer tilfellet at Tobias "roamer" (til Finland) og at Therese "roamer" (til Østerrike). Din oppgave er å illustrere domene-grenser i Figur 2 for 3 tilfeller. **Bruk tegne-arket gitt på side 17 i ditt svar.**
- b) (4%) Indikér minst ett grensesnitt i IMS hvor SIP ikke er i bruk. Hvilken protokoll er i bruk her?
- c) (4%) I IMS har P-CSCF'en flere roller, bl.a. å komprimere SIP på radiogrensesnittet og håndtering av sikkerhetsassosiasjoner. Angi *minst to andre roller/oppgaver* som P-CSCF er ansvarlig for i IMS.

- d) (6%)** Anta at en originerende baguette-tjeneste (tilsvarende til den originerende pizza-tjenesten i lab-øvingen) skal bli implementert i IMS ved hjelp av en AS (se Figur 2). Tjenesten skal omskrive my-baguette@... til SIP URIen for det baguette-kjøkkenet som er den nåværende favoritten til Tobias². Det er to AS'er i Figur 2. **d1) Forklar** hvilken AS i Figur 2 som er det naturlige stedet å implementere denne tjenesten. **d2) Forklar** noen problemer ("issues") som *kan* forekomme i det tilfellet at Tobias roamer i Norge.
- e) (4%)** I lab-øvingen bruke du bare én proksi-tjener, og den terminerende pizza-tjenesten var implementert direkte på denne proksien. Vis en skisse av hvordan den terminerende pizza-tjenesten kan bli lokalisert i en AS i et IMS system. Du skal indikere tydelig hva slags type CSCF denne AS skal knyttes til, og hvilket domene denne CSCF'en skal være lokalisert i. (se evt. på Figur 2 som en hjelpefigur).

Oppgave 4 (20%) Om GSM og mobilitet

- Carla er en mobilabbonent i Norge (GSM), med et norsk SIM-kort.
- Carlas mobilnummer i Norge er 90 90 90 90.
- Carlas abonnement har IMSI-nummer med verdi IMSI-C.
- N1-fix er et fastabbonent i Norge som har en gammeldags telefon uten elektronisk adressebok og uten muligheter til å sende '+'-tegnet.
- N2 er et MSISDN-nummer i Sverige, som for tiden er 'hjemme' (ikke-roamende)
- La MCC-C være "mobile country code"-del av IMSI-C; MNC-C være "mobile network code"-del av IMSI-C; og MSIN-C være "mobile subscriber identification number"-del av IMSI-C

Tilleggsinformasjon som *kan* være nyttig: Landskoden for Norge er 47 og for Sverige er den 46. Det internasjonale prefikset i Norge er 00, så en fast-abbonent i Norge som skal ringe en fastabbonent i Stockholm, Sverige vil måtte slå 00 46 8 xxxxxxx ("area code": 8).

- a) (3%)** Anta at Carla besøker Sverige og har mobiltelefonen på (dvs. at Carla er roamende og allerede registrert i Sverige med sin GSM-telefon).
Hva er telefonsnummeret som en bruker Arne på en fasttelefon med nummer N1-fix i Norge må slå for å etablere en telefonsamtale (med tale) til Carla?
- b) (3%)** Anta at Carla fortsatt er i Sverige. Hva er telefonnummeret en bruker Anna på mobiltelefonen med nummer N2 i Sverige må slå for å etablere en telefonsamtale (med tale) til Carla?
- c) (6%)** Forklar rollene til IMSI, MSISDN og MSRN som navn og identiteter i GSM i forhold til mobilitet. Du skal dekke både *registreringsfasen* og tilfellet med et *innkommende anrop* til denne mobile brukeren. Du kan bruke egen fritekst, eller du kan basere din forklaring på en av figur i appendikset kap. 4 (side 15) for å støtte din forklaring.
- d) (4%)** Carla aktiverer så "viderekobling ikke svar" ("call forwarding no reply" /CFNR) til sine foreldres telefonnummer +47 22123456, og slår så av telefonen for kvelden.
Forklar kort hva som skjer når Carla slår telefonen på igjen for å registrere seg. Du trenger ikke vise alle detaljer, men indiker noen *lokasjonsrelaterte data eller identiteter* som holdes av hhv HLR og VLR/MSC etter at registreringen er ferdig.

² Selvfølgelig skal dette utføres bare i det tilfellet at ringeren (caller) faktisk er Tobias, siden andre kunder kan ha andre preferanser i baguetter.

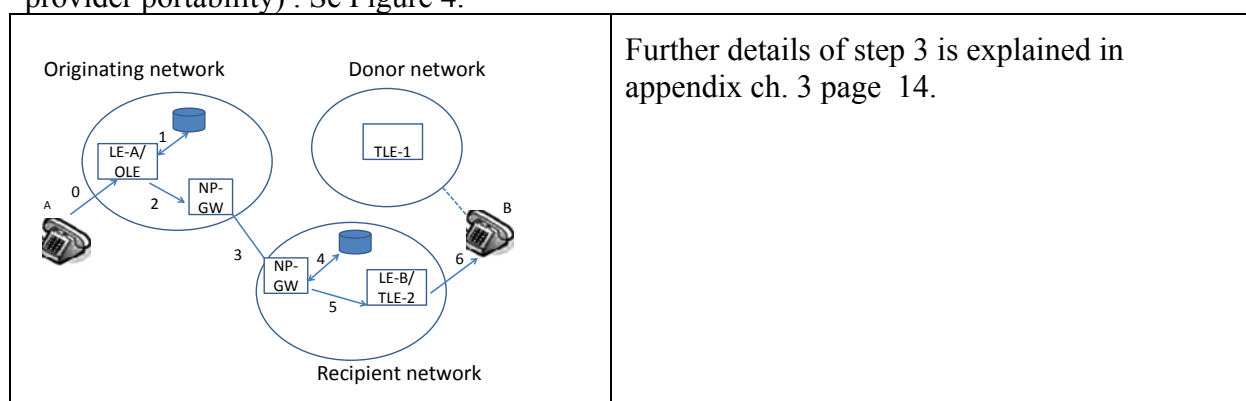
- e) (4%) VLRer (eller den kombinerte VLR/MSC) vil også holde noe data relatert til supplementære tjenester, og disse data vil også bli sendt i løpet av registreringsfasen. (“service para” is shown in Figure 8 on page 15 in the appendix).

Gi to eksempler på supplementære tjenester relatert til taleanrop, og for hver av dem indiker hva slags tjenesteparametre (“service para”) de kan trenge. *Minst en* av tjenestene du nevner må være knyttet til et utgående anrop (*originating call*).

Du skal holde deg til de standardiserte supplementære tjenestene, og du *trenger ikke* å omtale CAMEL. Du skal ikke se på SMS and GPRS, *kun* tale.

Oppgave 5 (10%) Om nummerportabilitet (NP) i Norge

Post- og teletilsynet (PT) definerer 3 typer portabilitet. Procedyren AcQ (“All call Query”) er definert av PT som standardprosedyren i PSTN for tjenesteleverandørportabilitet (service-provider portability) . Se Figure 4.



Figur 4 AcQ prosedyren slik den er definert av PT for tjenesteleverandørportabilitet (service provider portability).

Figure 5

- a) (4%) Forklar begrepene *donornett* (donor network) og *mottakernett* (recipient network) for tjenesteleverandørportabilitet. Du kan benytte et eksempel til å illustrere konseptene.

- b) (4%) *Tjenesteleverandørportabilitet* gir mening både i PSTN og i GSM.

Task: Gi en definisjon av *Tjenesteleverandørportabilitet* (*Service provider portability*) som fungerer som intendert både for PSTN og GSM.

- c) (2%) I GSM i Norge er *tjenesteleverandørportabilitet* vanlig. Hvilket av nummerene MSISDN og IMSI forblir uendret etter at abonnenten Carla får utført *tjenesteleverandørportering* fra en GSM-operatør med MNC-1 som ”mobile network code”, til en annen GSM-operatør med MNC-2 som ”mobile network code”. Grunngi svaret.

[Tilleggsinfo: For GSM nett så vil begrepet *tjenesteleverandørportabilitet* også gi mening for såkalte ”virtuelle mobiloperatører” (VMO), men siden vi ikke har sett på VMO i detalj i kurset, så skal du ikke bekymre deg om den detaljen. Her kan du anta at hver GSM-operatør har et komplett nett med både BS, BSC, VLR/MSC, GW-MSC og HLR. Du skal anta at alle mobile abonnenter er tilknyttet aksessnettet hos sin egen operatør så lenge de er i Norge (altså det vanlige ikke-roamende caset slik det er forelest).]

Appendix

1. Fault recovery principle from Audestad's book.

Ch. 6.1.6 in the ATN-boook lists 5 system requirements to all telecommunication systems. In this exam the task is to look at the last one of these, which is described as follows:

- “*Fault recovery*. Each process must be able to recover from fault on each own. If a process fails, all processes depending on it must be able to return to a safe state independent of the faulty process. “

2. Information related to exercise 2

2.1. Set up in the test cases reported

We are using 4 machines all running relevant SIP software, and we follow the traditional setup called “SIP trapedoid”. We used wireshark only on two of these machines, as detailed in Table 1.

Table 1 Set-up and configuration during the test cases

| | PC-1 | PC-2 | PC-3 | PC-4 |
|--------------------|--------|---------|---------|-------------------|
| IP-address | xx.246 | yy.192 | yy.193 | xx.427 |
| Role of the SIP SW | UA | Proxy | Proxy | UA |
| SIP SW details | X-Lite | Some-2 | Some-2 | SJPhone for Linux |
| Instances name | UA-A | Proxy-A | Proxy-B | UA-B |
| Wireshark | Yes | No | No | Yes |

Special info: UA-B was configured not to use auto-reply.

2.2. Preconditions

A is registered successfully with Proxy-A. B is registered successfully with Proxy-B.

2.3. Test execution

A chooses the entry “Bob” and presses the green button. (This will cause the UA-A to initiate a call towards Bob@....).

After the phone rings at UA-B, a human (Bob) presses the green button to accept the call.

(what happen after this is up to the candidate to describe as he/she follows the tasks described in exercise 2)

(The test results are given on the next page)

2.4. Partial test results first run (exercise 2a)

The data captured in Wireshark on PC-1 and PC-4 looked as follows:

Table 2 Data from Wireshark (data is captured at PC-1). Filter: SIP and SDP

NB: We are not showing what happened after the message at time T4

| Time | Source | Destin. | Protocol | Info |
|------|--------|---------|----------|---|
| T1 | xx.246 | yy.192 | SIP/SDP | Request: INVITE sip:Bob@ ... with session descr. |
| T2 | yy.192 | xx.246 | SIP | Status: 100 Giving a try |
| T3 | yy.192 | xx.246 | SIP | Status: 180 Ringing |
| T4 | yy.192 | xx.246 | SIP/SDP | Status: 200 OK with session description |
| | | | | |

Table 3 Data from Wireshark (data is captured at PC-4). Filter: SIP, SDP and RTP

NB: We are not showing what happened after the message at time t4

| Time | Source | Destination | Proto- col | Info |
|------|--------|-------------|---------------|--|
| t1 | yy.193 | xx.427 | SIP/SDP | Request: INVITE sip:Bob@ ... with session description |
| t2 | xx.427 | yy.193 | SIP | Status: 100 |
| t 3 | xx.427 | yy.193 | SIP | Status: 180 Ringing |
| t 4 | xx.427 | yy.193 | SIP/SDP | Status: 200 OK with session description |
| | | | | |

2.5. Partial test result 2nd test run (exercise 2f)

Table 4 Some data from Wireshark (data is captured on PC-1). Filter: SIP and SDP

| Time | Source | Destination | Protocol | Info |
|------|--------|-------------|----------|--|
| T1 | xx.246 | yy.192 | SIP/SDP | Request: INVITE sip:Bob@ ... with session description |
| T2 | yy.192 | xx.246 | SIP | Status: 100 Giving a try |
| T3 | yy.192 | xx.246 | SIP | Status: 200 OK with session description |
| T4 | yy.192 | xx.246 | SIP/SDP | Status: 180 Ringing |
| | | | | |

The data captured to and from UA-B is as in the previous case (see Table 3)

3. PSTN related information

3.1. Number portability in PSTN

The material here has been lectured, but is included for your convenience.

The procedure AcQ (All call Query) is defined by npt (PT)³ as the standard procedure in PSTN for service-provider portability. It is illustrated in Figure 6 as follows:

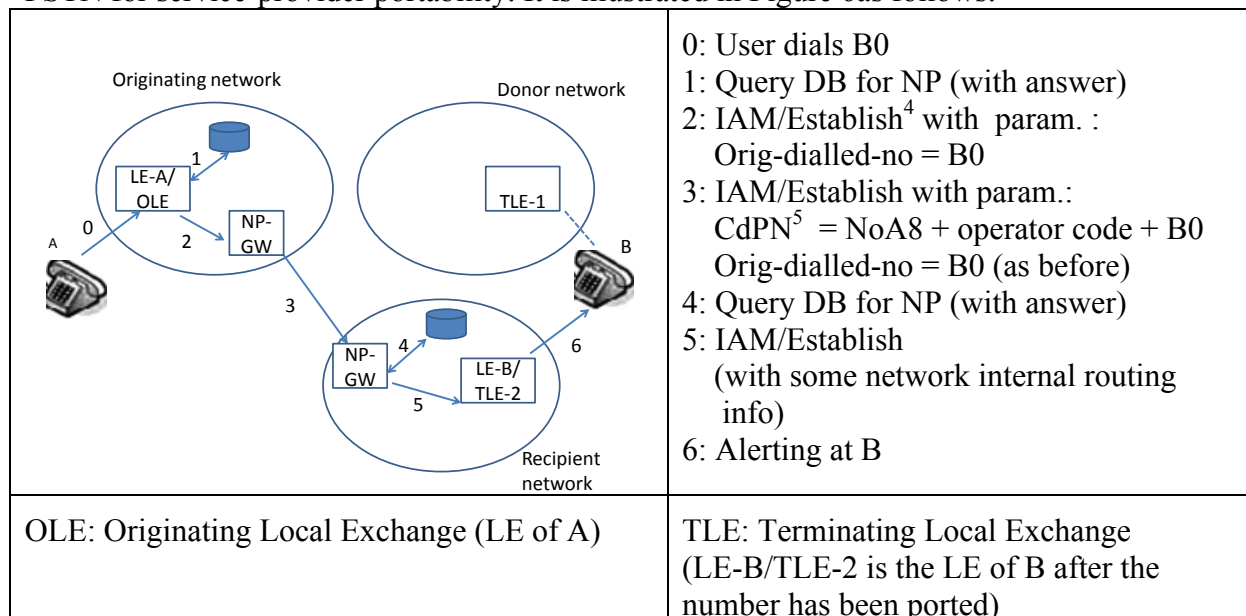


Figure 6 AcQ procedures as defined by npt (PT) for service provider portability.

Notes: Note the use of the prefix NoA with value 8 as part of the E.164 number in CdPN. This value has the meaning ported number. It is followed by operator code for the recipient network. Also note that CdPN is the routable number.

3.2. Some more information on CdPN in PSTN and IN

We will look into one parameter which is often termed CdPN (Called Party Number) and which hold the current number which one shall use to route the call. In case of an IN look-up to rewrite a phone number the Establish messages before and after the SSP will look as follows:

Establish (orig-dialled-no, CdPN CLId, P-bit,...) where CdPN = orig-dialled-no (B0) is used as the routing number in B-no-analysis (B0 may be 800 xxxxx)

Establish(orig-dialled-no, CdPN CLId, P-bit,...) where CdPN has got a new value (this value may be an ordinary fixed number such as 73 73 73 73)

Changing CdPN may also be used when dialing 112/113. CdPN is also used for number portability (as explained above in Figure 6).

³ npt is the acronym for Norwegian Post and Telecommunication Authority (In Norwegian the term is Post- og teletilsynet (PT)). www.npt.no

⁴ In the npt material the Establish message is termed IAM (Initial Address Method). These two termed and the term “call setup” may be used interchangeably in this course, and in your answers as well.

⁵ CdPN is an acronym for “Called Party Number”, i.e. this is the new “B-number” to analyse when routing the call.

4. GSM related information

Note that the details of BSC and BS are not shown on these figures in ttm4130.

4.1. Call setup of basic call

Audestad's book is offering the following illustration of a call setup in GSM.

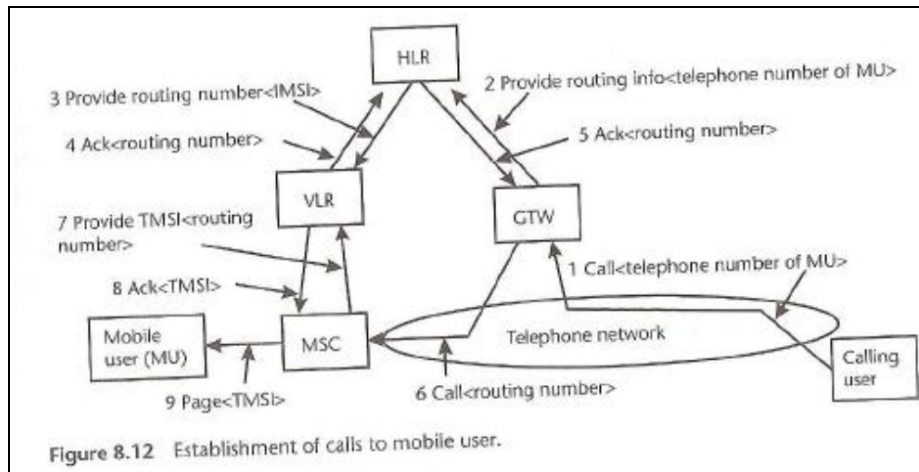


Figure 7 Call setup from PSTN to GSM shown as an informal collaboration diagram (GTW is the GW-MSC).

The most important parameters related to addressing are indicated. The name of the message between PSTN and gateway (GW-MSC/GTW) will be Establish. The message Establish is also called 'call setup'. In the material from npt the name IAM (Initial Address Message) is used instead of Establish. Routing number is called CdPN. You may choose any of these names.

4.2. Some VLR-HLR interaction in GSM

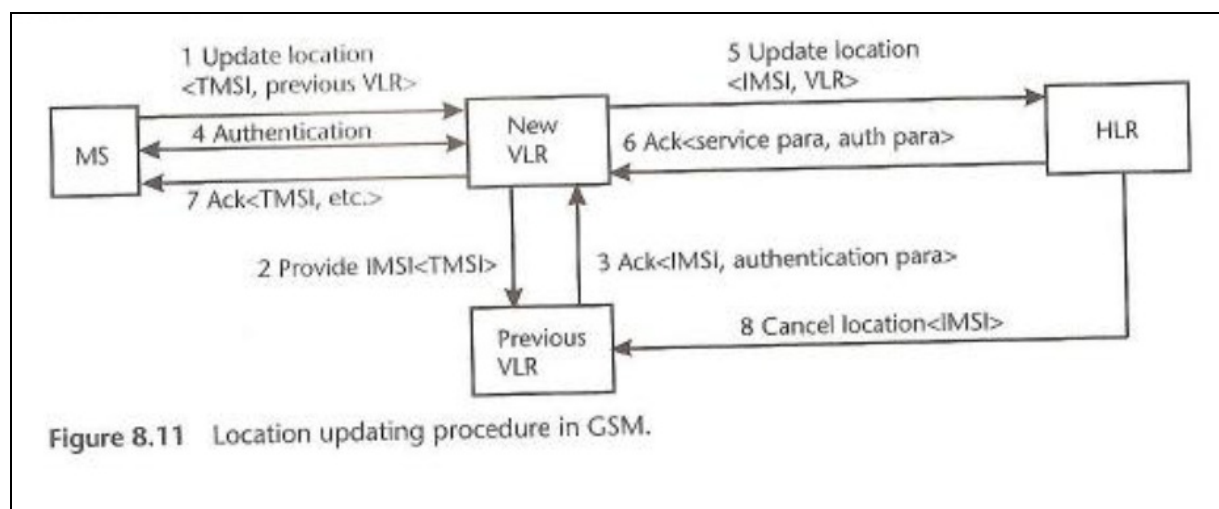


Figure 8. Figure from Audestad's book. Note the use of "service para" in this diagram

Please note that Figure 8 shows what happen when a MS moves from one VLR to a new VLR.

In the exam you are asked about *something a bit different*: Your task is to explain what will happen when the MS is registered with a new VLR when being switched on after being switched off (i.e. during the registration procedure). Figure 8 is considered to be of some aid, even if the case is a bit different.

Student number

page of

Study program

(fill in your own data here)

Use this sheet to answer exercise 2a) and 2b)

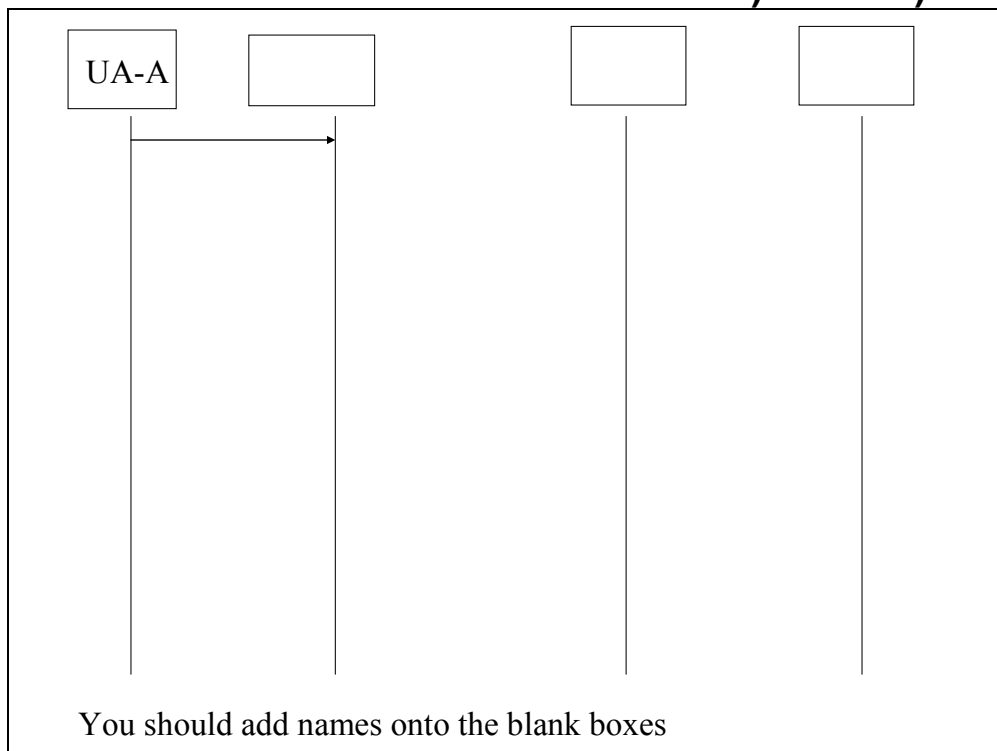
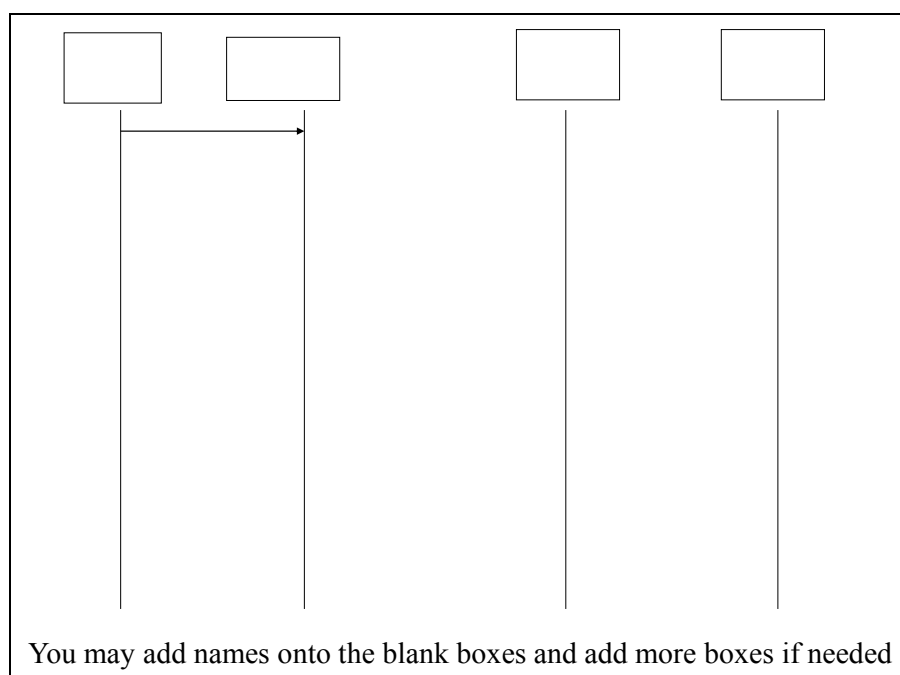


Figure 9 Answer to exercise 2a) 2b) and optional other parts of exercise 2 as well.

Extras which you may use to answer any task of your choice,

Fill in: Answer to exercise part



Student number

page of

Study program

(fill in your own data here)

Answering sheet for exercise 3a)

Add domain borders to these figures corresponding the following 3 cases:

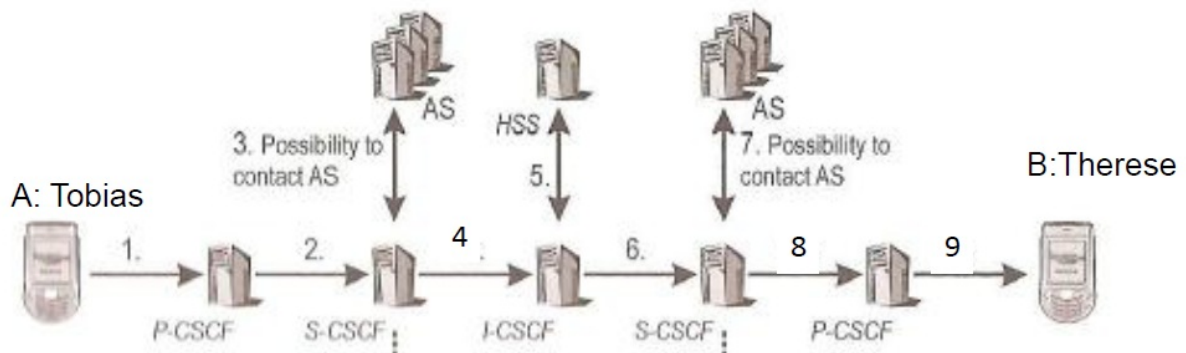


Figure 11 Case a1) Tobias is non-roaming, Theresa is roaming to Austria.

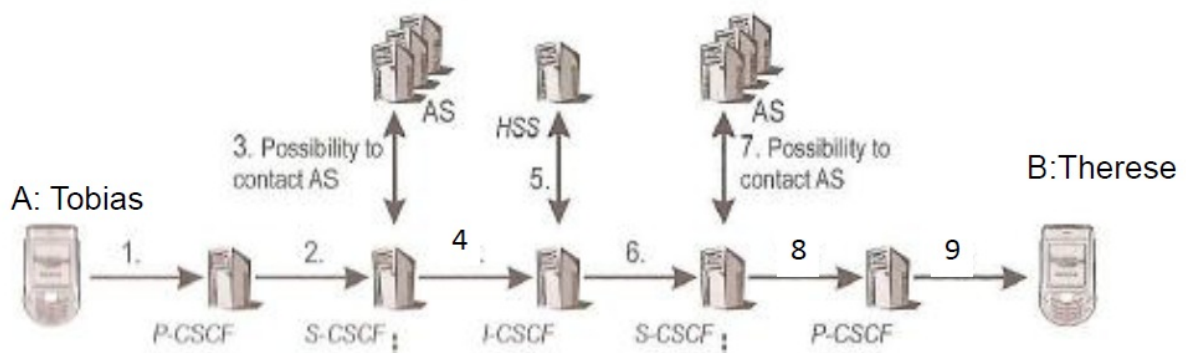


Figure 12 Case a2) Tobias is roaming to Finland, Theresa is non-roaming.

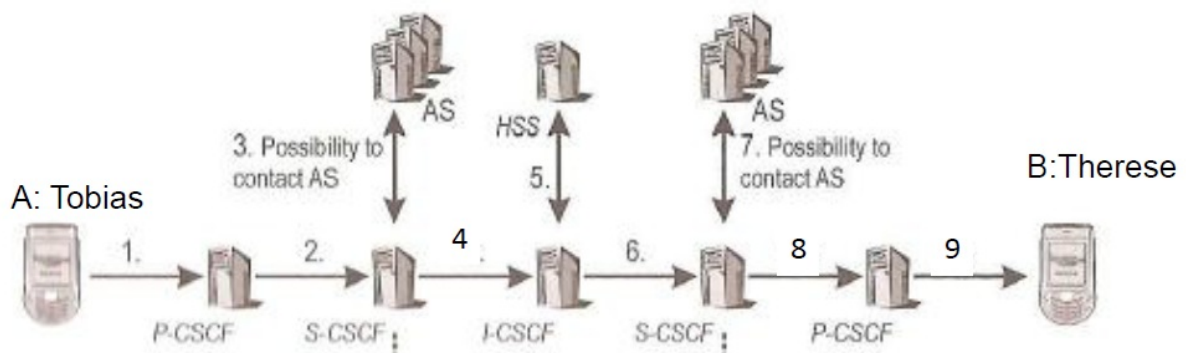


Figure 13 Case a3) Tobias is roaming to Finland, Theresa is roaming to Austria