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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Eksamensdato / Date:	5 juni 2012 / June 5th 2012
Eksamenstid /Time:	09:00-13:00 / 9:00 am – 1:00 pm
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** English is the master text (authoritative) (Norwegian text is for information only) Answer can be in 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):	2 pages (pages 12-13)
Sheets for answers:	4 sheets to be handed in (pages 14-17)
Sensurdato:	**26. juni 2011**

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

Write down 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 %) About SIP and IMS

a) List the 3 types of CSCF that are defined in IMS. List also a database entity.

b) Fill in the sheet on page 16 (follow the instructions).

Exercise 2. (13%) On GPRS

The following is an illustration of GPRS as of 1999.

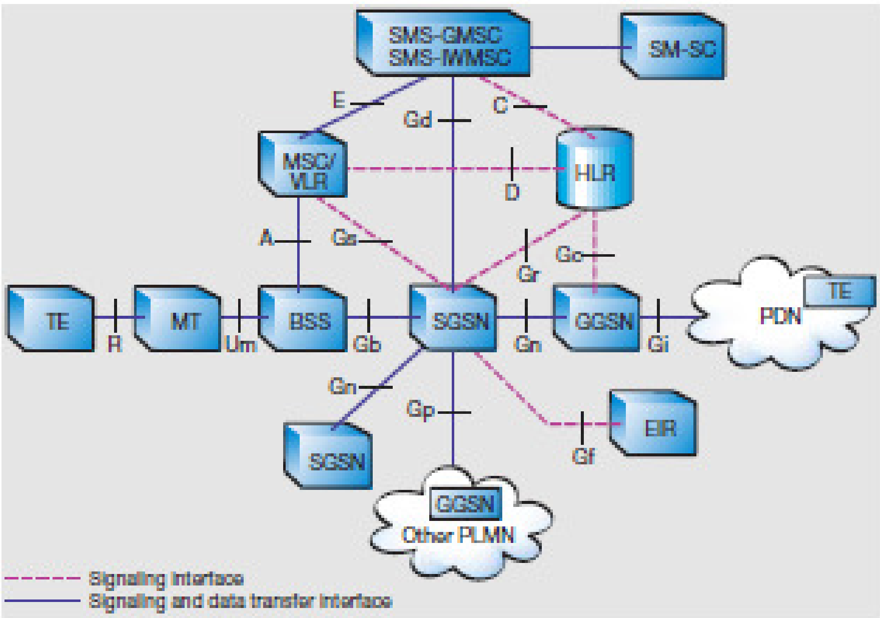


Figure 1 GPRS reference figure from 1999.

a)

a1) Explain what the acronym PDN means.
a2) Give an example of a PDN.
a3) X.25 is one technology that can be used in a PDN. List at least one other technology that is typically used in PDN.

b) As seen in Figure 1 SGSN is connected to BSS (Base Station Subsystem).
b1) Identify one other entity in that belongs to the same level in the network hierarchy as the SGSN.
b2) Explain briefly the role of SGSN.

c) Make an illustration or explain by text where in Figure 1the SS7-based signalling is used, and where IP-based signalling/messages is used.

d) **List** at least two parameters needed in order to establish a PDP context. Explain the role of APN (Access Point Name) during this process.

Exercise 3 (54%) About SIP and SDP as in IETF (almost as 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 cases are described in appendix page 12, and the results as seen in Wireshark are shown on page 13. This setup and software follows the IETF standard, and is not directly related to IMS and IMS entities.

This part of the exam counts for 54 points (54% of the total points), and the points on each letter a), b) etc. varies between 3 and 6 points

a) Part a) is about proxies running in so-called proxy-mode (as statefull proxies utilizing Record-Route, as you did in most of the lab). Look at the results from test run 1 given in Table 4 and Table 5 in Appendix page 13 and combine it with your knowledge of SIP as defined in RFC3261.
Use the drawing sheet on page 14 for your answer and follow the instructions given.

b) Draw a collaboration diagram (as a so-called SIP trapezoid) showing the complete the call setup. Illustrate the media flow(/s) and where the signalling is going. Number the signalling messages.

c) Illustrate what happens when after 2 min. of conversation the calling party decides to ”hang up” (i.e. Alice is pressing the red button on her user interface). Show SIP signalling, do not show media packets or media flows. Use the same MSC as in a) or a separate figure.

d) List all messages from your answer to a) which are containing the *SDP offer*, and list separately all messages which are containing the *SDP answer*. You may answer by marking with “o” and “a” on one or several messages on the previous answer to a) (page 14) or write the answer in plain text.

e) Table 1 shows some important parts of an SDP offer and the corresponding answer side by side:

Table 1 SDP offer and SDP answer side by side

Offer	**Answer**
o=alice 2890844526 2890844526 ++	o=bob 2890844730 2890844730 ++
m=audio 49170 RTP/AVP 0	m=audio 49100 RTP/AVP 0
a=rtpmap:0 PCMU/8000	a=rtpmap:0 PCMU/8000
m=video 51372 RTP/AVP 31	m=video 0 RTP/AVP 31
a=rtpmap:31 H261/90000	(an empty a-line)

Assume these values are form test run 1 and combine this with your answer from a) (page 14) in order to answer the following questions:
e1) Towards which IP address and port number will UA-A send audio?
e2) Will UA-A receive video from UA-B? In case of “yes”: On which IP address and port number will UA-A receive video? In case of “no”: Why not?

f) Explain the role of Via.

g) Explain the role of Record-Route in routing of signalling messages.

h) It is a fact that if the INVITE message is lost before reaching UA-B it will be resent by UA-A.

h1) Explain how UA-A can determine the need to resend INVITE.

h2) Explain the mechanism in the protocol to differentiate between a resending of a previous INVITE from a new INVITE (which may be sent towards the same callee). You may give the name of a particular header as part of your answer.

i) Now we are running the same test again. In this 2nd test run we have configured both proxies in so-called "direct mode" (as stateless proxies not utilizing Record-Route).

Answer by using the sheet on page 15 and follow the instructions.

j) Assume that packet loss causes the ACK message not to reach UA-B.

j1) What will UA-B do in this case?

j2) What will UA-A do in this case?

k) In a 3rd test run both proxies are in "proxy mode" (as statefull proxies) and the (partial) wireshark observations are given in Table 6 and Table 5 on page 13.

k1) Explain why you cannot be sure that Pr-B is *receiving* 200 after 180, even though the Wireshark results show that 200 is *sent* after 180 from UA-B.

k2) Explain what will happen signalling-wise if packet loss causes the ACK message to be sent from Pr-A but not reaching Pr-B.

l) Assume that UA-B has implemented the following policy: It will not *send* media before it has received the ACK message. We may use the term this "late media sending" for this.

Explain what is the earliest possible time (relative to the timeline for UA-B) that UA-B must be prepared to *receive* media packets according to RFC3264 (the offer-answer model). You may mark this time with a * on the MSC (page 14) or explain otherwise.

m) Assume that UA-A has implemented the following policy: It will send RTP media packets as soon as possible (before it is sending the ACK message). We may call this "early media sending" for this.

Give at least two reasons why UA-B must be prepared to receive P1 (first rtp packet) and P2 (ACK) in either order.

Exercise 4. (23%) Application server in IMS and IN-node in GSM+Camel

This part of the exam counts for 23 points (23% of the total points), and the points on each letter a), b), c) etc. varies between 3 and 6 points.

We start this task by illustrating the use of application servers (AS) in IMS, and the somewhat similar concept of an IN-node in Camel.

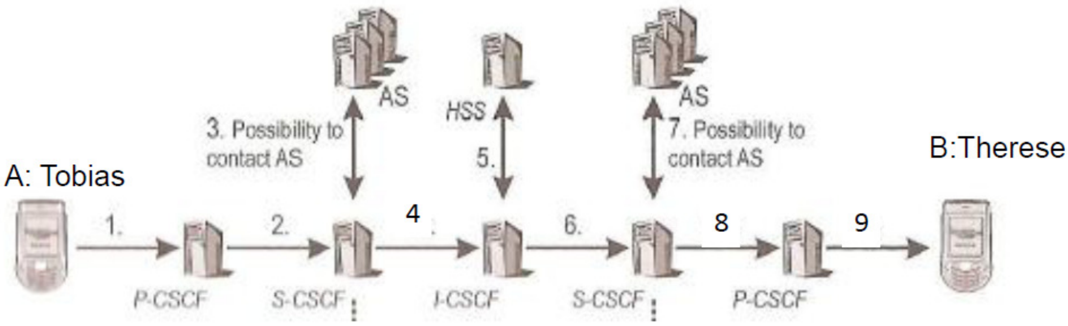


Figure 2 IMS Call setup as a simplified collaboration diagram (from IMS book, slightly edited). In this case oth originating side and terminating side are involving AS for special services. The HSS is in this case related to the terminating side.

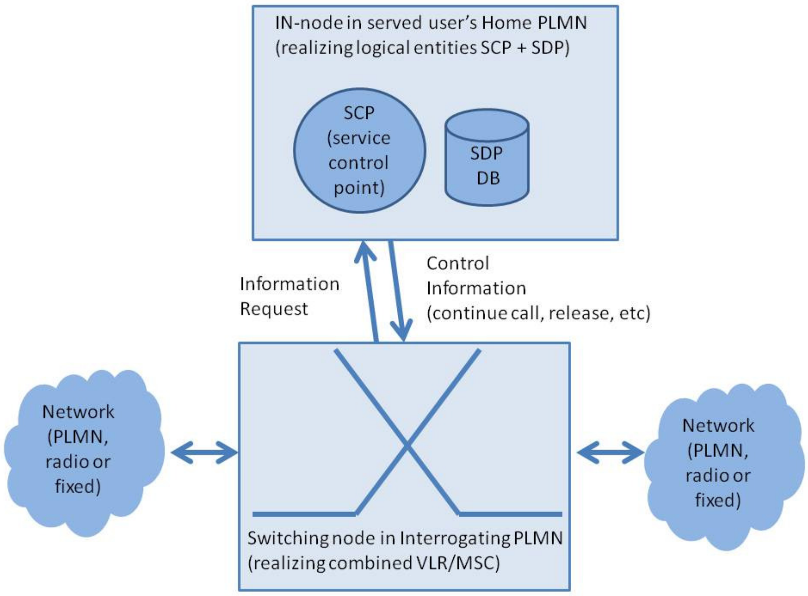


Figure 3 A simplified view of Camel during a call setup attempt (signalling). The MSC in placed in a so-called Interrogating PLMN (IPLMN). The MSC may or may not be a serving MSC of the particular user (depending on the case).

a) Fill in the sheet on page 17.

b) Identify one interface in Figure 2 which is not using SIP, and explain what protocol this interface is using. You may refer to this interface by referring to the corresponding number in Figure 2.

c) Explain the meaning of CSI within the context of Camel.

d) An IN-node in Camel acts as a kind of Application Server (AS) for GSM subscribers. This may support *originating* services like company-wide-short-number-service. In IMS the triggering of the AS on originating side happens in the home domain (see Figure 2). But the role of the home and the visited networks are *different* in GSM+Camel.

Assume user A is roaming and is originating a call which need interaction with the IN-node via Camel (e.g. because A subscribes to company-wide-short-number-service). In Camel this is called the MO-case (MO=Mobile Originated).

Draw a sequence diagram / collaboration diagram to illustrate the message sequence when this call is initiated. You are asked to draw a diagram at approximately the same level as in Figure 2. **Explain** how the *visited* network plays a role. **Hint:** It is a good idea to include a description of the role of the HLR on the originating side.

e) An IN-node in Camel acts as a kind of Application Server (AS) for GSM subscribers. Camel allows for IN-nodes in the home network to support *terminating* services like personal-assistant (a kind of Do-not-disturb service), which is directing some calls to Voicemail, and some calls through to the user, based on some non-standardized rules and personal data defined in the IN-node. In IMS the triggering of the AS on the terminating side happens in the *home* domain (see Figure 2). But the roles of the various network domains are *different* in GSM+Camel.

Assume user B is roaming and is receiving a terminating call which need interaction with the IN-node via Camel (e.g. because B has activated his personal-assistant service). In Camel this is called the MT-case (MT=Mobile Terminated).

Draw a sequence diagram /collaboration diagram to illustrate the message sequence when this call is routed towards user B. You are asked to draw a diagram at approximately the same level of detail as in Figure 2. **Explain** how the *interrogating* network (IPLMN) plays a role. **Hint:** It is a good idea to include a description of the role of the HLR on the terminating side.

There are two options for this to happen in Camel for the MT-case, chose *one* of the options in your answer.

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

Formuler egne antakelser i besvarelsen 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 SIP og IMS

a) List de 3 typene av CSCF som er definert i IMS. List også en databaseenhet.

b) Fyll inn arket på side 16 (følg instruksjonene).

Oppgave 2. (13%) Om GPRS

Det følgende er en illustrasjon av GPRS slik det så ut i 1999.

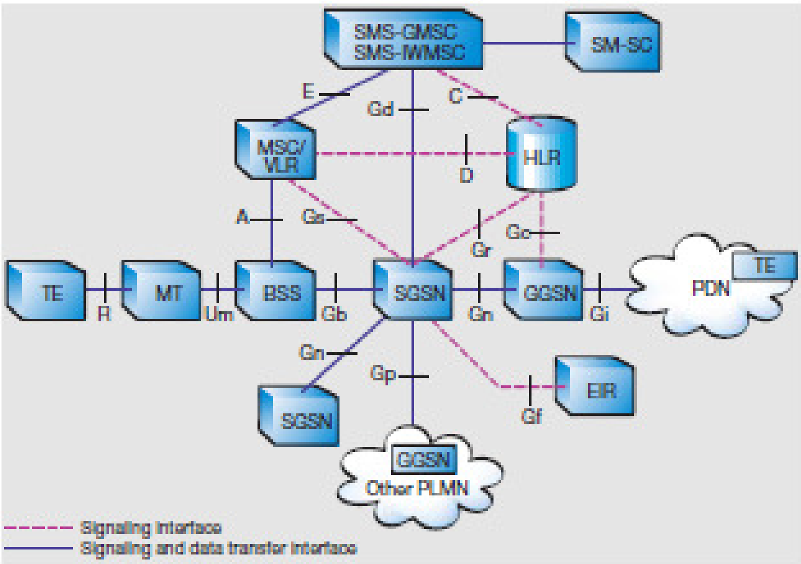


Figure 4 (/Figure 1) GPRS reference figure from 1999.

a)

a1) Forklar hva akronymet PDN betyr.
a2) Gi et eksempel på et PDN.
a3) X.25 er en teknologi som kan bli brukt i et PDN. List minst en annen teknologi som typisk er brukt i slike PDN.

b) Som vist i Figure 4 (fig. 1) er SGSN koblet mot BSS (Base Station Subsystem).
b1) Identifiser en annen enhet one other entity in Figure 4 (fig. 1) som er på samme nivå i nettverkshierarkiet som SGSN.
b2) Forklar kort rollen til SGSN.

c) Lag en illustrasjon eller forklar med tekst hvor Figure 4 (fig. 1) SS7-basert signalering er i bruk, og hvor IP-based signallering/meldingsutveksling er i bruk.

d) **List** minst to nødvendige parametre når man skal etablere en PDP kontekst. Forklar rollen til APN (Access Point Name) under denne prosessen.

Oppgave3 (54%) Om SIP og SDP fra IETF (veldig likt det vi hadde i lab)

Din oppgave i denne delen er å jobbe med data som kunne vært reelle testdata fanget med Wireshark i labben. Oppsettet (setup) , forbetingelsene (preconditions) og test-tilfellene (test cases) er beskrevet i appendikset side 12, og resultatene fra Wireshark er vist på side 13. Oppsettet og programvaren følger IETF standarden, og er ikke direkte relatert til IMS og enheter i IMS.

Denne delen av eksamen teller 54 poeng (54 % av totalen), og poengene på hver bokstav a), b) etc. varierer mellom 3 and 6 poeng.

a) Del a) handler om proxier som kjører i såkalt proxy-modus. (dvs. som tilstandsfulle proxier som benytter Record-Route, slik du gjorde i mesteparten av labben). Se på resultene fra test kjøring 1 som gitt i Table 4 and Table 5 i Appendix side 13 og kombinér det med din egen kunnskap om SIP som definert i RFC3261.
Bruk arket på side 14 når du besvarer a) og følg instruksjonene.

b) Tegn et kollaborasjonsdiagram (som et såkalt SIP trapesoid) som viser det komplette samtaleoppsettet. Illustrer mediaflyten(/e) og hvor signalleringen går. Nummerer signalleringsmeldingene.

c) Illustrér hva som skjer når anropende part bestemmer seg for å "legge på" etter 2 minutter (dvs. Alice trykker på den røde knappen på sitt brukergrensesnitt). Vis SIP signalling, men ikke vis media pakker eller mediaflyter. Bruk gjerne samme MSC som i a) (side 14) eller bruk en separat figur.

d) List alle meldinger fra testkjøring 1 som vise *SDP tilbud (offer)*, og list separat alle meldinger som inneholder *SDP svar (answer)*. Du kan svare med å markere med "o" og "a" på en eller flere meldinger i besvarelsen til a) (side 14) eller du kan skrive svaret i tekstlig form.

e) Table 2 viser de viktigste delene av et tilbud (offer) og et svar (answer) side om side:

Table 2 (samme som tabell 1) SDP tilbud (offer) og SDP svar (answer) side om side.

Tilbud (offer)	**Svar (answer)**
o=alice 2890844526 2890844526 ++	o=bob 2890844730 2890844730 ++
m=audio 49170 RTP/AVP 0	m=audio 49100 RTP/AVP 0
a=rtpmap:0 PCMU/8000	a=rtpmap:0 PCMU/8000
m=video 51372 RTP/AVP 31	m=video 0 RTP/AVP 31
a=rtpmap:31 H261/90000	(an empty a-line)

Anta disse verdiene kom fra testkjøring 1 og kombiner med svaret fra a) (side 14) for å besvare følgende spørsmål:
e1) Til hvilken IP adresse og port nummer vil UA-A sende audio?
e2) Vil UA-A sende video til UA-B? I tilfelle "ja": På hvilken IP addresse og port nummer vil UA-A motta audio? I tilfelle "nei": Hvorfor ikke?

f) Forklar rollen til Via.

g) Forklar rollen til Record-Route ved ruting av signalleringsmeldinger.

h) Det er et faktum at hvis INVITE-meldingen er tapt før den når fram til UA-B, så vil den bli sendt på nytt av UA-A.

h1) Forklar hvordan UA-A kan avgjøre behovet for å re-sende INVITE.

h2) Forklar mekanismen i protokollen for å skille mellom å re-sende en tidligere INVITE og å sende en ny INVITE (som forøvrig jo kan være til samme mottaker). Du kan godt angi navnet på et spesifikt hode (header) som en del av ditt svar.

i) Nå kjører vi same test igjen. I denne andre testkjøringen er begge proxiene konfigurert i såkalt "direkte modus" (direct mode) (dvs. som tilstandsløse proxier som ikke benytter Record-Route).

Svar ved å benytte arket på side 15 og følg instruksjonene.

j) Anta at pakketap forårsaker at ACK-meldingen ikke når fram til UA-B.

j1) Hva vil UA-B gjøre i dette tilfellet?

j2) Hva vil UA-A gjøre i dette tilfellet?

k) I en tredje testkjøring er begge proxiene i "proxy modus" (som tilstandsfulle proxier som benytter Record-Route) og (deler av) Wireshark-observasjonene er som gitt i Table 6 og Table 5 på side 13.

k1) Forklar hvorfor du ikke kan være sikker på at Pr-B *mottar* 200 etter 180, selv om Wireshark-resultatet viser at 200 er *sendt* etter 180 fra UA-B.

k2) Forklar hva som vil skje signallingsmessig hvis pakketap forårsaker ACK meldingen til å bli sendt fra Pr-A uten å nå Pr-B.

l) Anta at UA-B har implementert følgende policy: Den vil ikke *sende* media før den har mottatt ACK-meldingen. Vi kan kalle dette "sein mediasending".

Forklar hva som er tidligst mulige tid (relativt til tidslinjen for UA-B) at UA-B må være forberedt på å *motta* mediapakker i henhold til RFC3264 (the offer-answer model). Du kan markere dette med en * på MSC'et side 14 eller forklare på annen måte.

m) Anta at UA-A har implementert følgende policy: Den vil sende RTP-pakker så raskt som mulig (dvs. før den sender ACK-meldingen). Vi kan kalle dette "tidlig mediasending".

Angi minst to grunner til at UA-B må være forberedt på å motta pakkene P1 og P2 i vilkårlig rekkefølge (der P1 er første rtp-pakke og P2 er ACK)

Oppgave 4. (23%) Applikasjonstjener (AS) i IMS og IN-node in GSM+Camel

Denne delen av eksamen teller 23 poeng (23 % av de totale poeng), og poengene på hver bokstav a), b), c) etc. varierer mellom 3 og 6 poeng.

Vi starter med å illustrere bruken av applikasjonstjener (AS) i IMS, og med det delvis tilsvarende konseptet med en IN-node i Camel.

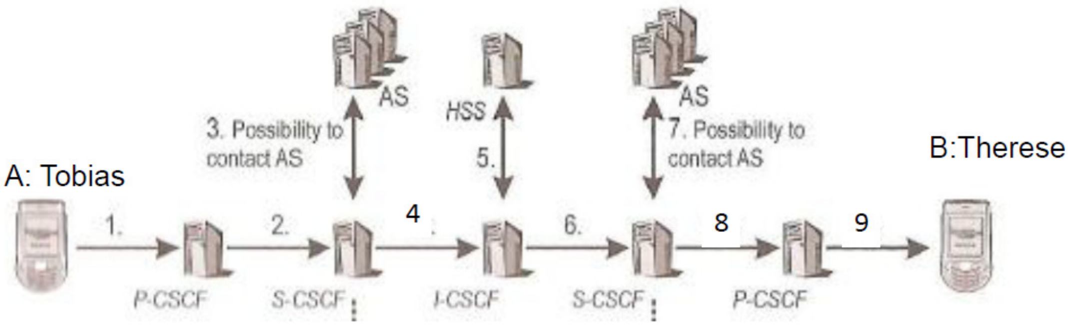


Figure 5 (/ Figure 2) IMS signallering vist som et forenklet kollaborasjonsdiagram (fra IMS-boka, lettere endret). I dette tilfellet er det vist at både originerende og terminerende side involverer AS for spesielle tjenester. Den viste HSS er i dette tilfellet på terminerende side.

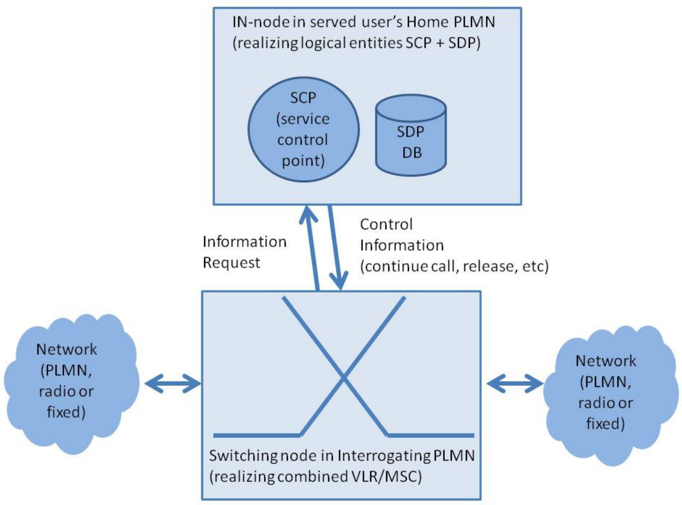


Figure 6 (/ Figure 3) Et forenklet syn på Camel under et samtaleoppsett (signallering). MSC-en er plassert i såkalt Interrogerende PLMN (IPLMN). MSC er kanskje - men ikke nødvendigvis - en betjenende (serving) MSC for den aktuelle bruker (dette avhenger at hvilket tilfelle man ser på).

a) Fyll inn arket på side 17

b) Identifiser ett interface i Figure 5 som *ikke* bruker SIP, og forklar hvilken protokoll dette interfacet bruker. Du kan referere til dette interfacet ved å referere til det korresponderende nummeret i Figure 5.

c) Forklar hva CSI betyr / står for innafor konteksten av Camel.

d) En IN-node i Camel tilsvarer en slags applikasjonstjener (AS) for GSM abbonenter. En slik node kan støtte *originerende* tjenester som felles-kortnummer-for-hele-bedriften. I IMS skjer triggingen av AS på den originating side i hjemmedomenet (se Figure 5). Men rollen til hjemmedomene og besøksdomene (visited network) er anderledes i GSM+Camel.

Anta at bruker A roamer og originerer en samtale som trenger interaksjoner med IN-noden via Camel (f.eks. fordi A abbonerer på tjenesten felles-kortnummer-for-hele-bedriften). I Camel kalles dette caset MO (Mobile Originated).

Tegn et sekvensdiagram / kollaborasjonsdiagram som illustrerer meldingssekvensen når denne samtalen initieres. Du bes om å tegne et diagram på ca. samme detaljeringsnivå som Figure 5**. Forklar** hvordan besøksnettet (visited network) spiller en rolle. **Hint:** Det er en god idé å inkludere en beskrivelse av rollen til HRL på originerende side.

e) En IN-node i Camel tilsvarer en slags applikasjonstjener (AS) for GSM abbonenter. Camel muliggjør at IN-noder i hjemmenettet kan støtte *terminerende tjenester* som f.eks. personlig-assistent (en slags Do-not-disturb tjeneste), som vil sende noen anrop til Voicemail, og noen anrop til brukeren basert på ikke-standardiserte regler og personlige data definert i IN-noden. I IMS skjer triggingen av AS på terminerende side i *hjemmenettet* (se Figure 6). Men rollene til de forskjellige nett-domenene er *annerledes* i GSM+Camel.

Anta at bruker B er roamende og mottar et terminerende anrop som trenger interaksjoner med IN-noden via Camel (f.eks. fordi B har aktivert sin personlige-assistent-tjeneste). I Camel kalles dette caset MT (MT=Mobile Terminated).

Tegn et sekvens diagram / kollaborasjonsdiagram som illustrerer meldingssekvensen når dette anropet rutes i retning av bruker B. Du bes om å tegne et diagram på ca. samme detaljeringsnivå som Figure 5. **Forklar** hvordan *interrogerende nett* (IPLMN) spiller en rolle. **Hint:** Det er en god idé å inkludere en beskrivelse av rollen til HRL på terminerende side.

Det er to opsjoner i Camel for dette caset (MT-case), velg én av opsjonene i ditt svar.

Appendix

1. Information related to exercise 3 (SIP in the lab)

1.1. Set up in the test cases reported

We are using 4 physical machines all running relevant SIP software, with no use of virtual machines. We follow the traditional setup called "SIP trapezoid" having two proxies.

In exercise 3a)- 3c) (test run 1) you shall assume that both proxies are configured to be in so-called "proxy-mode" (or as a statefull proxy, utilizing Record-Route header).

In exercise 3.i) (test run 2) you shall assume that both proxies run in so-called "direct-mode" (or as stateless proxy, not using Record-Route). In both cases the signaling SIP runs over UDP (as in your lab).

We used wireshark only on two of these machines, as detailed in Table 3.

Table 3 Set-up and configuration during the test cases (ignore the start of all IP addresses)

Physical entity	PC-1	PC-2	PC-3	PC-4
IP-address	(..).246	(..).192	(..).193	(..).427
Role of the SIP SW	UA	Proxy	Proxy	UA
SIP SW details	Some-1 (supporting also video)	Some-2	Some-3	Some-4
Instances name	UA-A	Pr-A	Pr-B	UA-B
Wireshark	Yes	No	No	Yes

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

1.2. Preconditions

Alice on UA-A is registered successfully with Pr-A. Bob on UA-B is registered successfully with Pr-B. UA-A has "Bob" registered in the address book.

1.3. Test execution

Alice on UA-A chooses the entry "Bob" in the address book 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 3)

1.4. Test results (see next page)

1.5. Test verdict

Test is passed (this means that all SIP signaling is according to specification RFC3261), that all SDP messages are according to offer-answer model, and that audio worked both ways.

Partial test results first test run (task 3 a) and second test run (task 3 i)

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

Table 4 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 T5.

Time	Source	Destin.	Protocol	Info
T1	(..).246	(..).192	SIP/SDP	Request: INVITE sip:Bob@ ... with session descr.
T2	(..).zzz	(..).246	SIP	Status: 100 Giving a try
T3	(..).zzz	(..).246	SIP	Status: 180 Ringing
T4	(..).zzz	(..).246	SIP/SDP	Status: 200 OK with session description
T5	(..).246	(..).xxx	SIP	Request: ACK

NB: The value of the source and destination is withheld for some of the messages. The candidate shall be able to find out the value of zzz and xxx by themselves.
NB2: Notice that the values of IP addresses marked zzz and xxx may differ between the test runs. The values of some of the headers (not shown) may also vary between the test runs.

Table 5 Data from Wireshark (data is captured at PC-4). Filter: SIP and SDP
NB: We are not showing what happened after the message at time t3

Time	Source	Destination	Protocol	Info
t1	(..).193	(..).427	SIP/SDP	Request: INVITE sip:Bob@ … with session description
t2	(..).427	(..).yyy	SIP	Status: 180 Ringing
t 3	(..).427	(..).yyy	SIP/SDP	Status: 200 OK with session description

NB: The value of the destination is withheld for some of the messages. The candidates shall be able to find out the value of yyy by themselves. (yyy may vary between the test runs)

Partial test result 3rd test run (exercise 3 k)

Table 6 Data from Wireshark (data is captured on PC-1). Filter: SIP and SDP

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

NB1: The value of the source or destination is withheld for some of the messages. The candidates shall be able to find out the value of uuu by themselves.

The test data captured on PC-4 is given in Table 5.

Candidate number .... **page ..... of ......**

Study program .... **(fill in your own data here)**

Use this sheet to answer exercise 3a) (proxy mode)

This task is about proxies running in so-called proxy-mode (as statefull proxies utilizing Record-Route, as you did in most of the lab). Look at the results from test run 1 given in Table 4 and Table 5 in Appendix page 13 and combine it with your knowledge of SIP as defined in RFC3261.

Task a1) Complete instance names and IP-addresses on the boxes on top.

Task a2) Draw one MSC showing the whole set of signalling messages being exchanged within the given timeframe between all 4 involved entities. Use the data from the Wireshark observations, and add messages on the entities where no Wireshark observation is given. Mark clearly which messages are not captured by Wireshark, by adding "no" (for "not observed") on all relevant messages.

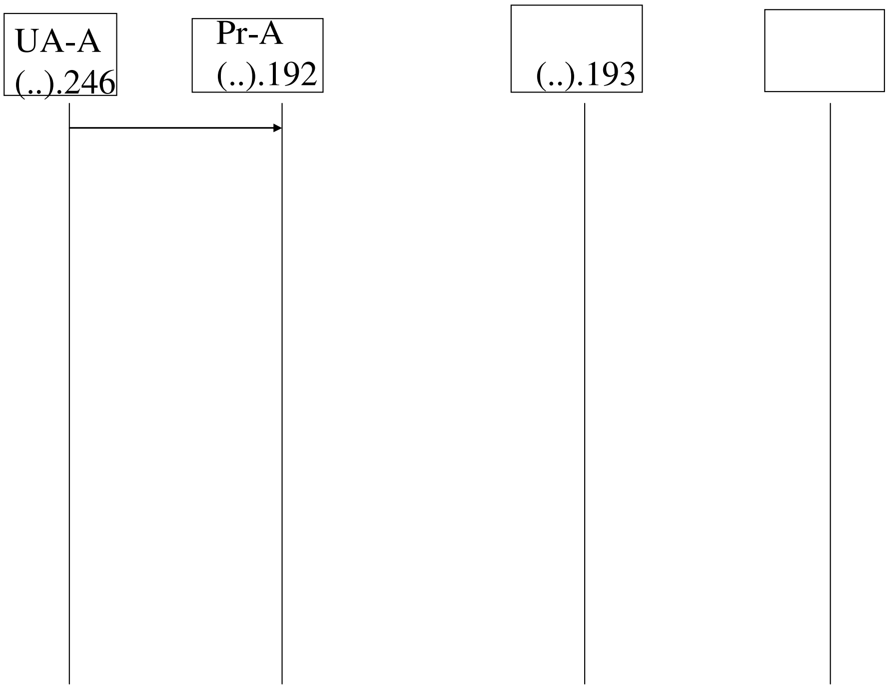


Figure 7 Answer to exercise 3a) here. (Use the same diagram to answer 3c) if you want and have room). Answers to other parts of exercise 3 may also be added here if you find that feasible.

Use the space below for any exercise (such as 3c) (if you want)

Candidate number .... **page ..... of ......**

Study program .... **(fill in your own data here)**

Use this sheet to answer exercise 3i) (direct mode)

Now we are running the same test again. In this 2nd test run we have configured both proxies in so-called "direct mode" (as stateless proxies not utilizing Recourd-Route). The results are given by Table 4 and Table 5 on page 13.

Task a1) Complete instance names and IP-addresses on the boxes on top.

Task a2) Draw one MSC showing the whole set of signalling messages being exchanged within the given timeframe between all 4 involved entities.

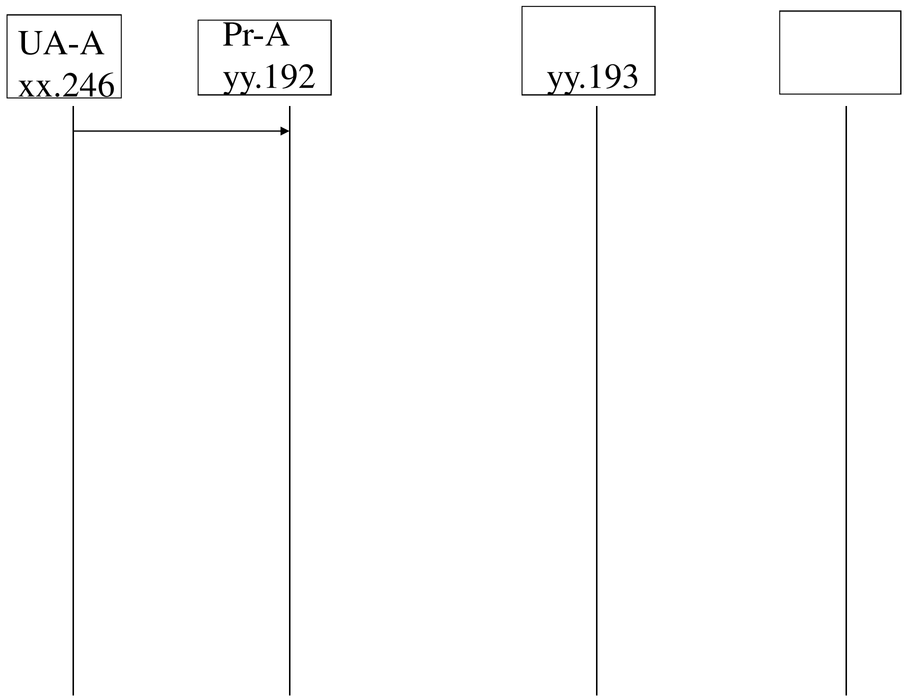


Figure 8 Answer to exercise 3i) here.

Empty space may be used by the candidate.

Candidate number .... **page ..... of ......**

Study program .... **(fill in your own data here)**

Use this sheet when answering exercise 1 b)

Task b1): Fill in correct names instead of …-CSCF, …-CSCF and …-CSCF.
Task b2): Complete the following call flow (message sequence chart) on the terminating side. Do this by filling in all messages between the first and the last arrow shown, i.e. all messages relating to the terminating side from INVITE to ringing. Show SIP messages, but show also briefly message(/s) to/from HSS.

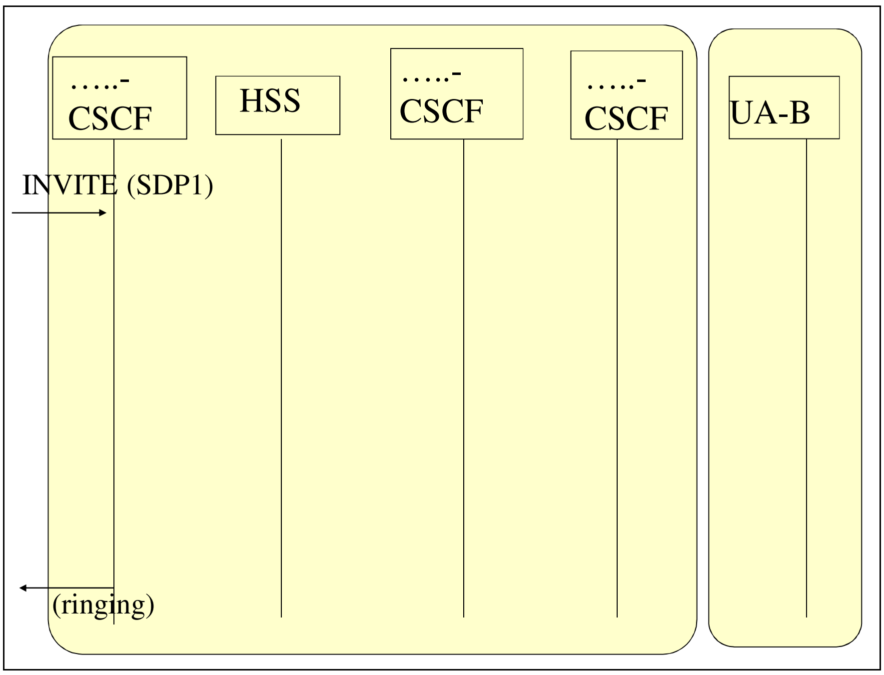
<table>
<tr><th>Originating side</th><th>Terminating side (Bob is non-roaming in this case)</th></tr>
<tr><td>From Alice
Via….
Via Alice's home domain</td><td></td></tr>
<tr><td></td><td>
</td></tr>
<tr><td>To Alice's home domain
For further routing to Alice's UA.</td><td></td></tr>
</table>

Figure 9 Answer to exercise 1 b) goes in the right hand-side of this figure

Candidate number .... **page ..... of ......**

Study program .... **(fill in your own data here)**

Answering sheet for exercise 4a) on IMS

Add domain boarders to these figures corresponding to the following 2 cases.

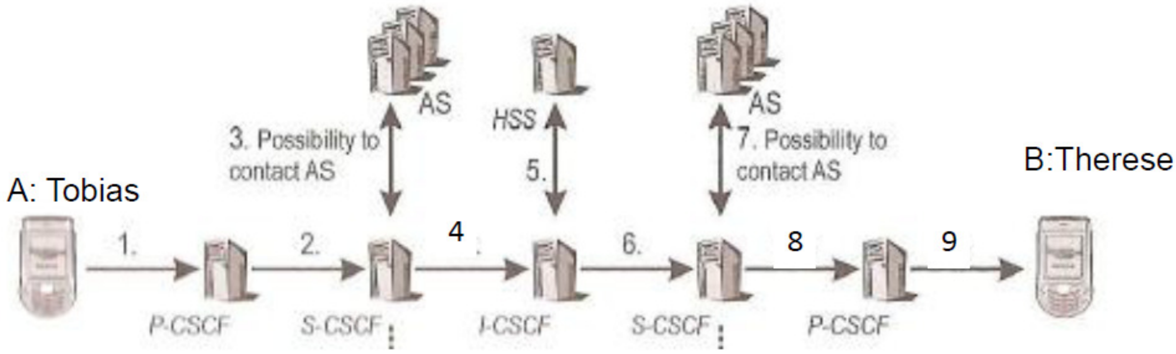


Figure 10 Case: Tobias is non-roaming, Theresa is roaming to Austria.

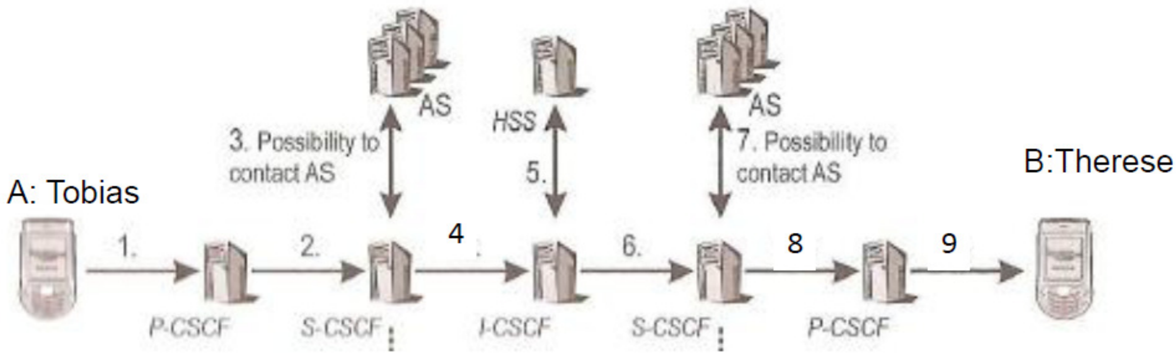


Figure 11 Case: Tobias is roaming to Finland, Theresa is non-roaming.