

**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:	10. august 2009 / August 10th 2009
Eksamentid /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 hjelpeemidler/	D: Ingen trykte eller håndskrevne hjelpeemidler tillatt Bestemt, enkel kalkulator tillatt
Språkform / language:	English English is the master text (authoritative) (Norwegian text is for information only) Answer can be in nynorsk, bokmål or English
Number of pages in English:	8 pages (including 4 pages to be handed in)
Antall sider bokmål:	8 (sidene 10-17, inkludert 4 sider som skal leveres inn)
Antall sider nynorsk:	0
Appendix (in English):	2 (pages 18-19)
Sensurdato¹:	31. august 2009

¹ Merk! Studentene må primært gjøre seg kjent med sensur ved å oppsøke sensuroppslagene. Evt. telefoner om sensur må rettes til sensurtelefonene. Eksamenskontoret vil ikke kunne svare på slike telefoner.

You should **start by reading through all the material** and then decide which sequence you want to use when answering the exercises. Note the 4 pages that shall be handed in.
Short answers are requested.

Make your own assumptions if the text is unclear, or information is missing.

Exercise 1. (28 %) About SIP and IMS

a) (3 %)

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

b) (5%)

Describe at least one task/role performed by each of the 3 types of CSCF **during registration**. In total you may get points for 5 different tasks (but you should list between 1 and 2 tasks per entity)

It is up to the candidate to decide on the sequence to answer b) and c), as they are of course interdependent.

c) (9 %)

Draw a call flow (or message sequence chart) of the registration procedure from a REGISTRATION is sent from the UA until a 200 OK is received back to the UA. This procedure consists of **two phases**, where the first phase ends with a 401 Unauthorized challenge back to the UA. Hint: There are some similarities between the MSCs for the first and second phase. **Use the separate sheet on page 3 when answering this.**

d) (5 %) Describe at least one task/role performed by each of the 3 types of call session control functions (...-CSCF) **during a session establishment** (i.e. during call setup). In total you may get points for 5 different tasks (but you should list between 1 and 2 tasks per entity).

It is up to the candidate to decide on the sequence to answer d) and e), as they are of course interdependent.

e) (6 %)

Assume both Alice and Bob are properly registered. Then Alice is calling Bob. I.e. Alice is sending an INVITE from her UA via her home network. At some point this INVITE reaches Bob's home network.

Task 1: Fill in correct names of the CSCFs involved.

Task 2: Fill in the proper name (response value in SIP) of the last message (ringing)

Task 3: Complete the following call flow (message sequence chart) by filling in all messages between the first and the last message shown, i.e. those messages that relates to the terminating side.

Use the separate sheet on 4 page when answering this.

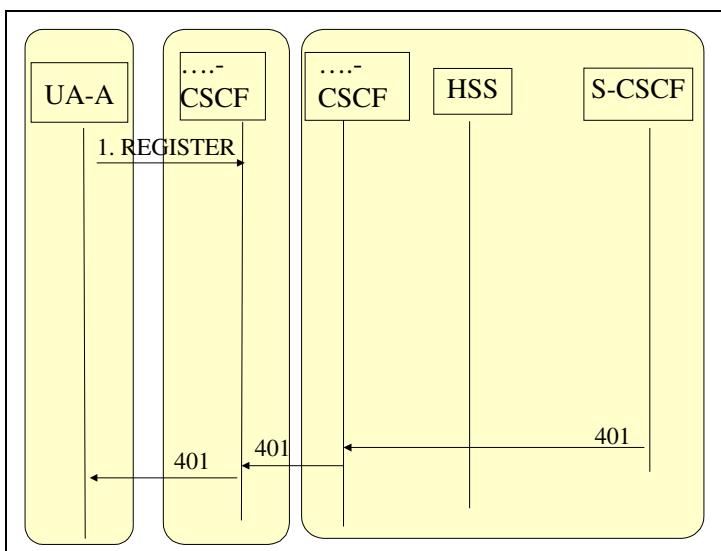
Candidate number

Page ... out of
(fill in all appropriate numbers)**Use this sheet when answering Exercise 1 c) (9 %)**

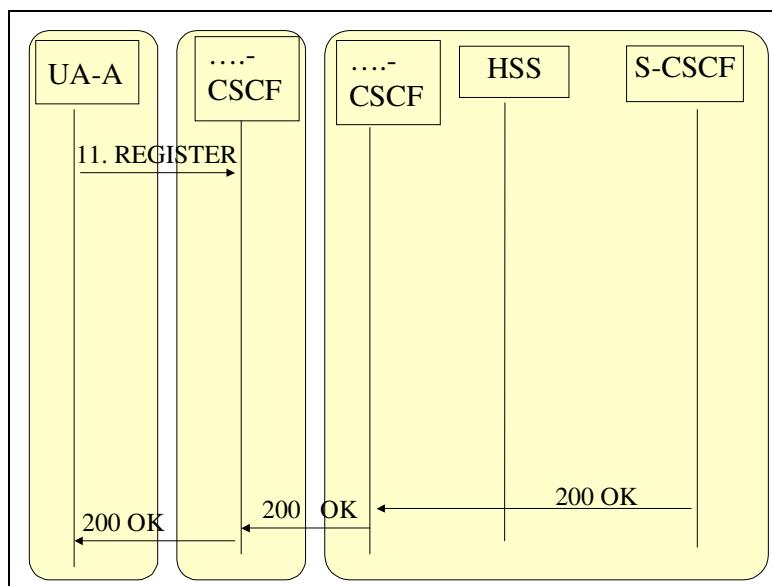
The 3 shaded areas represents the enduser domain, the access (/visited) domain and the home domain (of Alice in this case).

Task 1: Fill in correct names of the CSCFs involved.

Task 2: Complete the following two call flows (message sequence charts) by filling in all messages between the first and the last message shown. The first part shall consist of approximately 8-10 messages in total.



Figur 1 Phase 1: the initial part of a registration, S-CSCF asks for a challenge in order to authenticate UA-A. This is done via 401.



Figur 2 The second phase of the registration. For simplicity you may assume that the first message here is number 11.

Candidate number

Page out of
(fill in all appropriate numbers)

The 3 shaded areas represents the home domain (of Bob), the access (/visited) domain and the enduser domain.

Use this sheet when answering exercise 1.e) (6 %)

Task 1: Fill in correct names instead of ...-CSCF, ...-CSCF and ...-CSCF.

Task 2: Fill in the proper name / response value in SIP of the last message (ringing)

Task 3: Complete the following call flow (message sequence chart) by filling in all messages between the first and the last message shown, i.e. all messages relating to the terminating side.

<i>Originating side</i>	<i>Terminating side</i>
From Alice Via.... Via Alice's home domain	
	<pre> graph TD CSCF1[.....-CSCF] --> CSCF2[.....-CSCF] CSCF2 -- "INVITE (SDP1)" --> CSCF1 CSCF2 -- "ringing" --> CSCF1 CSCF3[.....-CSCF] UA_B[UA-B] </pre>
To Alice's home domain For further routing to Alice's UA.	

Exercise 2 (16 %) SIP protocol details

- a) (5 %) Explain the role of the following header fields in SIP
- o Via
 - o From
 - o Contact
- In particular explain the *difference* between From and Contact
- b) (3%)
- Explain the role of the header Content-type, and how this is used with SDP to do media negotiation.
- c) (8%)
- For each MSC give the answer ‘legal’ or ‘not legal’. For each illegal MSC explain what is wrong. Comments are welcome also on legal MSCs (i.e. if it deviates from a normal case)
- (See separate sheet to be handed in)

Candidate number

Answering sheet to to 2 d (8%)Page out of
(fill in all appropriate numbers)

<p>Message sequence chart</p> <pre> graph LR UA_A[UA-A] -- INVITE --> P1[Proxy] P1 -- INVITE --> P2[Proxy] P2 -- INVITE --> UA_B[UA-B] UA_B -- "200 OK" --> P2 P2 -- "200 OK" --> P1 P1 -- "200 OK" --> UA_A UA_A -- ACK --> P1 P1 -- ACK --> P2 P2 -- ACK --> UA_B </pre>	<p>Legal or not? Why not? Comments are encouraged also on the legal MSCs (if any of them are legal)</p> <p>Y/N: Correction / Comment:</p>
<pre> graph LR UA_A[UA-A] -- INVITE --> P1[Proxy] P1 -- INVITE --> P2[Proxy] P2 -- INVITE --> UA_B[UA-B] UA_B -- "200 OK" --> P2 P2 -- "200 OK" --> P1 P1 -- "200 OK" --> UA_A UA_A -- ACK --> P1 P1 -- ACK --> P2 P2 -- ACK --> UA_B </pre>	<p>Y/N: Correction / Comment:</p>
<pre> graph LR UA_A[UA-A] -- INVITE --> P1[Proxy] P1 -- INVITE --> P2[Proxy] P2 -- INVITE --> UA_B[UA-B] UA_B -- "200 OK" --> P2 P2 -- "200 OK" --> P1 P1 -- "200 OK" --> UA_A UA_A -- "180 ringing" --> P1 P1 -- "180 ringing" --> P2 P2 -- "180 ringing" --> UA_B </pre>	<p>Y/N: Correction / Comment:</p>

Candidate number

Page ... out of
(fill in all appropriate numbers)**Exercise 3 (16 %) M-IP and GSM****If space is too small, use an ordinary answering sheet to write your answers.**

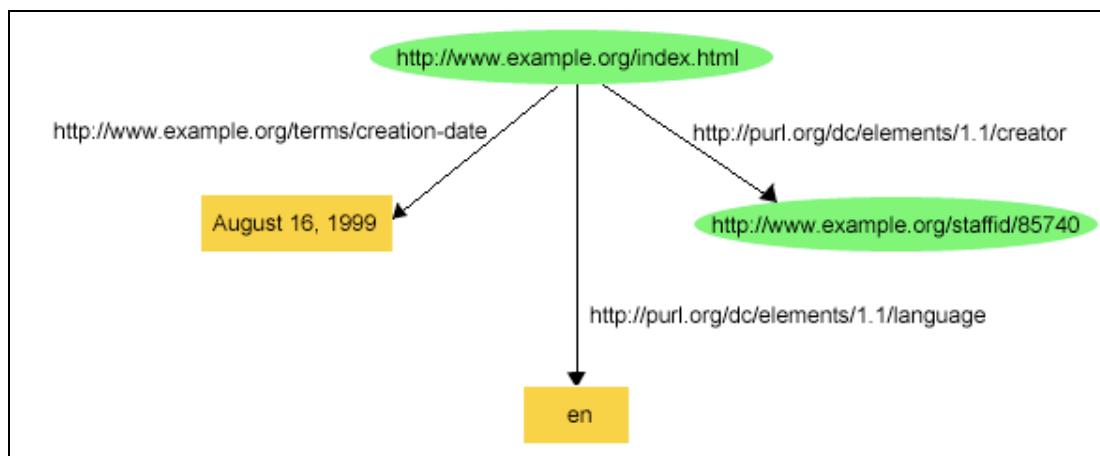
Mobile IP (M-IP) has some similarities with GSM, but there are also several differences.

We assume here that M-IP in IPv4 is used and without route optimization

All the following claims are wrong and you shall correct them. Claims 3-4 are for M-IP in IPv4 and without route optimization.

<p>HLR in GSM is similar to HA in M-IP in the following sense:</p> <p>Claim1: In GSM all voice (media) traffic will pass through the HLR</p>	<p>Claim1 about GSM is wrong. Correct behaviour in GSM is:</p>
<p>HLR in GSM is similar to HA in M-IP in the following sense:</p> <p>Claim2: HLR knows the current location/address of the ‘mobile node’ (which in GSM in this particular case is the called party B), and may inform the ‘corresponding node’ (which in GSM is the calling party A) about the current address.</p>	<p>Claim2 about GSM is wrong. Correct behaviour in GSM is:</p>
<p>VA in M-IP is similar to BS/BSC in the following sense:</p> <p>Claim3: VA1 will perform a soft handover by ensuring that all packets from VA1 will be received also when a new VA2 is involved due to mobility. This will be obtained by VA1 forwarding the packets to VA2</p>	<p>Claim3 about M-IP is wrong because:</p>
<p>HA is M-IP is similar to HLR in GSM in the following sense:</p> <p>Claim 4: HA holds a service profile for each mobile nodes (MNs)</p>	<p>Claim4 about M-IP is wrong because:</p>

Exercise 4 (14%) On resource discovery



Figur 3 An example of a resource described in RDF (via graphical notation)

RDF uses URIs to *identify* things (resources).

a) (4%)

Explain the mechanism used in RDF in order to *describe the properties* of the resources. (You may use Figur 3 to illustrate, if you find it feasible)

XML in general was designed to allow anyone to design their own document format, and then write documents according to the defined format.

b) (4%)

Explain how XML is used with RDF

c) (6 %)

One important property with RDF is that it is machine readable. List at least 3 ways to utilize the machine readable property, or in other words list 3 things that RDF can be used for.

Exercise 5 (12 %) Value added services and IN.

- a) Explain what IP means in the context of IN physical plane, and explain how it is used. This entity supports an SS7 interface and one other interface. Explain which interfaces the IP supports and where the interfaces are connected.
- b) List 4 different logical entities in the distributed functional plane. **Do not** list the entities used for service creation and service management

Assume TelcoX is offering two versions of a service like personal-do-not-disturb. Ordinary-P-DND and Advanced-P-DND. The services differ in the service logic. (The service logic is somewhat similar to the P-DND service explained in the appendix)

- c) Assume Ordinary-P-DND uses a white list of 10 phone numbers (to be let through always). Explain the role of SDP during service execution in this case.

Exercise 6 (14 %) Mobility

a) (6%)

GSM may be said to support two types of terminal mobility. List the types of terminal mobility supported in GSM. Which entities are involved in the different types?

b) (4 %)

Does GSM support personal mobility? Offer a definition of personal mobility make a sensible discussion.

In GSM one separates between discrete mobility and continuous mobility (i.e. mobility during an active call (/session) supporting handover of an active voice stream).

c) (6%)

Explain how Mobile-IP supports mobility during an active TCP session. Explain why it is a bit strange to call this continuous mobility.

Du bør starte med å **lese gjennom hele oppgavesettet**, og deretter velge en rekkefølge for din besvarelse. Merk at 4 ark skal leveres inn.

Korte svar er ønsket.

Gjør dine egne antakelser om teksten er uklar eller om informasjon mangler.

Oppgave 1. (28 %) Om SIP og IMS

a) (3 %)

Angi de 3 typene av CSCF som er i bruk i IMS. Angi også en database.

b) (5%)

Beskriv minst en oppgave/rolle for hver av de 3 typene av CSCF **underveis i registrering**. Totalt kan du oppnå poeng for 5 forskjellige oppgaver (men du skal liste mellom 1 og 2 oppgaver per enhet)

Det er opp til kandidaten å bestemme rekkefølgen mellom b) og c), da det selvsagt er avhengigheter mellom dem.

c) (9 %)

Tegn et meldingssekvensdiagram (MSC eller ‘call flow’) for registeringsprosedyren fra meldingen REGISTRATION er sendt fra UA’en og til en 200 OK er mottatt tilbake til denne UA. Denne prosedyren har **2 faser**, der den første fasen ender med en melding 401 Unauthorized som en utfordring (challenge) tilbake til UA’en.

Hint: Det er noen likheter mellom MSCene for fase 1 og fase 2.

Bruk separat ark på side 11 når denne oppgava besvares.

d) (5 %) Beskriv minst en oppgave/rolle for hver av de 3 typene av CSCF **underveis i samtale/sesjonsoppsettet** (dvs. underveis i ’call setup’-fasen). . Totalt kan du oppnå poeng for 5 forskjellige oppgaver (men du skal liste mellom 1 og 2 oppgaver per enhet).

Det er opp til kandidaten å bestemme rekkefølgen mellom d) og e), da det selvsagt er avhengigheter mellom dem.

e) (6 %)

Anta at både Alice og Bob er korrekt registrert. Så ringer Alice til Bob. Dvs. Alice sender en INVITE fra sin UA via sitt hjemmenett. Etter en stund vil denne INVITE nå Bob’s hjemmedomene (home network).

Del 1: Fyll inn korrekte navn på de involverte CSCF’ene Engelske navn (eller forkortelser) er naturlig å bruke.

Del 2: Fyll inn det rette navnet / responsverdien i SIP for den siste meldingen (ringing)

Del 3: Fullfør den følgende sekvensdiagrammet. Du skal fylle ut alle meldinger mellom den første og den siste meldingen, dvs. det som angår terminererende side
Bruk separat ark på side 12 når denne oppgava besvares.

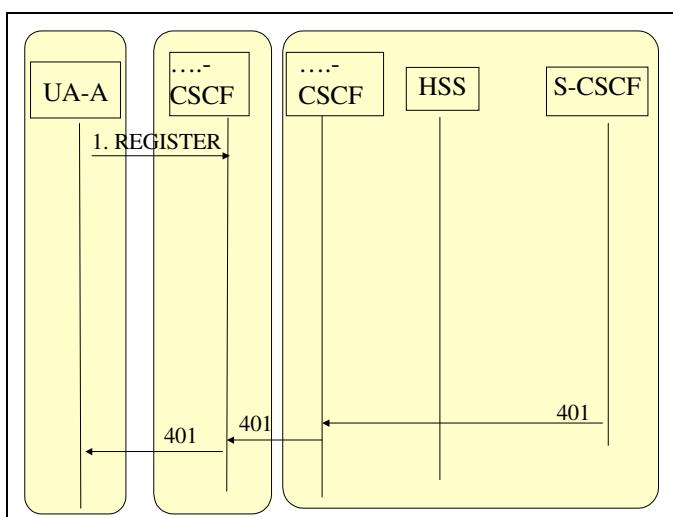
Kandidatnummer

Side av
(fyll inn)**Bruk dette arket når du besvarer Oppgave 1 c) (9 %)**

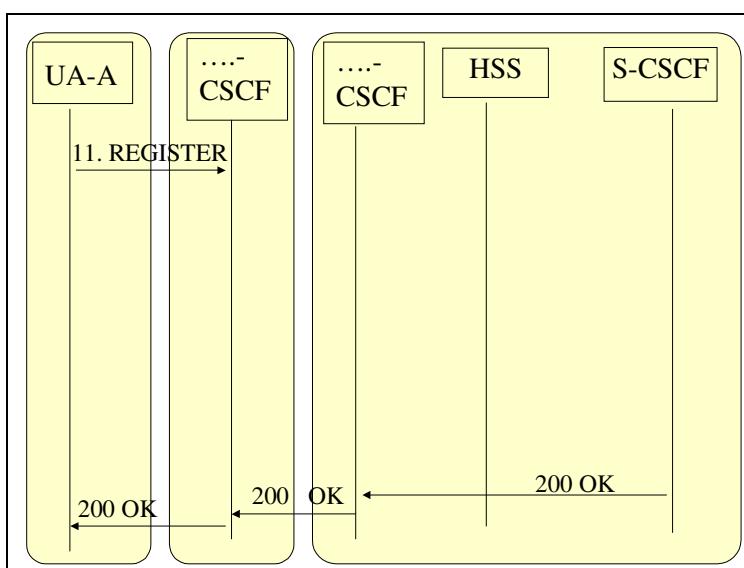
De 3 fargede områdene representerer sluttbrukerområdet, aksessdomenet / besøkt område ('visited domain') og hjemmenettet ('home domain') (til Alice i dette tilfellet).

Del 1: Fyll inn korrekte navn på de involverte CSCF'ene. Engelske navn (eller forkortelser) er naturlig å bruke.

Del 2: Fullfør de følgende to sekvensdiagrammer. Du skal fylle ut alle meldinger mellom den første og den siste meldingen. Den første fasen skal bestå av ca. 8-10 meldinger totalt.



Figur 1Fase1: Den initielle delen av registreringen. S-CSCF utfordrer (asks for a challenge) så den kan autenisere UA-A. Det gjøres via 401.



Figur 2 The andre fasen av registreringen. For enkelhets skyld kan du anta at den første meldingen her har nummer 11.

Kandidat nummer

Side av
(fyll inn)**Bruk dette arket når du besvarer Oppgave 1.e) (6 %)**

De 3 fargeide områdene representerer Bobs hjemmenettdomene ('home domain'), aksessdomenet / besøkt område ('visited domain') og sluttbrukerområdet/domenet.

Del 1: Fyll inn korrekte navn istedenfor ...-CSCF, ...-CSCF og ...-CSCF. Engelske navn (eller forkortelser) er naturlig å bruke.

Del 2: Fyll inn det rette navnet / responsverdien i SIP for den siste meldingen (ringing)

Del 3: Fullfør den følgende sekvensdiagrammet. Du skal fylle ut alle meldinger mellom den første og den siste meldingen. Dvs det som angår terminerende side.

Originerende side	Terminererende side
Fra Alice Via.... Via Alice sitt hjemmenett (home domain)	
	<pre> sequenceDiagram participant Alice participant CSCF1 participant HSS participant CSCF2 participant UA_B Alice->>CSCF1: INVITE (SDP1) activate CSCF1 CSCF1-->>HSS: activate HSS HSS-->>CSCF2: deactivate HSS deactivate CSCF1 activate CSCF2 CSCF2-->>UA_B: deactivate CSCF2 </pre> <p>The diagram illustrates a sequence of messages between network components. It starts with Alice sending an INVITE (SDP1) message to the first CSCF (labeled '.....-CSCF'). This message triggers a query to the HSS. The HSS then sends a message back to the first CSCF. Finally, the first CSCF sends a message to the UA-B (User Agent-B) located in the visited domain.</p>
Til Alice sitt hjemmenett (home domain) For videre ruting til Alice sin UA.	

Oppgave 2 (16 %) SIP protokoll detaljer

a) (5 %) Forklar rollen til hver av de følgende headere i SIP

- Via
- From
- Contact

Spesielt skal forskjellen på From og Contact forklares

b) (3%)

Forklar rollen til headeren Content-type, og hvordan denne brukes med SDP for å oppnå forhandling om mediatyper.

c) (8%)

For hvert MSC angi 'lovlige' eller 'ikke lovlige'. For hvert ulovlig MSC angi hva som er galt. Kommentarer er ønsket også på lovlige MSC'er (for eksempel om det er avvik fra et normalt tilfelle) (Se separat ark som skal leveres inn)

Kandidat nummer.....

Svarark til Oppgave 2 d (8%)Side av
(fyll ut)

Meldingssekvensdiagram (MSC)	Lovlig eller ikke? Hvorfor ikke? Kommentarer er ønsket også på lovlige MSC'er (om noen av dem er lovlige)
<pre> sequenceDiagram participant UA_A as UA-A participant Proxy1 as Proxy participant UA_B as UA-B UA_A->>Proxy1: INVITE activate Proxy1 Proxy1->>UA_B: INVITE deactivate Proxy1 UA_B-->>UA_A: 200 OK UA_A-->>UA_B: 180 ringing UA_B-->>UA_A: 200 OK UA_A-->>UA_B: 200 OK </pre>	J/N: Korrigering / kommentar:
<pre> sequenceDiagram participant UA_A as UA-A participant Proxy1 as Proxy participant UA_B as UA-B UA_A->>Proxy1: INVITE activate Proxy1 Proxy1->>UA_B: INVITE deactivate Proxy1 UA_B-->>UA_A: 200 OK UA_A-->>UA_B: ACK UA_B-->>UA_A: 200 OK UA_A-->>UA_B: ACK UA_B-->>UA_A: ACK </pre>	J/N: Korrigering / kommentar:
<pre> sequenceDiagram participant UA_A as UA-A participant Proxy1 as Proxy participant UA_B as UA-B UA_A->>Proxy1: INVITE activate Proxy1 Proxy1->>UA_B: INVITE deactivate Proxy1 UA_B-->>UA_A: 200 OK UA_A-->>UA_B: ACK UA_B-->>UA_A: 180 ringing UA_A-->>UA_B: ACK UA_B-->>UA_A: 200 OK UA_A-->>UA_B: ACK </pre>	J/N: Korrigering / kommentar:

Kandidatnummer

Side av
(fyll inn)**Oppgave 3 (16 %) M-IP og GSM****Hvis det er for lite plass på dette arket, kan du benytte et vanlig svarark**

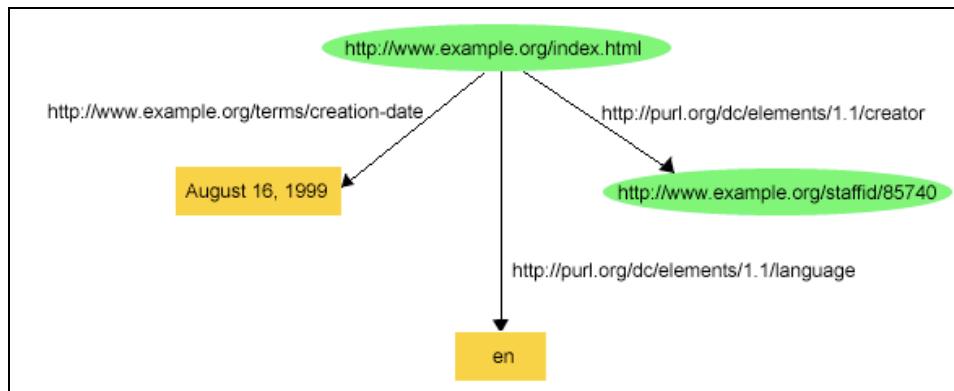
Mobil IP (M-IP) har noen likheter med GSM, men det er også flere forskjeller.

Vi antar her at M-IP i IPv4 er i bruk og at ruteoptimalisering ikke brukes.

Alle de følgende påstandene er gale, og du skal korrigere dem. Påstandene 3-4 gjelder for M-IP i IPv4 uten ruteoptimalisering.

HLR in GSM likner på HA i M-IP på følgende måte: Påstand1: I GSM så går all tale-(media-)trafikk via HLR	Påstand1 om GSM er gal. Den korrekte oppførselen i GSM er som følger:
HLR i GSM likner på HA i M-IP på følgende måte: Påstand2: HLR kjenner til den nåværende lokasjonen/addressen til den 'mobile node' (som i GSM i dette tilfellet er oppringeren, 'called party' dvs. B), og kan informere den 'korresponderende node' (som i GSM er 'calling party' dvs. A) om den nåværende adressen.	Påstand2 om GSM er gal. Den korrekte oppførselen i GSM er som følger:
VA i M-IP likner på BS/BSC på følgende måte: Påstand3: VA1 vil utføre en 'soft handover' ved å sørge for at alle pakker fra VA1 vil bli mottatt også når en ny VA2 er involvert pga. mobilitet. Dette oppnås ved at VA1 videresender pakkene til VA2	Påstand3 om M-IP er gal fordi:
HA i M-IP likner på HLR i GSM på følgende måte: Påstand4: HA har en tjenesteprofil ('service profile') for hver mobil node (MN)	Påstand4 om M-IP er gal fordi:

Oppgave 4 (14%) Om ressursavdekking (resource discovery)



Figur 3 ET eksempel på en ressurs beskrevet i RDF (via grafisk notasjon)

RDF bruker URller til å *identifisere* ting (ressurser).

a) (4%)

Forklar mekanismen brukt i RDF for å *beskrive egenskapene* til disse ressursene. (Du kan bruke Figur 3 til å illustrere, om du finner det passende)

XML generelt var designa for å tillate enhver å definere sitt eget dokumentformat, for så å skrive dokumenter i henhold til det definerte formatet.

b) (4%)

Forklar hvordan XML er brukt sammen med RDF.

c) (6 %)

En viktig egenskap med RDF er at det er maskinlesbart. Angi minst 3 forskjellige måter å nyttiggjøre seg denne maskinlesbarheten på, eller med andre ord: Angi 3 ting RDF kan brukes til.

Oppgave 5 (12 %) Verdiøkende tjenester og IN.

a) Forklar hva IP betyr i konteksten av IN fysisk plan, og forklar hvordan den brukes.

Denne enheten støtter et SS7 interface og et annet interface. Forklar hvilke interfacet IP'en støtter og til hvilke enheter de er tilkobla.

b) List 4 forskjellige logiske enheter i distribuert funksjonelt plan. **Ikke** list enheter som brukes for tjenestekreering og tjenestehåndtering (service management)

Anta TelcoX via IN tilbyr to versjoner av en tjeneste som personlig-ikke-forstyrr (P-DND) (Ordinary-P-DND and Advanced-P-DND) Disse to tjenestene er forskjellige i sine tjenestelogikker. (Tjenestelogikken er for begge noe lik den P-DND tjenesten som er beskrevet i Appendix).

c) Anta at Ordinary-P-DND bruker en hvite-liste (white list) på 10 telefonnummer (som alltid skal slippes gjennom). Forklar rollen til SDP under utføring av tjenesten i dette tilfellet.

Exercise 6 (14 %) Mobilitet

a) (6%)

Forklar hvordan GSM kan sies å støtte 2 typer av terminalmobilitet. Angi disse typene av terminalmobilitet som er støtta i GSM. Hvilke enheter er involvert i de forskjellige typene?

b) (4 %)

Støtter GSM personlig mobilitet? Angi en definisjon av personlig mobilitet og gi en fornuftig diskusjon.

I GSM skiller man mellom diskret mobilitet og kontinuerlig mobilitet (dvs. mobilitet underveis i en pågående samtale (sesjon) som støtter medflytting(handover) av en aktiv talestrøm).

c) (6%)

Forklar hvordan Mobil IP (M-IP) støtter mobilitet underveis i an aktiv TCP-sesjon.
Forklar hvorfor det er litt rart å kalle dette for kontinuerlig mobilitet.

Appendix

All material listed here may be referred to in your answers (if found feasible).

1. “Personal-DND” Service descriptions²

The ‘Personal-do-not-disturb’ (P-DND) is a not standardized service which can be realized in several ways.

Two very different ways to realize such a service can be described as the endpoint centric way and the network centric way. These two approaches will be described below.

1.1. The endpoint centric realization of Personal-DND:

Carl has bought a PSTN answering machine (AM) of the type that is located in his own home. This AM is programmable³ to a certain extent. The AM also contains a button where Carl can register his personal-state as being ‘occupied’. Every second push will change his personal-state from ‘occupied’ to ‘free’ and vice versa and a lamp on the AM will change from red to green accordingly.

Carl has programmed his AM as follows (sketch of service data and service logic)

Data:

```
Black_list: <Up to 10 phone numbers on E.164 format >
COMMENT I do NOT want to talk to these under any circumstances ENDcomment;

White_lists: <Up to 10 phone numbers on E.164 format >
COMMENT I may want to talk to these under certain conditions ENDcomment;

Personal_state: <binary value: 'occupied' or 'free'>
```

Service logic:

```
IF <CLId not present>
    THEN <answer on AM using Message1 ("stupid idiot!")>
ELSEIF <CLId on Black_list>
    THEN <answer on AM using Message1 ("stupid idiot!")>
ELSEIF <CLId on White_list> and personal_state = 'occupied'
    THEN <answer on AM using Message 3 ("leave a message")>
ELSEIF personal_state = 'free'
    THEN <answer call on the phone>;
```

1.2. The network centric way to realize Personal-DND:

The company Tele-Harmonia is a networked 3rd party service provider and uses Parlay/OSA call control. It might be that Tele-Harmonia also uses some User interaction in Parlay/OSA and maybe also some User Status features.

Tele-Harmonia offers a similar service to Carl’s original service, but Tele-Harmonia is offering several additional features:

- Similar to Carl’s programmable answering machine both a personal black list and a personal white list are offered. However the size of each list is now much bigger.
- Tele-Harmonia is also integrating a presence server into their Personal-DND service.

² This material is almost identical to the material given in the ordinary exam. P-DND may differ from the services Ordinary-DND and Advanced-DND as described in Exercise 5.

³ This is an imaginary description. I am not aware of any real Answering machine product aiming for the private market offering these features. However within SIP one may easily assume a SIP B2B UA which could be programmable and behave like this.

2. IMS handling of “caller-ID” and “caller-ID-restriction”

The various caller-IDs and the privacy handling (a kind of caller-ID restriction) are described in the IMS book (in chapter 10.12), and a short version follows here.

In IMS there are several caller identities as illustrated by the following:

```
INVITE sip:Theresa@home2.hu SIP2.0
From: "Your brother" <sip:tobi@brother.com>; tag=veli
To: "My beloved sister" <sip:therera@sister.com>;
P-Preferred-Identity: <sip:tobias@home1.fr>
Privacy: None
```

The P-Preferred-Identity is inserted by the UA (UA of Tobias in this case). But the **originating P-CSCF** will check this identity, by asserting that it is received over a valid IPSec SA (Security Association) and matching with the previous registration. If the originating P-CSCF finds the P-Preferred-Identity to be asserted, it will use this Id, but this time as P-Asserted-Identity.

The SIP message will then be sent to the S-CSCF on the originating side. It may look as follows:

```
INVITE sip:Theresa@home2.hu SIP2.0
From: "Your brother" <sip:tobi@brother.com>; tag=veli
To: "My beloved sister" <sip:therera@sister.com>;
P-Asserted-Identity: <sip:tobias@home1.fr>
Privacy: None
```

In case privacy is “id” the originating S-CSCF must then check whether Theresa’s home network is within the same trust domain. If not the P-Asserted-Identity must be removed.

The **terminating side** must check the P-Asserted-Identity and the privacy header value. In case the value is set to “id”, the P-Asserted-Identity shall not be sent to Theresa’s UA.

However in this case no privacy restriction applies. Thus the “real name” of Theresa’s caller (i.e. <sip:tobias@home1.fr>) will be sent to her UA.

2.1. The call flow resulting from this description is as follows

The call flow can be briefly illustrated as follows (only the initial messages are shown and only the parameters affecting caller-ID is shown). The actions on the way at the CSCFs are indicated in <>-brackets:

```
UA --> INVITE (P-Preferred-Identity: <sip:tobias@home1.fr>) --> P-CSCF
<P-CSCF checks P-Preferred-Identity with prev. Reg. via IPSec SA>
P-CSCF --> INVITE (P-Asserted-Identity:<sip:tobias@home1.fr>) --> S-CSCF
<S-CSCF checks trust relationship with terminating side>
S-CSCF --> INVITE(P-Asserted-Identity:<sip:tobias@home1.fr>) --> I-CSCF
<I-CSCF interacts with HSS and S-CSCF>
S-CSCF --> INVITE(with P-Asserted-Ident.:<sip:tobias@home1.fr>) --> P-CSCF
<P-CSCF checks privacy header not set to "id" >
P-CSCF --> INVITE (with P-Asserted-Identity: <sip:tobias@home1.fr>) --> UA
```

In this call flow it is assumed that the outcome at each check is ‘yes’. If last check results in ‘no’ the P-Asserted-Identity will be withheld.