Norges teknisk-naturvitenskapelige universitet
Institutt for telematikk

EKSAMENSOPPGAVE I TTM4130 – EXAM IN TTM4130
TTM4130 - Tjenesteintelligens og mobilitet
TTM4130 – Service intelligence and mobility SOLUTION p. 2-11

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Eksamensdato / Date:	10. august 2009 / August 10th 2009
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 (authorative)
	(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)
Sensurdato[1]:	**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.

P-CSCF Proxy…
I-CSCF Interrogating…
S-CSCF Serving …
HSS: Home Subscriber Services is the natural DB to list

Comment: Question b and d differ a bit from the description in the book. The book treats each entity, and describesthe role of each in 2.2.1.1, 2.2.1.2 and 2.2.1.3 respect. (i.e. during reg. and call/session setup in one go).

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)

Solution (see IMS book 2.2.1 or A.Roach's presentation):
P-CSCF *acts as initial network entry point for the UE. It is responsibe for the security association i.e. after the registration the P-CSCF knows the final network point (S-CSCF, the reg. ID's and the security association). P-CSCF also knows the address of CDF (charging data function)*
I-CSCF *is the contact point within an operatos network. It may perform THIG (topology hiding). During registration is is responsible for assigning the right S-CSCF (and/or the application server) to be used (based e.g. on user profile data)*
S-CSCF *will during registration store service profiles. These profiles are used during session establishment (somewhat similar to the suppl. services in PSTN in the simplest case) S-CSCF is responsible for generating a challenge to the UE and to verify the response during the registration (see c) below.)*

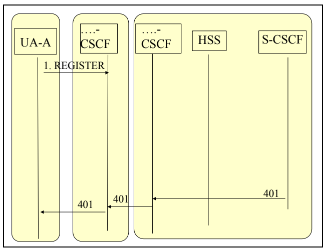
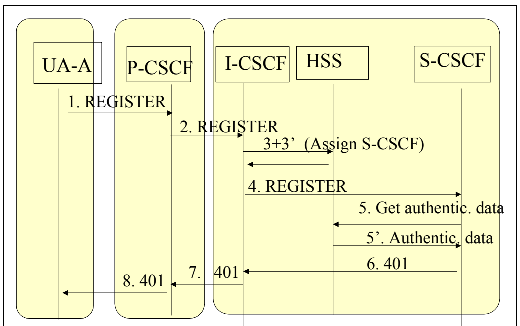
It is up to the candidate to decide on the sequence to answer b) and c), as they are of course interdependent.

(Added: The intetion was that tasks/roles should be something different from 'sending message further to XXX, wgich anyone can see from the MSC)

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.

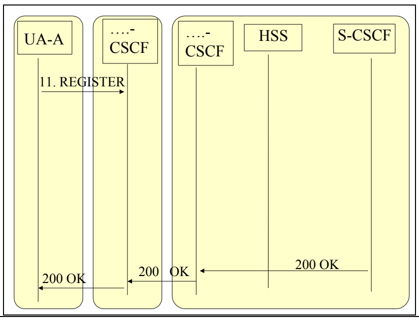
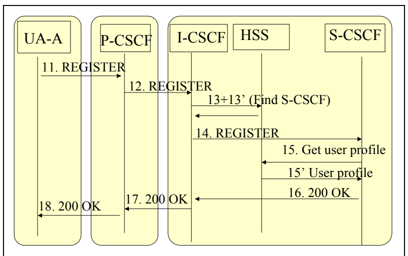
The 3 shaded areas represents the enduser domain, the access (/visited) domain and the home domain (of Alice in this case).

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.

	 Phase 1: the initial part of a registration, S-CSCF asks for a challenge.
	Solution phase 1

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.

Note to solution: It is important that HSS is consulted before the S-CSCF is chosen, i.e. that messages 3+3' is not forgotten. The choice of S-CSCF depends on the service profile (and more, maybe on load) and a lookup in HSS is needed before assigning S-CSCS. (Exact names of these messages are not shown in fig. 3.1 in IMS book, and is not required in a solution, similar comment for 5+5')

	 Phase 2: the new registration, with an answer to the challenge From 3.1 but using slightly different numbers, starting with 11. 13+13' and 15+15' are shown without exact name of messages
	Solution phase 2

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

Comment: The numbering should now make it easy to see the similarities between the first and second phase 4 corr. to 14 etc. Note some differences in mess. 3 and 13 and 5 and 15. It is a minor fault if 13+13' is forgotten. It should be shown that HSS transfers user profile to S-CSCF (15+15') (called 15 in IMS book, and shown as two-way arrow)

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).

Solution (1 point per correct tasks, other tasks than those listed here may be honored as well):
P-CSCF: *It will **relay session and media information** in case operator wants to use PDF and SBLP (session based local policy) and may be responsible for **emergency calls**. P-CSCF is responsible for **SIP compression of incoming calls** (e.g. of an INVITE coming from a calling party via this P-CSCF targeting a UE of the called party, this will not be visible in the call flows below though).*
***I-CSCF** is responsible **for routing** incoming requests to the right S-CSCF (**THIG** may be used both during reg. and during call setup and may be hounored twice)*
***S-CSCF** is responsible for **maintaining the session state** and **invoke supplementary services** both for incoming calls (terminating side) as well as for outgoing calls (originating side). It is also responsible for **routing** the call further (e.g. to a BGCF or to an I-CSCF for the termninating side.*

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

(Added: The intetion was that tasks/roles should be something different from 'sending message further to XXX, wgich anyone can see from the MSC)

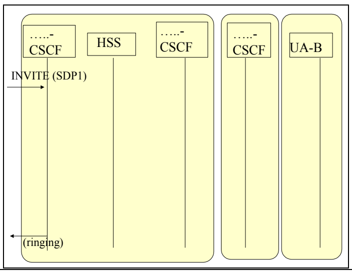
Candidate number ………………..

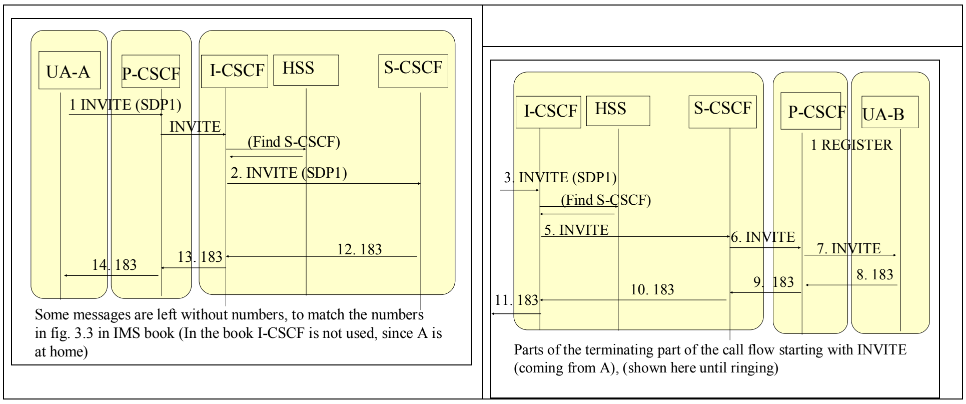
Page …. out of …….
(fill in all appropriate numbers)

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.

Originating side	Terminating side
From Alice Via…. Via Alice's home domain	
	
To Alice's home domain For further routing to Alice's UA.	



Some messages are left without numbers, to match the numbers in fig. 3.3 in IMS book (In the book I-CSCF is not used, since A is at home)

Parts of the terminating part of the call flow starting with INVITE (coming from A), (shown here until ringing)

Figur 3 This MSC is drawn as 2 figures in .ppt. If proper SDL/UML/MSC is used message 3 INVITE shall go from (A's / left) S-CS.. to I-CS.. in the right part. Ditto for 11 (received by S-CS.. before 12 is sent)

Exercise 2 (16 %) SIP protocol details

a) (5 %) Explain the role of the following header fields in SIP

- Via
- From
- Contact

In particular explain the *difference* between From and Contact

Solution:
***Via** (2p) is used to force the signaling to follow a given path, by adding and deleting entities to the 'stack'. This makes the methods and the replies follow the same path. One example of use is to tell the signaling to follow a path via I-CSCF. (In this way THIG (topology hiding) may be obtained).*
***From** (1p) is your public ID suitable to list on a business card, and to show in a service showing 'lost calls' or other call logs. **Contact (1p)** is the current contact point, valid for as long as the current registration and hence not suitable to place on a business card **(1p for the difference)**.*

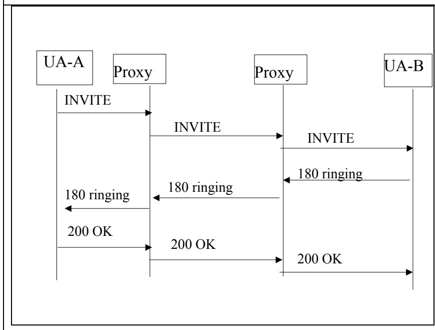
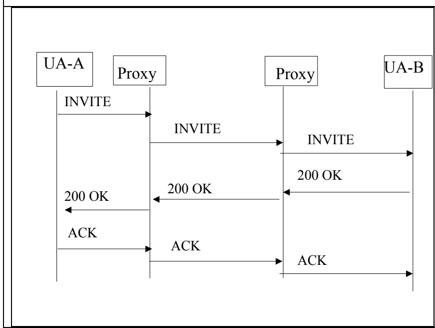
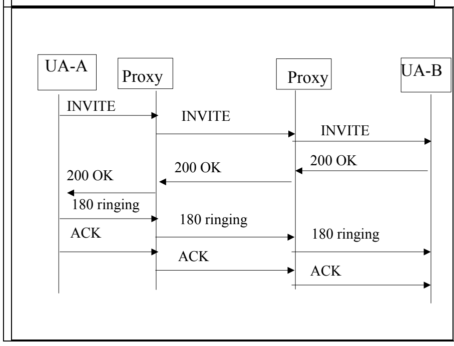
b) (3%)
Explain the role of the header Content-type, and how this is used with SDP to do media negotiation.
Solution:
*Content type is used to tell the content type (!), i.e. the type of the payload carried **(1p)** . In SIP the content type is most often SDP, so that the SDP details are sent as payload in the INVITE, 200 OK and ACK messages **(1p)** . In this way media negotiation is obtained **(1p)***

Example of media negotiation is obtained: A proposal for media may be sent as SDP1 in INVITE, and a subset of this (SDP2) may be sent back in 200 OK. SDP2 will then be ack'ed in a new message ACK. Typical case is INVITE from A to B, and ACK also from A to B. More complicated negotiations may also take place

[The mechanism can also be used for re-negotiation of media types, and in such a case also B may propose new media types SDP5 in a (Re)-INVITE) and A may answer with SDP6 in 200 OK, B may ack this. (For the very interested reader the initial INVITE may have empty payload (SDP=Ø) and B2BUAs may also change SDP)]
Some students mixed content-type in SIP with media types in SDP, not good!

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)

Answering sheet to 2 d (typo for c) (8%)

Message sequence chart	Legal or not? Why not? Comments are encouraged also on the legal MSCs (if any of them are legal)
	Y/N: N (1p) Correction: (1—2 points) 200 OK goes in wrong direction (it shall always be a response to INVITE) This MSC breaks the rule that all responses (1xx, 2xx,++) shall be a response to a previous method (which goes in the opposite direction) Ack'ing of 200 is done by a sep. method ACK (See next MSC for a correct version)
	Y/N: Y (1 p) Comment (1 p.) It is a bit strange not to use 180 ringing before 200 OK, but it follows the rules of SIP methods: Each method (like INVITE) followed by one or more responses, ending with a final response. Also the direction of ACK is correct In a typical call flow (both 100 and) 180 may be inserted from right to left before 200 OK.
	Y/N: N (1 p.) Correction: (1-2 p.) Since 200 is a final response, 180 cannot be sent after 200 Also 180 goes in the wrong direction Comment: (1p) Before 200 is sent responses such as 100 trying and 180 ringing may be used (from right to left)

[for first and 3rd MSC: It might be that a not shown INVITE has gone from B to A previously, (but we assume this is not the case here) Session ID, seq.number etc, may be used to detect such cases.
In the cases explained above 200 OK in first case (/ 180 in third case) may belong to this different session establishment, and the MSC is (partial) but correct]
(this is not needed in an answer)
Just answering Y in case 1 and case 3 gives 0p, the 'lacking' (previous INVITE must be mentioned to obtain points for a 'Y' in case 1 and 3.

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.

HLR in GSM is similar to HA in M-IP in the following sense: **Claim1**: In GSM all voice (media) traffic will pass through the HLR	Claim1 about GSM is wrong. Correct behaviour in GSM is: *In GSM only the signaling will go through the HLR, while the voice/media will follow a path between MSC's.*
HLR in GSM is similar to HA in M-IP in the following sense: **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.	Claim2 about GSM is wrong. Correct behaviour in GSM is: *Both HA and HLR knows the current location (/address) of the mobile node (MN). (In fact HLR may know only indirectly via MS-RN) It is however wrong that HLR in GSM will inform the caller (corresponding node about this location. Such information is kept within the network(s) in GSM*
FA in M-IP is similar to BS/BSC in the following sense: **Claim3:** FA1 will perform a soft handover by ensuring that all packets from FA1 will be received also when a new FA2 in involved due to mobility. This will be obtained by FA1 forwarding the packets to FA2	Claim3 about M-IP is wrong because: *There are no mechanisms for soft handover in M-IP. Instead all lost packets (typically those arriving at FA1 after MN has moved to FA2) will be retransmitted (from the CN via HA) and will be received via FA2 (hopefully in the second attempt). This works fine for TCP and file transfer, but is not suitable for real time services (voice etc.).*
HA is M-IP is similar to HLR in GSM in the following sense: **Claim 4:** HA holds a service profile for each mobile nodes (MNs)	Claim4 about M-IP is wrong because: *HLR is involved in supplementary services like CFU, CFNR ++. A service profile indicates some service differentiation (based on e.g. subscription data) Such service diff. is unknown to M-IP.)* *It is wrong that HA has any notion of service profiles. The only service a HA does is to forward all packets and to keep track of CoA in order to perform this task.*

Comment: Oral information on exam about the VA/FA typo.

lillk 8/11/09 11:00 AM
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lillk 8/11/09 11:04 AM
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lillk 8/11/09 11:00 AM
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lillk 8/11/09 11:04 AM
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lillk 8/11/09 11:00 AM
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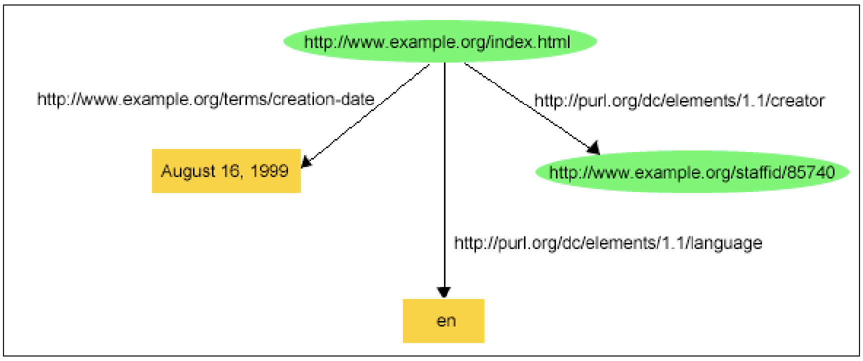
Exercise 4 (14%) On resource discovery



Figur 4 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 feasable)

*Solution: Resources are described to properties and property values. (**4p** for this simple general answer) As one example: The resource (http://www.example.org/index.html) is described by 3 properties (creator, language and date): creator with property value [...]staffid/85740), property: language (value: en (meaning English) and property: creation-date with value Aug. 16th 1999 **(also 4p** for using the example way to explain, max 4 p))*

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

Solution: RDF defined a specific version of XML, namely RDF/XML. This is used to represent the RDF information is a machine readable form, and to exchange the information between machines.

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.

Solution: Any of these gives 2 point each (max 6 point). Other good answers will also e honored

- *For resource discovery:– In order to improve search routines*
- *To catalog items: Description of contents and interrelationship between different content object on a web page.*
- *For content rating*
- *Description of an assembly of pages that constitutes a common "logical" document*
- *By intelligent agents:– As a tool that enables knowledge transfer*

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.

Solution: (4 %) IP means Intelligent Peripheral, and is a physical entity realizing some SRF functionality. Typically used to play announcements and/or to record messages fir voice mail etc. SS7 is used between SCP and IP, while both SS7 and circuit switch voice is established between IP and SSP.

b) List 4 different logical entities in the distributed functional plane. **Do not** list the entities used for service creation and service management

Solution: (4 %) SSF, SCF, SDF, SRF, (alternatively: CCAF, CCF) (not: SMF, SCEF)

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.

Solution: (4%) The same service logic may be populated with different data for each user. E.g. each user may have their personal phone numbers stored in SDP. Also the current status for this user (busy/free) may be stored in SDP (and fecthed upon activation of the service). When the service is activated ('triggered') the SCP behaves according to the (common) service logic, and asks the SDP for user specific data (such as the personal list)

Comment (beyond what is expected): This allows the SCP to behave like a state machine, and allows scalability via reentrance. (This is further elaborated in ttm4160 where implementation of state machines are gone through in detail)

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?

Solution: GSM supports both discrete term. mob. (DTM) and continuous terminal mobility (CTM). DTM is obtained via registration when turning on the phone, and by update messages when moving when not in a call. DTM involves TE, BS, BSC, MSC,HLR and VLR. CTM is obtained via Handover procedure and involves BS1, BS2 and BSC in the normal case. (In some cases several BSC may be involved).

Students answering inter-domain mobility vs intradomain mob (roaming to foreign netw.) is also obtaining full score

b) (4 %) Does GSM support personal mobility? Offer a definition of personal mobility make a sensible disussion.

Personal mobility is defined proximately as " the ability to use all subscribed (personal) services regardless of the physical terminal currently in use" (slight modification from LN p. 57)
(alternatively look up def. In Teddy of in NGN):

Def 2: Mobility for those scenarios where the user changes the terminal used for network access at different locations (from TR 180 001 V1.1.1 which is part of the cyllabus.
Def. 1 is also relevant: ability of a user to access telecommunication services at any terminal on the basis of a personal identifier, and the capability of the network to provide those services according to the user's service profile (from SPAN more similar to LN def.)

It is not easy to move a SIM card between terminals in GSM (but it is possible when not in a call, sometimes after removal of the battery). Hence it is not so easy in GSM to use an 'arbitrary' (public) terminal and obtain your own services. Natural answer is no: GSM does not offer personal mobility, but offer a very personal phone with great terminal mobility (also internationally).

A good descussion on how SIM-card mob. relates to pers.mob. is also honoured (must be good though)

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.

Solution: M-IP supports mobility during a TCP session by retransmission of lost packets to the new FA. I.e. the CN does a retransmission via the HA to the (new) FA. This is obtained via buffer mechanisms existing in TCP (3 p)

However TCP ***has no strict timing requirements****. Continuity is a term normally associated with 'very small' intervals[2], not with the time intervals (in seconds) involved in TCP. (3 p)*

Note that there are two things that shall be answered here.

lillk 9/23/09 4:01 PM
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[2] In mathematical terms one do not speak about 'very small' but 'arbitrarily small',as in the definition of a contnious function f: For all epsilon>0 there exists a delta >0 such that [abs (d) <delta implies abs(f(x+d) –f(x)) <epsilon].
In real time voice arbitrarily small time intervals have only theoritical interests. The human sensor (ear, brain etc) determines the limits. This limit is 'very small' (compared to the many seconds delays allowed in TCP).

Du bør starte med å **lese gjennom hele oppgavesettet,** og deretter velge en rekkefølge for din besvarelse. Merk at 4 ark skal leveres inn. De norske sidene i løsningen viser original sidebrekking. Her vises det ingen løsninger

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 avhengiheter 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 avhengiheter 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 terminerende side
Bruk separat ark på side 12 når denne oppgava besvares.

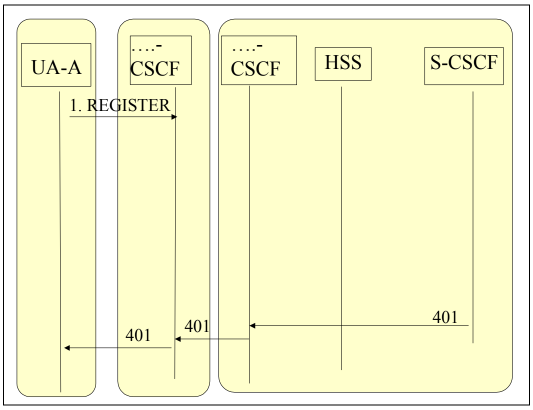
Kandidatnummer ………………..

Side ….. av ……….
(fyll inn)

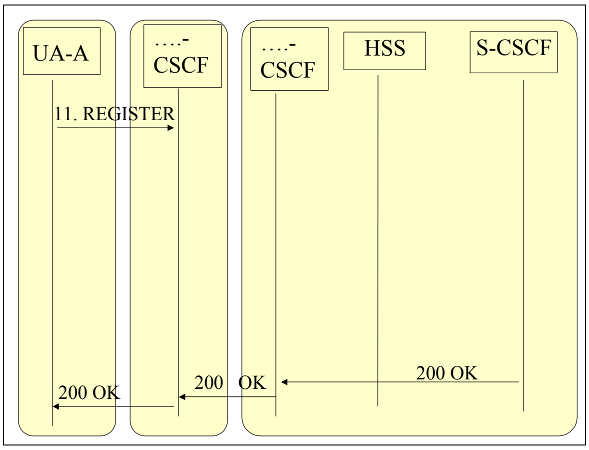
Bruk dette arket når du besvarer Oppgave 1 c) (9 %)

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 regisreringen. 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.

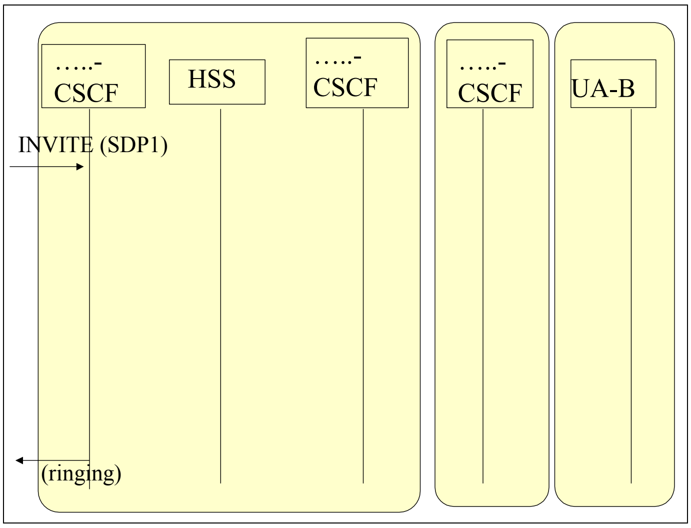
Kandidat nummer ………………..

Side ….. av ……….
(fyll inn)

Bruk dette arket når du besvarer Oppgave 1.e) (6 %)

De 3 fargede 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	***Terminerende side***
Fra Alice Via…. Via Alice sitt hjemmenett (home domain)	
	
Til Alice sitt hjemmenett (home domain) For videre ruting til Alice sin UA.	

Oppgave 2 (16 %) SIP protokolldetaljer

a) (5 %) Forklar rollen til hver av de følgende headere i SIP
- Via
- From
- Contact

Spesielt skal forskjellen på From og Contact forklares

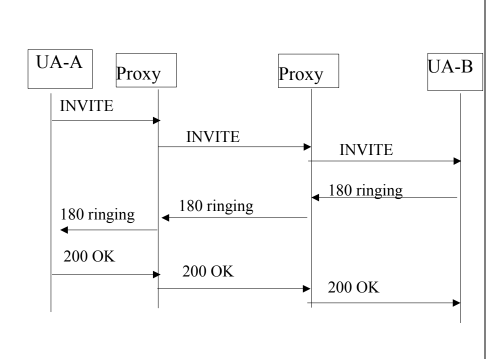
b) (3%)
Forklar rollen til headeren Content-type, og hvordan denne brukes med SDP for å oppnå forhandling om mediatyper.

c) (8%)
For hvert MSC angi 'lovlig' eller 'ikke lovlig'. 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)

<table>
<tr><td>Meldingssekvensdiagram (MSC)</td><td>Lovlig eller ikke?
Hvorfor ikke?
Kommentarer er ønsket også på lovlige MSC'er (om noen av dem er lovlige)</td></tr>
<tr><td>


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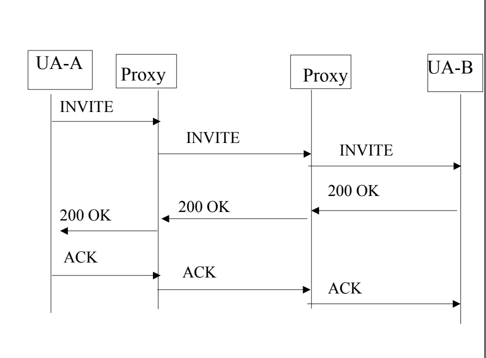

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Korrigering / kommentar:</td></tr>
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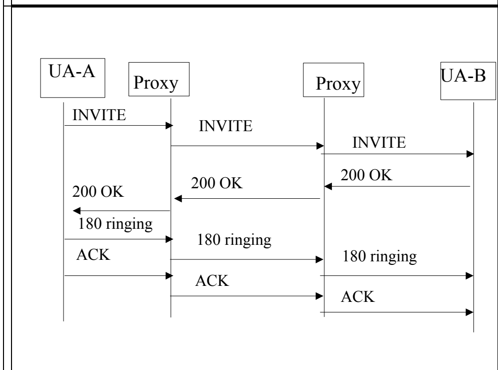

</td><td>J/N:
Korrigering / kommentar:</td></tr>
</table>

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:
FA i M-IP likner på BS/BSC på følgende måte: **Påstand3:** FA1 vil utføre en ‘soft handover’ ved å sørge for at alle pakker fra FA1 vil bli mottatt også når en ny FA2 er involvert pga. mobilitet. Dette oppnås ved at FA1 videresender pakkene til FA2	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:

comment: Oral info. om exam about the VA/FA typo.

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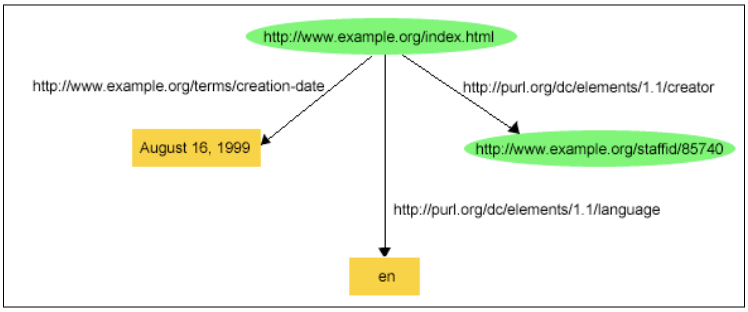
Oppgave 4 (14%) Om ressursavdekking (resource discovery)



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

RDF bruker URIer 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 interfacer 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 %)
Støter 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 refered to in your answers (if found feasable).

1. "Personal-DND" Service descriptions[3]

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 as service can be the 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 programmable[4] 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 circomstances 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.

---

[3] 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.

[4] 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.

Comment in the solution: Appendix 2 was kept, as it might be useful in exercise 1

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