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:	4 juni 2009 / June 4th 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:	4 (pages 2-5)
Antall sider bokmål:	4 (sidene 6-9)
Antall sider nynorsk:	0
Appendix (in English):	7 (pages 10- 16 with text)
Sheets for drawings	2 sheets which may be handed in
Sensurdato[1]:	**25. juni 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. In some places there are some dependencies that are listed, but it is possible to start e.g. with exercise 3, 4, 5 or 6 if you do not want to start with SIP. 2 sheets with MSC diagrams to be filled in by the canditate may be used to save some drawing and are attached at the end. **Short answers are requested.**

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

Exercise 1. (26 %) About SIP (SIP proper as defined by IETF)

a) (8 %) List 4 of the main entities in the SIP architecture and describe the main tasks of each of the entities.(UA-C + UA-S = US, this counts as one entity). Do not list DNS

b) (9 %)
Explain the role of SDP in SIP.
List also other major protocols to be used with SIP and explain how they complement the functionality in SIP.
Explain how SIP is similar to HTTP and explain also some major differences

c) (6 %)
Show a call flow for call/session establishment between Alice (caller) and Bob (callee/ called party) until the media flow is established.
- You may use an MSC, or you might use a 'free' drawing showing a call flow with message numbers You may use the "SIP trapezoid" . You may assume SDP1 is a voice proposal and that this will be accepted by B.
- You need *not* include any re-negotiation of SDP. You need *not* include forking
- You *shall include* methods (like INVITE and more) and response messages (like 180, 200) etc. all the way until the media flow is established.

d) (3 %)
Show how Bob propose new media types and how this proposal is accepted by Alice. You may let SDP2 represent this new voice+video proposal. This message exchange shall take place after the call flow in c) is sucessfully carried out.
- *Note:* To ease the sensors reading of this call flow you shall number the first message now as message 21 (regardless of how many messages you actually used in c).
- You *should not* "crash" the new messages onto the drawing used in c) in a manner that makes it unreadable. (A small MSC where each message follows in a timely fashion should do the job here.)

Exercise 2 (9 %) SIP protocol details (SIP proper)

a) (6 %) Call-ID is one header field in the INVITE method.
Task: List *4 other header fields* used in INVITE . For each of the chosen header fields you shall explain briefly how they are used (i.e. what meaning they have)
- You should preferably include header fields that you find most important (though all header fields are of course important in different ways). Try *not* to include header fields that are specific for IMS.

b) (3%) Illustrate a SIP transaction which starts with INVITE and contains 3 messages. You may make a new example or re-use a part of a call flow from exercise 1.
- *Please make a new drawing* to ease the reading for the sensors.
- It is also *recommended* that this drawing separates UA-C and UA-S.

Exercise 3 (15 %)

a) Explain Mobile IP (M-IP). Explain the main entities involved and the basics of how M-IP works.

b) Explain how route optimization in M-IP works when using IPv6.

c) Explain two major issues if M-IP is to be used with realtime multimedia streams. You shall not cover the route optimization from b)
Hint: Think of how GSM handles mobility in the visited network
Hint: IETF has a proposal using 'Hierarchical M-IP' (HMIP) and using a Mobility Anchor Point (MAP) in the network visited by the mobile node

Exercise 4 (24 %) Miscellaneous

a) (3 %) What is this?
The question is 'what', not the 'meaning behind' the text. A short answer is requested.

```
v=0
o=Alice 2790844676 2867892807 IN IP4 192.0.0.1
s=Let's talk about swimming techniques
c=IN IP4 192.0.0.1
t=0 0
m=audio 20000 RTP/AVP 0
a=sendrecv
m=video 20002 RTP/AVP 31
a=sendrecv
```

b) (6 %)
Explain briefly some business considerations behind Parlay/OSA.
How are business considerations in Parlay/OSA similar and different from IN?

c) (15 %) Answer the following questions with yes or no: (Guessing gives no credit)

1. The entity in IN conceptual model called SCF is a logical entity.
2. The entity in IMS called S-CSCF is a physical entity.
3. The framework (FW) in OSA is a Java Framework allowing 3rd party programmers to write program code to be run on the HW owned by the network operator owning the FW.
4. In IMS the HSS is the entity where service execution of value added (or supplementary) services takes place (An example of such a service is some kind of call forwarding)
5. Mobile IP was initially designed to be used for nomadic users, but has later turned out to be an optimal protocol to handle mobility for VoIP.
6. In IN SS7 is used for the signalling between SSP-SCP
7. When a voice-menu is used in IN (press 1 for English etc.) then a voice-connection will be established towards the SCP.
8. Parlay/OSA standarizes the interactions (interfaces) between a 3rd party and the network via so-called SCS (Service Capability Servers)
9. Parlay/OSA standarizes the interactions (interfaces) between so-called SCS (Service Capability Servers) and nodes like S-CSCF and HLR

Exercise 5 (14%) Value added services, service management and IN.

Supplementary services are often standardized, but an interesting issue is to allow rapid introduction of new value added services *without* needing to standardize the service description or add new parameters (e.g. to the Establish method).

One example of such a service may be Personal-DND service. Such a servie may be realized in different ways and have various service logics and associated data. See appendix section 3 for more description of some Personal-DND services.

Some of the description will also use the term malicious call. See appendix section 2.4

We will follow Alice and Carl during a time where Carl and Alice starts dating:

- Alice has 2 different phone subscriptions (one on PSTN and one on GSM).
- Carl has the personal-DND service as described in Appendix section 3.1
- At Time T1 Carl wants Alice to pass through his Personal-DND service.
- Later Alice is threatened by her ex-boyfriend David. She finds out that she needs to change to new phone numbers.
- At time T3 Alice gets new phone numbers that are ’non-listed’ (but not strictly secret). This means that her new phone numbers will not be searchable on 1881.no or similar yellow/white pages services. She chooses to keep the CLIP supplementary service.
- After some more time Alice realizes that some of her friends have leaked her mobile phone number to David and she is getting threatening calls at night.
- At time T4 she is changing phone numbers again, still opting for ‘non-listed’ numbers and this time she is determined to subscribe to CLIR (Calling Line Identity Restriction) on both her PSTN and GSM numbers

a) (6 %)

- **Subtask:** Explain briefly what actions Carl must do at/after time T1 in his private ’service management system’ on his answering machine. (This service management system is probably manual)
- **Subtask:** Explain briefly what actions Carl will need to take at/after time T3 on his answering machine in order for his personal-DND service to work properly (i.e. letting Alice through).
- **Subtask:** Explain what difficulties Carls enters into at Time T4. Is it enough to update with the new phone numbers of Alice? How can the service from Tele-Harmonia described in appendix section 3.2 serve Carl under these new conditions in a way different from the endpoint centric version

b) (8 %)

Explain briefly how the PSTN/ISDN network operator may create/build/realize some Personal-DND service via IN. You may choose to describe *either* the ”global functional plane” (using BCSM and SIBs) or the details of the physical realization (using SSP, SCP, IP etc) in the so-called ”Physical Plane”

- It is requested that you describe *only one* of these two planes (of your own choice)
- It is not expected that you remember names of SIBs. You may uses SIBs named MESSAGE, CONNECT and RELEASE and optional other SIBs of your choice.

Exercise 6 (12%) Service mobility

Assume in this exercise that Alice and Bob subscribes to a standardized supplementary service SS1. In order to be a bit more concrete you may assume that SS1 is either:

- CLIR (Calling Line Identity Restriction) (subscribed to by Alice) or
- CFNR (Call Forwarding on No Reply) (subscribed to by Bob)

You may choose between CFNR and CLIR as you find most appropriate depending on your knowledge from the syllabus and your ability to find useful information in the appendixes. Note that CLIR in GSM is briefly explained at the end of Appendix section 2.2.

Assume also that Bob is currently roaming into a foreign domain and is assigned an MSC in this domain. Assume also that Alice is a GSM subscriber.

a) (4 %) Explain the role of the HLR for service profile for the service SS1 and how the HLR interact with other entities in order to realize SS1.
   - I.e. explain how SS1 (either CLIR or CFNR) in realized in GSM, by using Alice and Bob as concrete examples. Alice is at home (in GSM) calling Bob.
   - Hint 1: You may look at Figure 1, (but change the originating side a bit since now Alice is on GSM). Before this call takes place Bob has registered and you should find out the call flow (sequence diagram) for Bob registering at this location. Since this registration naturally involves the HLR the answer to the question lies here.
   - Hint 2: It might be useful to recall how HSS in involved in an IMS registration for service profile data and S-CSCF selection, *though the procedures are not exactly the same in GSM.*
   - Hint 3: Unless you remember how CFNR in realized in GSM, you may want to look at how CLIR is realized in PSTN in appendix section 2.2.

The following definitions of service mobility are given is the various syllabus material.

Def.1 (from Lecture Notes (LN) translated to English): **Service mobility** (alternative names: program mobility / software mobility or actor mobility) will allow SW modules / actors (like code, objects and processes) to be moved from one machine to another machine

Def. 2 (from ETSI T TR 180001, Tispan, NGN release definition): **Service mobility**: mobility, applied for a specific Service

- NOTE: I.e. the ability of a user to use the particular (subscribed) service irrespective of the location of the user and the terminal that is used for that purpose.

b) (4 %) Does GSM offer service mobility for CFNR and CLIR according to Def.2? Make a sensible (short) discussion in addition to answering yes or no.
   - Hint: You may find some useful information in a correct answer to 6.a)

c) (4 %) Does GSM use service mobility according to Def.1 for a service like CLIR (Calling Line Identity Restriction) or CFNR (Call Forwarding on No Reply). Make a sensible (short) discussion in addition to answering yes or no.
   - Hint: You may find some useful information in a correct answer to 6.a)

NORSK TEKST HUSK AT ENGELSK TEKST ER AUTORITATIV (MASTER)
Du bør starte med å **lese gjennom alt materialet**, og så bestemme deg for en rekkefølge du ønsker å starte i. Noen steder er det indikert noen avhengigheter, men du kan starte med f.eks. oppgave 3, 4, 5, eller 6 hvis du ikke ønsker å starte med SIP. **Korte svar er ønskelig.**

Lag dine egne antakelser om teksten er uklar eller det mangler informasjon.

Comment [I1]: Disse sidene må oversettes. Jeghar såvidt begynt..

Pultene på eksamen er små så sidebrekkingen er derfor svært viktig

Juster evt. ned ved manuelt å sette font 10 isteden (hvis nødvendig.)

Oppgave 1. (26 %) About SIP (SIP 'proper' (rein) som definenert av IETF)

Comment [I2]: Her har jeg ikke brukt autonummerering, vi klarer oss uten det

a) (8 %)
List 4 av hovedentitene i SIP arkitekturen and describe the main tasks of each of the entities.(UA-C + UA-S = UA teller som en entitet). Du skal ikke liste DNS

b) (9 %)
Forklar rollen til SDP in SIP.
List opp andre viktige protokoller som brukes sammen med SIP og forklar hvordan de kompemeterer funksjonalitet i SIP.
Forklar hvordan SIP er lik HTTP og forklar noen viktige forskjeller mellom SIP og HTTP

c) (6 %)
Show a call flow for call/session establishment between Alice (caller) and Bob (callee/ called party) until the media flow is established.
- You may use an MSC, or you might use a 'free' drawing showing a call flow with message numbers
- You need *not* include any re-negotiation of SDP. You may assume SDP1 is a voice proposal and that this will be accepted by B. You need *not* include forking
- You *shall include* methods (like INVITE and more) and response messages (like 180, 200) etc. all the way until the media flow is established.

d) (3 %)
Vis hvordan Bob foreslår nye media typer og at dette forslaget blir aksepter av Alice. Du kan anta at SDP2 representere dette nye tale+video forslaget. This message exchange shall take place after the call flow in c) is sucessfully carried out.
- *Note:* To ease the sensors reading of this call flow you shall number the first message now as message 21 (regardless of how many messages you actually used in c).
- Du skal ikke 'kline' de nye meldingene oppå tegninga brukt i c) på en slik måte at det blir uleselig. (En liten MSC hvor hver melding følger på hverandre i en tids-orden skulle gjøre jobben her.)

Oppgave 2 (9 %) SIP protokoll detaljer (SIP 'proper' / rein SIP)

a) (6 %) Call-ID is one header field in the INVITE method.
Task: List *at least 4 other* header fields used in INVITE . For each of the chosen header fields you shall explain briefly how they are used (i.e. what meaning they have) You should preferably include header fields that you find most important (though all header field are of course important in different ways). Do *not* include header fields that are specific for IMS.

b) (3%) Illustrate a SIP transaction which starts with INVITE and contains 3 messages. You may make a new example or re-use a part of a call flow from exercise 1.
- *Please make a new drawing* to ease the reading for the sensors. It is also recommended that this drawing separates UA-C and UA-S.

NORSK TEKST HUSK AT ENGELSK TEKST ER AUTORITATIV (MASTER)

Comment [I3]: Selv med denne 'advarselen/ fraskrivelsen, så bør det trippelsjekkes at det er rimelig overenstemmelse mellom de to tekstene

Oppgave 3 (15 %)

a) Explain Mobile IP (M-IP). Explain the main entities involved and the basics of how M-IP works.

b) Explain how route optimization in M-IP works when using IPv6

c) Explain two major issues if M-IP is to be used with realtime multimedia streams. You shall not cover the route optimization from b)

Oppgave 4 (24 %) Miscellaneous

a) (3 %) What is this?
The question is 'what', not the 'meaning behind' the text. A short answer is requested.

```
v=0
o=Alice 2790844676 2867892807 IN IP4 192.0.0.1
s=Let's talk about swimming techniques
c=IN IP4 192.0.0.1
t=0 0
m=audio 20000 RTP/AVP 0
a=sendrecv
m=video 20002 RTP/AVP 31
a=sendrecv
```

b) (6 %)
Explain briefly some business considerations behind Parlay/OSA.
How are they similar and different from IN?

c) (15 %) Svar på følgende spørsmål med ja eller nei. (Rein gjetning gir ingen poeng)

- 1: Entiteten i IN sin konseptuelle modell som kalles SCF er en logisk entitet.
- 2: Entiteten i IMS som kalles S-CSCF er en fysisk entitet.
- 3: The framework (FW) in OSA is a Java Framework allowing 3rd party programmers to write program code to be run on the HW owned by the network operator owning the FW.
- 4: In IMS the HSS is the entity where service execution of value added (or supplementary) services takes place (An example of such a service is some kind of call forwarding
- 5: Mobile IP was initially designed to be used for nomadic users, but has later turned out to be an optimal protocol to handle mobility for VoIP.
- 6: I IN brukes SS7 for signalleringen mellom SSP-SCP
- 7: Når en tale-meny brukes i IN ("Trykk 2 for norsk" etc.) så blir det etablert en tale-kanal til SCPen.
- 8: Parlay/OSA standardiserer interaksjonene (interfacene) mellom en 3dje part og nettet via såkalte SCS (Service Capability Servers)
- 9: Parlay/OSA standardiserer interaksjonene (interfacene) mellom så-kalte SCS (Service Capability Servers) og noder som S-CSCF og HLR.

NORSK TEKST HUSK AT ENGELSK TEKST ER AUTORITATIV (MASTER)

Oppgave 5 (14%) Verdiøkende tjenester, tjenestehåndtering og IN.

Verdiøkende tjenester er ofte standardiserte, men en interessant problemstilling er å få til raskt å lage nye verdiøkende tjenester *uten* å behøve å standardisere tjenestebeskrivelsen eller å måtte innføre nye parametre (f.eks. i Establish meldingen)

Et eksempel på en slik tjeneste er Personlig-DND tjeneste. En slik tjeneste kan realiseres på forskjellige måter og ha forskjellige tjenestelogikker og tilhørende data. Se appendix seksjon 3 for mer omtale av noen Personlig-DND tjenester.

Vi bruker termen ”malesjøsk”(ondsinnet) anrop (malicious call). Se også appendix 2.4

Vi skal følge Alice og Carl i en tidsperidoe der Carl and Alice starter å ’deite’:

- Alice har 2 forskjellige telefon abonnement (ett for PSTN og ett for GSM).
- Carl har Personlig-DND tjeneste som omtalt i Appendix seksjon 3.1.
- Ved tidspunkt T1 ønsker Carl at Alice skal slippe gjennom hans Personlig-DND tjeneste.
- Seinere blir Alice truet av sin ex-kjæreste David. Hun finner ut at hun må bytte til nye telefonnummer.
- Ved tidspunkt T3 får Alice nye telefonnummer som er ’ikke-listet’ (men som ikke er strengt hemmelig). Dette betyr at hennes nye telefonnummer ikke er søkbare via 1881.no eller tilsvarende tjenester som gulesider/hvite-sider. Hun bestemmer seg for å beholde CLIP som tilleggstjeneste.
- En stund seinere forstår Alice at noen av hennes venner har lekket hennes mobilnummer til David og hun mottar truende anrop om natta.
- Ved tidspunkt T4 skifter hun telefonnummer igjen. Hun velger frotsatt ’ikke-listede’ telefonnummer. Denne gangen velger hun å abonnere på CLIR (Calling Line Identity Restriction, restriksjon på A-nummervisning) på både PSTN og GSM.

a) (6 %)

- **Deloppgave:** Explain briefly what actions Carl must do at/after time T1 in his private ’service management system’ on his answering machine. (This service management system is probably manual)
- **Deloppgave:** Explain briefly what actions Carl will need to take at/after time T3 on his answering machine in order for his personal-DND service to work properly (i.e. letting Alice through).
- **Deloppgave:** Explain what difficulties Carl enters into at Time T4. Is it enough to update with the new phone numbers of Alice? How can the service from Tele-Harmonia described in appendix section 3.2 serve Carl under these new conditions in a way different from the endpoint centric version.

b) (8 %)

Forklar kort hvordan PSTN/ISDN nettverksoperatøren kan kreere/bygge/realisere en variant av Personal-DND via IN. Du kan velge mellom å beskrive *enten det* globale funksjonelle planet (med BCSM og SIBer) eller detaljene på fysisk nivå (der du bruker SSP, SCP, IP etc) i det såkalte fysiske plan

- Du er bedt om å gi løsning på *bare ett* av disse to planene (etter egen valg). En skisse som viser de viktigste elementene og aksjonene er tilstrekkelig.
- Du er ikke forventet å kunne navn på SIB’er. Du kan bruke SIB’er kalt MESSAGE, CONNECT og RELEASE og evt. andre SIBer etter eget ønske.

NORSK TEKST HUSK AT ENGELSK TEKST ER AUTORITATIV (MASTER)

Oppgave 6 (12%) Tjenestemobilitet

Assume in this exercise that Alice and Bob subscribes to a standardized supplementary service SS1. In order to be a bit more concrete you may assume that SS1 is either:

- CLIR (Calling Line Identity Restriction) (subscribed to by Alice) or
- CFNR (Call Forwarding on No Reply) (subscribed to by Bob)

You may choose between CFNR and CLIR as you find most appropriate depending on your knowledge from the syllabus and your ability to find useful information in the appendixes. Note that CLIR in GSM is briefly explained at the end of Appendix section 2.2.

Assume also that Bob is currently roaming into a foreign domain and is assigned an MSC in this domain. Assume also that Alice is a GSM subscriber.

a) (4 %) Forklar rollen til HLR for tjenesteprofilhåndtering for tjenesten SS1 og vis hvordan HLR interagerer med andre entiteter for å realisere SS1.
   - Dvs. forklar hvordan SS1 (enten CLIR eller CFNR) er realisert i GSM, ved å bruke Alice og Bob som konkrete eksempler. Alice er hjemme (i GSM) der hun ringer til Bob.
   - Hint 1: Du kan se på Figure 1, (men forandre originerende side litt siden Alice nå er på GSM). Før dette anropet finner sted så har Bob registrert seg og du bør finne ut meldingsflyten (sekvensdiagrammet) for Bob når han registrerer seg på denne lokasjonen. Siden denne registreringen naturlig involverer HLRen, så finnes svaret på spørsmålet her.
   - Hint 2: Det kan være nyttig a rekapitulere hvordan HSS er involvert i en IMS registrering for håndtering av tjenesteprofildata og seleksjon av S-CSCF, *men prosedyrene er ikke eksakt de samme i GSM.*
   - Hint 3: Med mindre du husker hvordan CFNR in realisert i GSM, kan det hende at du vil se på hvordan CLIR is realisert i PSTN in appendix seksjon 2.2. Merk at that CLIR i GSM er kort forklart til slutt i Appendix seksjon 2.2.

De følgende definisjonene av tjenestemobilitet er gitt forskjellige steder i pensum.

Def.1 (fra kompendiet (LN)) **Tjenestemobilitet** (alternative betegnelser: programmobilitet, aktørmobilitet) tillater programvaremoduler/aktører (kode, objekter, prosesser) å bli overført fra en maskin til en annen.

Def. 2 (fra ETSI T TR 180001, Tispan, NGN release definition, oversatt til norsk): **Tjenestemobilitet**: mobilitet, relatert til en spesifikk tjeneste

- NB: Dvs. muligheten en bruker har til å bruke denne spesifikke tjenesten (som er abonnert på) uavhengig av brukerens lokasjon og uavhengig av terminalen som brukes.

b) (4 %) Tilbyr GSM tjenestemobilitet for CFNR og CLIR ifølge Def.2? Lag en fornuftig (kort) diskusjon i tillegg til å svare ja eller nei.
   - Hint: Du kan finne nyttig informasjon i et korrekt svar på 6.a)

c) (4 %) Bruker GSM tjenestemobilitet ifølge Def.1 for en tjeneste som CLIR (Calling Line Identity Restriction) eller CFNR (Call Forwarding on No Reply). Lag en fornuftig (kort) diskusjon i tillegg til å svare ja eller nei.
   - Hint: Du kan finne nyttig informasjon i et korrekt svar på 6.a)

Appendix

The appendix contains some material from the syllabus (LN, other reading material) as well as some material from new sources (not listed on the syllabus).

Some of the material given here is refered to in a specific exercise in the exam, while other material is given for your convienience.

All material listed here may be refered to in your answers.

1. GSM related information

The following figure is given in http://www.m-indya.com/gsm/gsmntaspects.htm (from the English syllabus). Note a small error in this figure.

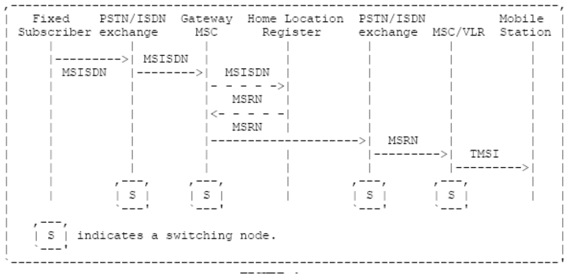
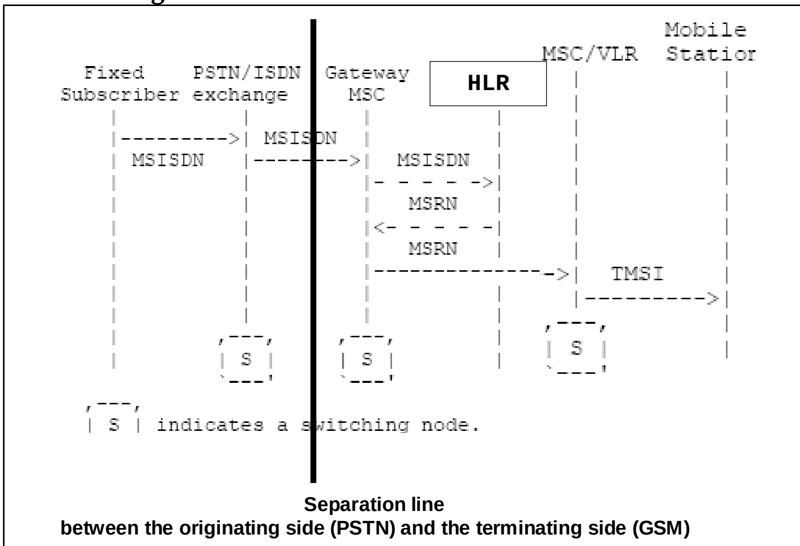


Figure 1 GSM call flow with location look up. Note a minor error

A **correct figure of call establishment in GSM** is as follows:



Legend:

MSRN: MobileStation Routing Number

TMSI: Temporal Mobile Station Identity

MSC: Mobile Switching Center

(MSC may also mean Message Sequence Chart)

Figure 2 GSM call flow)or MSC) from PSTN to GSM with location look up (correction). Fat line is added

The message names are not shown on these figures, instead it is the most important parameters related to addressing that are indicated. Also note that the details of BSC and BS are not shown on these figures.

Relating to PSTN material in Figure 3 (page 11), the name of the message between PSTN and GW-MSC will be Establish (with parameter diallednumber being the MSISDN of the B party as detailed in Figure 4). (The message Establish is also called ’call setup’ or similar).

2. PSTN related information

We will start by showing an ordinary call flow in PSTN (the names of the messages are 'abstract' (and not according to the real names in ISUP signaling). The names follow the material from Prof. Audestad (given here as Figure 5). It is recommended that you read 2.1 and 2.2 first and treat 2.3 as optional material.

2.1. PSTN (a simplified version)

Here we will show a high level view of how call setup work in PSTN. We have chosen to use 'abstract' names, and these names are in line with the names used by Audestad (see 2.3 for details).

In line with GSM we will use the term TE for the phone. We also find it useful to separate the phone from the human end user.

Note that in the general case there are more switches involved (and hence a longer chain of NNI interfaces), but this simple example shows the necessary details.

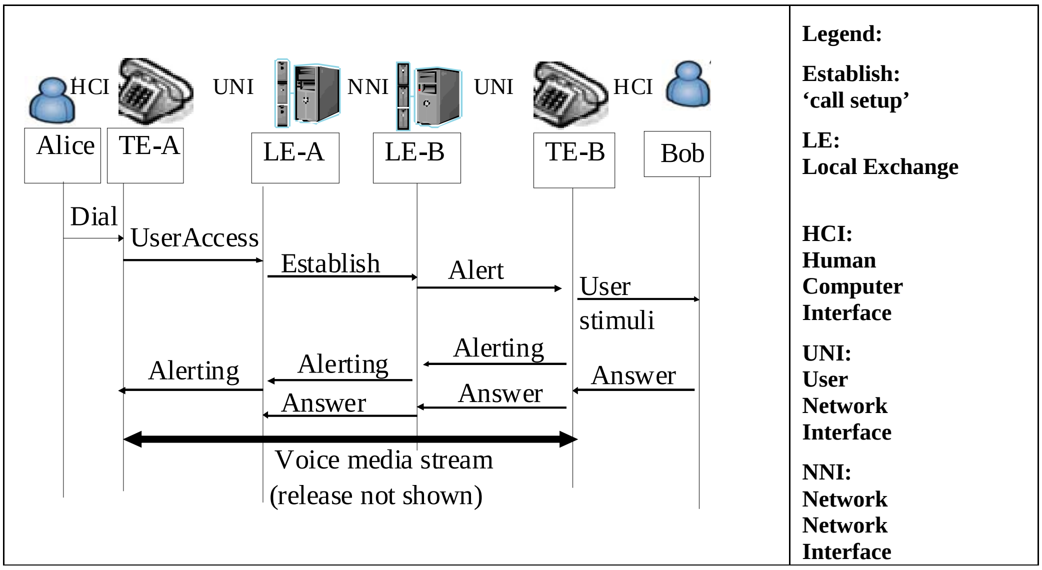


Figure 3 High level illustration of PSTN call setup (simplified case). Some parameters shown in Figure 4

This figure illustrates a call flow in PSTN and you should be able to compare this with the material given for GSM in Appendix section 1 and with SIP related material from the syllabus. In the latter case UserAccess, Establish and Alert will all correspond to INVITE.

Note that in PSTN there is no call and connection separation. The establishment of the connection (bearer) is performed by the Switches (LE-A and LE-B in this simplified example) as the call is progressing. (This is of course different from SIP) The details of the switching (connection establishment) is shown in 2.3, but these details can be ignored in this exam.

Important parameters in the message Establish are: diallednumber, CLId and the p-bit. The latter serves as a privacy indication.
(p-bit = 1 indicates that the privacy restriction apply as described in the next section 2.2)

2.2. CLIP and CLIR as two Supplementary Services (SS) in PSTN

CLId = Calling Line Identity (A-nummer på norsk)
CLIP = CLId Presentation (A-nummer-visning på norsk)
CLIR = CLId (presentation) Restriction
p-bit = privacy bit (indicating whether CLId shall be presented or not to the called party)

CLIR is a SS related to the calling party (A), and it allows calling party (A) to determine[2] that his/her own CLId shall not be presented towards the called party (B). Part of this SS execution takes place in the local exchange of the called party (B) as will be explained below.

CLIP is a SS related to the called party (in the old days B was charged for this SS). CLIP is relying on CLId to be passed in the network from LE of A towards the LE of B. LE of B is then responsible to check the p-bit and remove the CLId if so indicated.

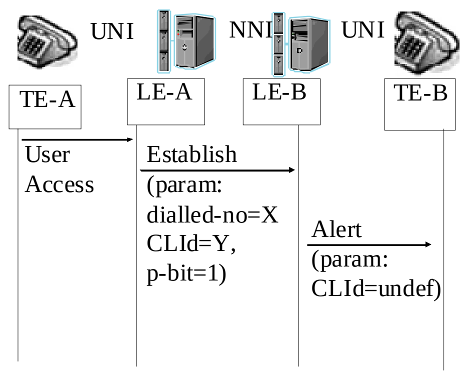
The behavior of LE-A on the originating side is partly described as follows:

```
IF <SS CLIR is active for A>
      THEN <include CLId in Establish and use p-bit=1>
      ELSE <include CLId in  Establish and use p-bit=0>;
```

The behavior of LE-B on the terminating side is partly described as follows:

```
IF <callerID is emergency number> THEN  <include CLId in Alert message>;
COMMENT overrule p-bit for emergency services ENDcomment;
ELSE IF p-bit = 1
      THEN <withhold CLId from Alert message>
      ELSE <include CLId in Alert message>;
```

In short: This means that CLId is sent within the network[3] also in case that calling party does not want the CLId to be presented to the enduser. CLId is then withheld by the LE before the alert message (also known as 'call setup' or similar) is sent over the UNI to the called party. In real life an exception apply for the emergency services 911/113 etc. and thus this is described above (as explained by the `COMMENT`).



Explanation:

Diallednumber = X,
where X = 22 22 33 44
which is the public phone number of Bob in Oslo

CLId (Calling line Identity) = Y,
where Y=73 73 74 75
which is the public phone number of Alice in Trondheim

Figure 4 A subscribes to CLIR and B subscribes to CLIP. Details of some parameters shown

The services CLIP and CLIR exist in GSM and then CLIR is realized as follows: In GSM B's MSC will handle the CLIR (p-bit) at the terminating side. *Note* that in GSM it is not really the CLId that is presented, but an identity representing the *user* (not the access *line*). However the term CLIR is still used in GSM.

---

[2] The selection may be done per call via e,g, *28# (or other more or less user friendly actions)

[3] Via the so-called NNI interface(/s)

2.3. Two figures from Prof. Audestad showing PSTN connections ++

These two figures are given here as supplementary information.

You should be able to manage with only the material from 2.1 and 2.2. However if you are already familiar with Audestad's material (from the syllabus) you will probably have interest in comparing these figures with the figures in 2.1 and 2.2.

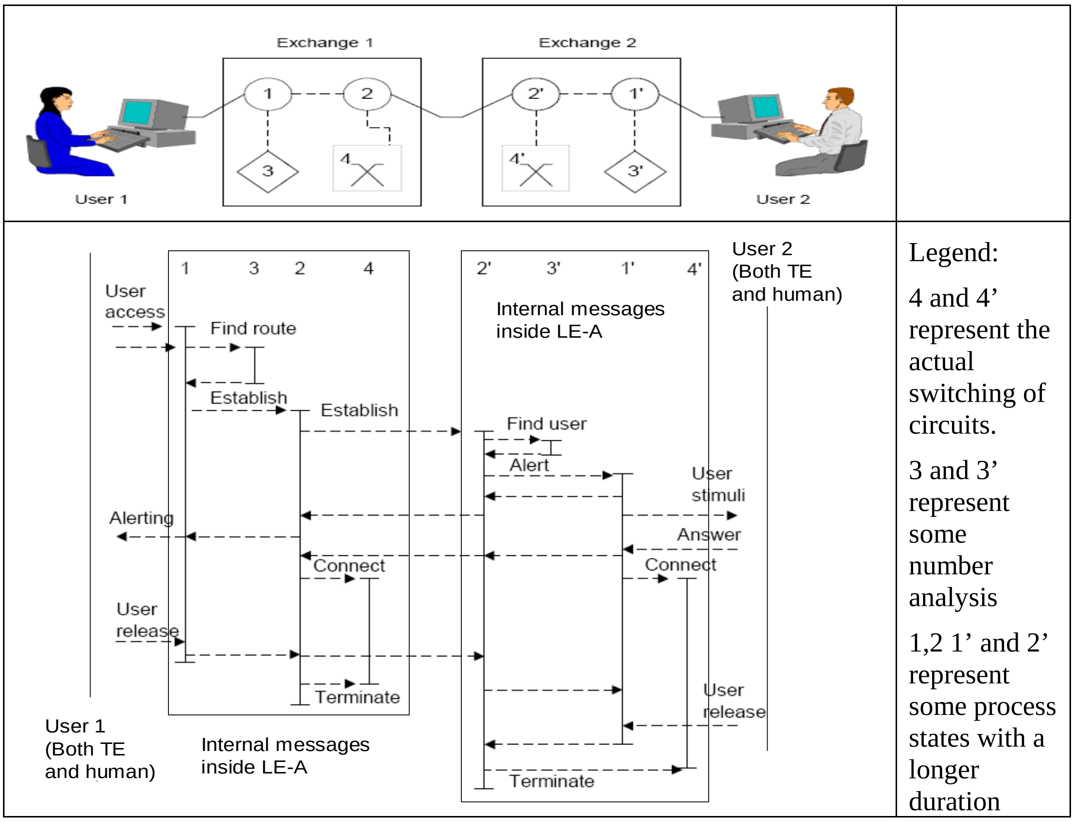


Figure 5 Two figures based on material from prof. Audestad.

Note that User 1 here is both TE and human user, while TE and human is separated in the figures in 2.1 and 2.2 (indicating UNI and HCI as two different interfaces). Because of this difference the message 'Alert' is shown as an internal message here while 'Alert' and 'user stimuli' are used on the UNI and HCI interfaces in Figure 3.

The details between UNI and TE and human user are better illustrated in Figure 3. While the inner parts of the LE is better shown in Figure 5.

In case you wonder why CLId is sometimes not shown on international calls even if you subscribe to CLIP you may see this footnote[4]

[4] There are several versions of NNI and on the international NNI interface the national operator might chose to withhold the CLId in case he does not trust that the other operator will respect the privacy bit (p-bit). See also Appendix section 4 is you wonder how IMS handles this issue.

2.4. MCID (Malicious[5] Call Identification) as defined by ETSI (Teddi database) (valid for PSTN)

”The MCID supplementary service is a [standarized] service that enables an incoming call to be identified and registered. The following call information is registered: - called party number; - calling party number; - local time and date of the invocation in the network serving the called user; and - as a service provider option: calling party sub-address (if provided by the calling user). The information is not available to the terminal equipment under the control of the called user nor the calling user. The information is stored at a location(s) under the control of the network operator. The MCID supplementary service can either be invoked during the active phase of the call, or after the active phase for a limited period […]”

A registration via MCID may be a first step leading to a criminal investigation at a later point.

3. “Personal-DND” Service descriptions

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.

It might be useful to read appendix 2.4 on MCID before reading this material.

3.1. 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[6] 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 <CLId on White_list> and personal_state = ’free’
      THEN <answer call on the phone>;
```

[5] The word *malicious* is derived from *malice* which means ’desire to harm others’ (according to Webster’s)

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

3.2. 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 offers a common black list of telephony sellers (to avoid phone spam).
- Via Parlay/OSA Call Control Tele-Harmonia has access to CLId of all incoming calls related to Carl. In this way Tele-Harmonia will be able to let Alice through and still keep telephony sellers using CLIR away from Carl's phone and home.[This will require some changes in the logic given in section 3.1]
- Tele-Harmonia is also integrating a presence server into their Personal-DND service. Hence Carl has now many options to choose from such as "occupied", "free", "lunch", "rather busy" etc. and the number of voice messages is increased accordingly. The message "stupid idiot" may of course be replaced with an immediate release of the call (if wanted)
- Tele-Harmonia also offers Carl an additional feature which we may call 'personal-malicious-call-barring' which is included in the personal-DND service. Each time Carl has received what he perceives as a 'malicious call' he may not only report it as a Malicious call (as explained in 2.4), but he might also *automatically add this phone number to his/her personal black-list*.
  This is done via a menu on the (programmable) phone.
  It can also be done via a webpage where all calls from the last X minutes are listed anonymously without CLId visible to Carl. In this way Tele-Harmonia respect the p-bit of the caller, and keep the trust of the network providers. (Even though the caller is perceived malicious s/he deserves some privacy.)
  It can also be done via an action such as dialing *27# while the other party is still active in call, or within Y sec. after the other party has hung up in a way more similar to the way it is realized in PSTN on dumb endpoints. (In the latter case Tele-Harmonia would need to get an indication of this interaction over Parlay/OSA).

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

It *might* be that the following material can be useful for you. It is up to each candidate to make use of this material.

This description is for IMS (not for SIP proper as defined by IETF).

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 UE (UE 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 UE.

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.

You should notice how the header Privacy has two values “None” and “id” is similar to the p-bit 0 and 1 as explained for PSTN in appendix 2.2

4.1. 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:

```
UE --> 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>) --> UE
```

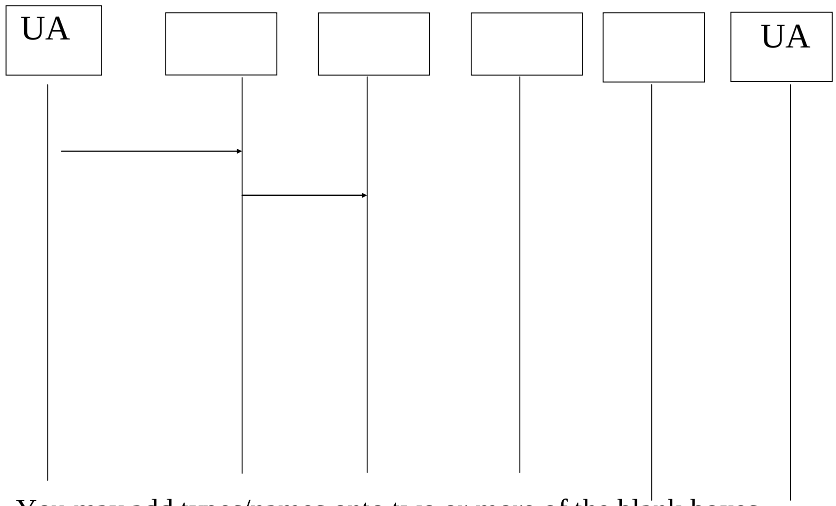
In this call flow it is assumed that the outcome at each check is ‘yes’. If last check results in ‘no’ the P-Asserted-Identity will be withheld.

Student number .... page ..... of ......

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

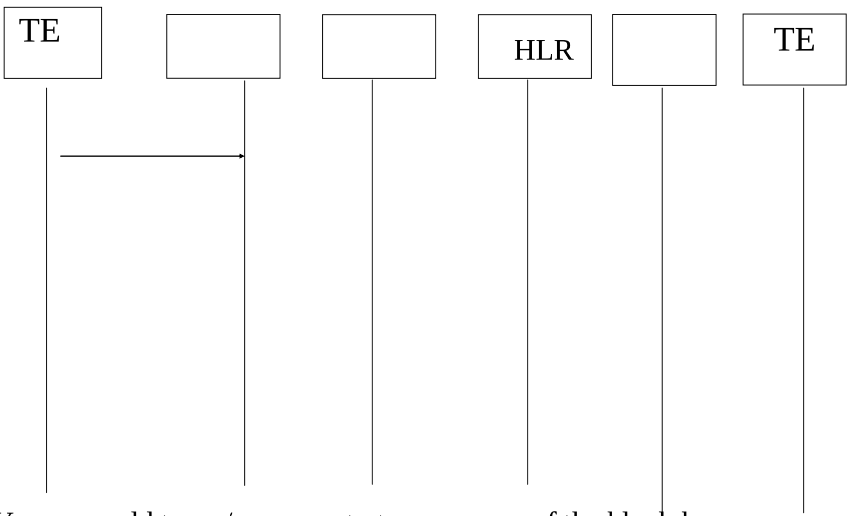
You *may* complete one or two of these MSC's as part of your answers

Answer to exercise ....



You may add types/names onto two or more of the blank boxes and use this to complete your answer on any SIP related exercise

Answer to exercise ....



You may add types/names onto two or more of the blank boxes and use this to complete your answer on any GSM-related exercise

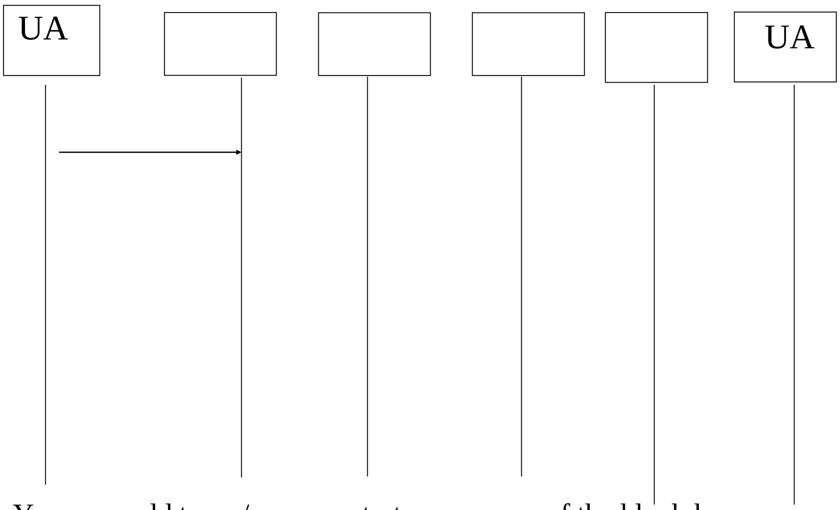
You may ignore some of the vertical lines or even add extra lines.

Student number .... page ..... of ......

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

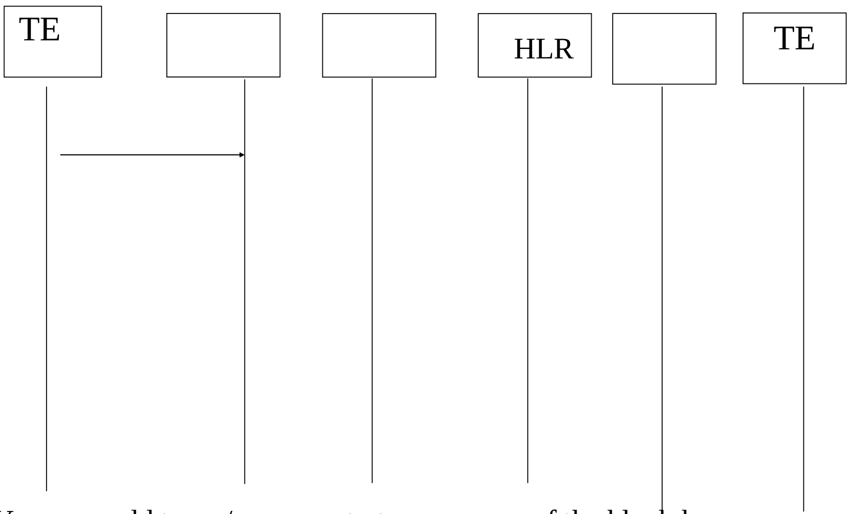
You *may* complete one or two of these MSC's as part of your answers

Answer to exercise ....



You may add types/names onto two or more of the blank boxes and use this to complete your answer on any SIP related exercise

Answer to exercise ....



You may add types/names onto two or more of the blank boxes and use this to complete your answer on any GSM-related exercise

You may ignore some of the vertical lines or even add extra lines.