S-72.423 Exercise 2. Solutions

Return your answer no later than on Tuesday 28.10.2003 at 16:00 into the course's P.O. box at the third floor of the E-wing.

Please, include the following information in your answers:
- Your name
- Your student number

It may be that you won't find answers to the questions straight from the lecture material. You may have to look for information from the textbooks and Internet. Good luck for information search!
To this exercise you may answer in English, Finnish or Swedish.

1. ISDN: List and describe 5 supplementary services other than mentioned in lecture notes.

(Lecture notes: CLIP / CLIR, Call forwarding / waiting / hold, Charging supplementary services;
CLIP = Calling line identification presentation, CLIR= Calling line identification restriction)

See Understanding Telecommunications 2, pages 249 – 254:
Some supplementary services for example:
Euro-ISDN1, Stage 1:
- MSN: multiple subscriber number allows up to eight numbers to be associated with one BRA.
- Terminal portability means a user can move his terminal from one jack to another during a call in progress.

Euro-ISDN2:
- UUS: user-to-user signalling allows a user to send and receive a limited amount of user information over the D-channel either before or during a call.
- The following variants are available:

UUS-1: message transfer during the set-up phase
UUS-2: message transfer when the B-subscriber receives ringing signals
UUS-3: message transfer during a call in progress
- SUB: subaddress allows a user of an access to be identified by adding digits to the subscriber number.
- CUG: closed user group is a group whose members are not free to make calls to addressees outside the group. Access to the group can also be barred.

etc….
Supplementary services in Euro-ISDN3 or later stages:
- UUS-2 and UUS-3, see above.

AOC: advice of charge allows the A-subscriber to be advised of usage-based charging
  - during call set-up (AOC-S);
  - during the call (AOC-D); or
  - at the end of the call (AOC-E).

Call waiting informs a user during a call in progress if another subscriber is trying to call him. The user can either accept or reject the new call. In Euro-ISDN, this service is available for BRA only.
CCBS: completion of call to busy subscriber automatically sends a callback signal to the B-subscriber when his line becomes free.
ETC…

2. Briefly explain the way of using and the differences of FISU, LSSU & MSU in CCS.

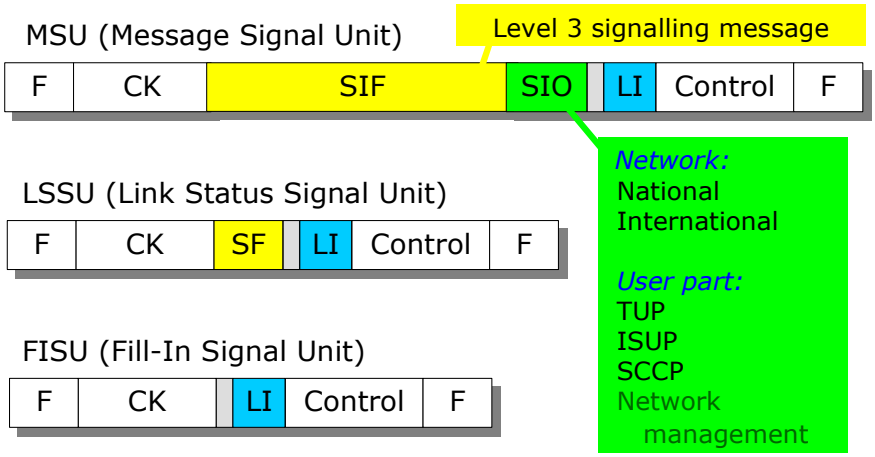
See also: http://www.cisco.com/univercd/cc/td/doc/product/tel_pswt/vco_prod/ss7_fund/ss7fun04.htm

Signaling information is passed over the signaling links in messages, which are called signal units. Signal units are continuously transmitted in both directions on any link that is in service.

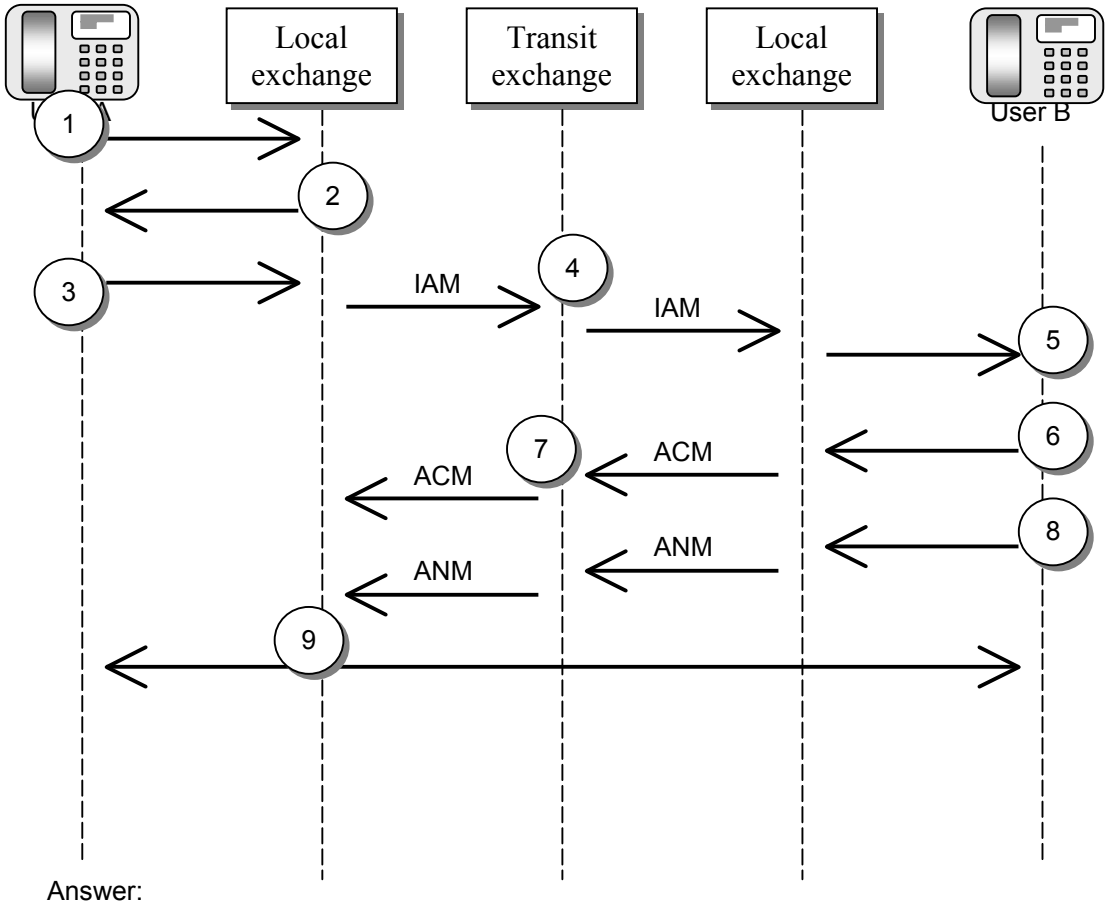
MSU (Message Signal Unit):
All signaling associated with call setup and teardown, database query and response, and SS7 management.
The received frame is MSU if LI > 2 (number of octets)

LSSU (Link Status Signal Unit):
Contains signalling messages for link supervision
The received frame is LSSU if LI = 1 or 2
LSSU's signals the initiation of link alignment, quality of received traffic, and status of processors at either end of the link.
LSSUs do not require any addressing information because they are only sent between signaling points.

FISU (Fill-In Signal Unit):
Can be used to monitor quality of signalling link
The received frame is FISU if LI = 0
FISUs do not carry any information; they simply occupy the link when there are no LSSUs or MSUs.
FISUs support the monitoring of link traffic because they undergo error checking. They can also be used to acknowledge the receipt of messages using backward sequence number (BSN) and backward indicator bit (BIB)



3. The figure below presents the signalling during the establishment of a circuit switched call from a subscriber behind an analogue line in the PSTN (User A) to a subscriber behind a digital ISDN line (User B). Your task is to shortly describe the numbered functions (in other words "what is going on") and why these functions are needed.



Answer:

1) User A picks up handset => off-hook detection at user A local exchange (sudden increase in current)
2) Exchange sends dial tone => user A hears that local exchange is "alive"
3) User A sends dialing information, for instance using DTMF (Dual Tone Multi-Frequency) signaling
4) Number analysis in the exchange => the exchange must know to which next exchange the IAM message should be sent
5) Q.931 SETUP message to user B => user B terminal starts alerting
6) Q.931 ALERTING message to user B local exchange
7) ISUP ACM message is sent in link-by-link fashion through the SS7 signaling network (no number analysis, only routing tables are used)
8) User B answers and Q.931 CONNECT message is sent to local exchange
9) After receiving the ISUP ANM message, the voice connection is "cut through" at user A local exchange. Usually charging of the call starts at this point (in Intelligent Network applications, other charging scenarios may exist).

4. **ATM**

Let us assume that compressed digital speech with a bit rate of 12 kbit/s is sent over an ATM link (cell bit rate = 150 Mbit/s) using AAL 1.
a) What is the packing delay at the sending side of the ATM link when each ATM cell is fully packed? Is there a similar delay at the receiving side of the ATM link?
b) What is the packing efficiency when the packing delay is not allowed to exceed 10 ms?
c) When using AAL 2, how many compressed speech signals can be multiplexed over one ATM virtual circuit? (We assume: Start field = 1 byte, CPS header = 3 bytes, CPS packet payload = 8 bytes)

Solution:

a) The payload of an ATM cell consists of a SAR-PDU containing 47 bytes of information plus a small header (1 byte). It takes 47 x 8 bits / 12000 bits/s = 31.3 ms to fill the ATM cell payload. Answer: packing delay = 31.3 ms.
b) Answer: no. The information can be reassembled immediately as soon as the ATM cell has arrived.
c) 10 ms corresponds to 12000 bits/s x 0.01 s = 120 bits = 15 bytes. Answer: The packing efficiency is 15 bytes / 47 bytes = 32 % (in other words 68 % of the ATM payload is empty).
d) In AAL2 layer the CID field allows up to 248 individual users within each AAL2-structure (see http://www.gdc.com/inotes/pdf/aal2tut.pdf, page 7).
Answer: 248

If we use several VC's let's calculate how many compressed speech signals can be sent over a 150 Mbit ATM link:
One could think that the answer is 150 Mbit/s / 12 kbit/s = 12500 compressed speech signals. However, this is not true since part of the ATM cell consists of headers. The real answer is 12500 x 47/53 x 8/(8+3) = 8061 compressed speech signals. Answer: 8061 compressed speech signals can be multiplexed over a single ATM virtual circuit.