Course 5 – 6
Signaling system 7 (SS7)

- Signaling System 7 (SS7) is an architecture for out-of-band signaling in support of the call-establishment, billing, routing and information exchange functions of the public switched network (PSTN).

(Signaling refers to the exchange of information between call components required to provide and maintain to service)

  - includes functions performed by a signaling system network and a protocol which controls this network.
  - it is characterized by high-speed packet data and out-of-band signaling.
  - applications supported by SS7 are:
    - PSTN.
    - ISDN.
    - Interaction with Network Databases (databases storing information related to the telecommunication network)and Service Control Points for service control
    - Mobile Services.
    - Administration and Maintenance operations of Networks.

- SS7 networks provide the following functionality:
    - Basic call setup, management, billing, and release.
    - Enhanced call features such as call waiting, call forwarding, calling party name/number display/restriction/rejection, and three-way calling.
    - Handling congestion and priorities.
    - Wireless services such as PCS, wireless roaming, and mobile subscriber authentication.
    - Local number portability (LNP).
    - Toll-free and toll services.
    - Exchange of database information between NEs (Network Elements).
    - Network management for efficient and secure communications.

- Out-of-band signaling – the signaling does not take place over the same path as the conversation; a separate digital channel is used for exchange of signaling information between switching nodes, channel called signaling link.
  - dedicated signaling links transmit information at rates of 56kbps or 64kbps.
  - ISDN D channel extends the concept of out-of-band signaling to the interface between the subscriber and the switch.
  - advantages of out-of-band signaling
    - it allows the transport of more data at higher speed (a data link of 56kbps can carry data much faster than the MF technique) - faster call setup.
    - it allows signaling at any time during the entire duration of the call.
    - it enables signaling with network elements having no direct trunk connections – more efficient use of the voice circuit, especially on international or long distance calls.
    - it ensures improved control over fraudulent network usage.
    - it offers support for more services.

- the simplest possible implementation of the out-of-band signaling is to allocate one of the paths between a pair of interconnected switches as signaling link – it is about an associated signaling to a group of trunks – see fig. 1.
- associated signaling works well as long the signaling is performed between switches connected by direct trunk connections – in this particular case associated signaling is simple and efficient.

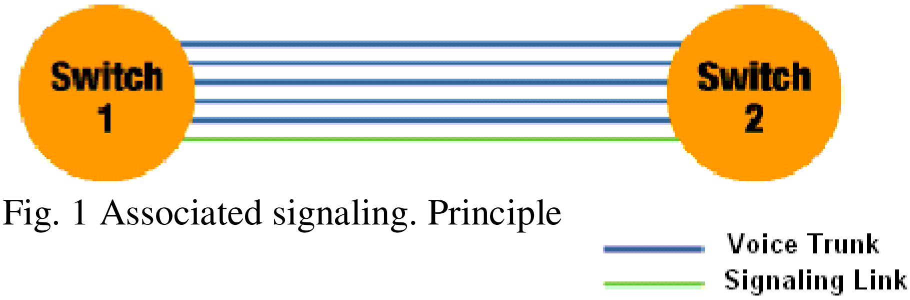

Fig. 1 Associated signaling. Principle

- differently to the associated signaling, the SS7 signaling system implements a signaling network that enables any node to exchange signaling with any other node – this is technologically not possible if associated signaling is used.
- both the associated and the SS7 type signaling are common channel signaling techniques.

- **The SS7 architecture –** the network includes the following three essential components, interconnected by signaling links:
  - Service Switching Points (SSPs) – SSPs are telephone switches (local offices or transit offices) equipped with SS7 capable software and terminating signaling links – they originate, terminate, or switch the call; an SSP sends signaling messages to other SSPs to setup, manage and release voice circuits, operations required to complete a call. An SSP may also send a query to a database (SCP) to determine how to route a call (for example toll-free calls). The SSP nodes are the points where the service users access the network, using an access protocol.
  - Signaling Transfer Points (STPs) – STPs are the packet switches of the SS7 network. They receive and route incoming signaling messages toward the proper destination. An STP routes each incoming message to an outgoing signaling link based on the routing information contained in the SS7 message. These equipments act as network hubs and improve the utilization of the of the SS7 network by eliminating the need for direct links between signaling points. The intermediate nodes, STPs, act as SS7 routers to provide multiple paths to a destination in order to handle failures within the network.

    ♦ STPs also offer specialized routing functions for toll-free 800 numbers, calling card numbers or mobile subscriber identification.

    ♦ STPs may also be used to screen the messages exchanged with other networks.

    ♦ STPs are usually deployed in redundant not co-located pairs – they work redundantly to perform the same function.

- Signaling (service) Control Points (SCPs) – SCPs are databases that provide information necessary for advanced call-processing capabilities. SCPs are usually deployed in matted pair configurations in separate physical locations, one of the SCP acting as a backup system. SCP executes network and data control functions such as billing or free-phone number translation.

- The availability of SS7 networks is critical for call processing – without exchange of signaling information between SSPs it is not possible to complete any interswitch call – for this reason, the SS7 network is built using a highly redundant architecture – each individual element have to meet imposed requirements for availability. Protocols were defined between the interconnected elements, protocols which provide error correction and retransmission capabilities to allow continuous services in the event of signaling point or link failures.
- Each signaling point in the SS7 network is uniquely identified by a numeric point code (PC). These codes are carried in the signaling messages exchanged between signaling points to identify the origination point (OPC) and destination point (DPC) of each message. Each signaling point uses a routing table to select the appropriate signaling path for each message.
- The general structure of a digital telephone network with SS7 signaling is presented in fig.2

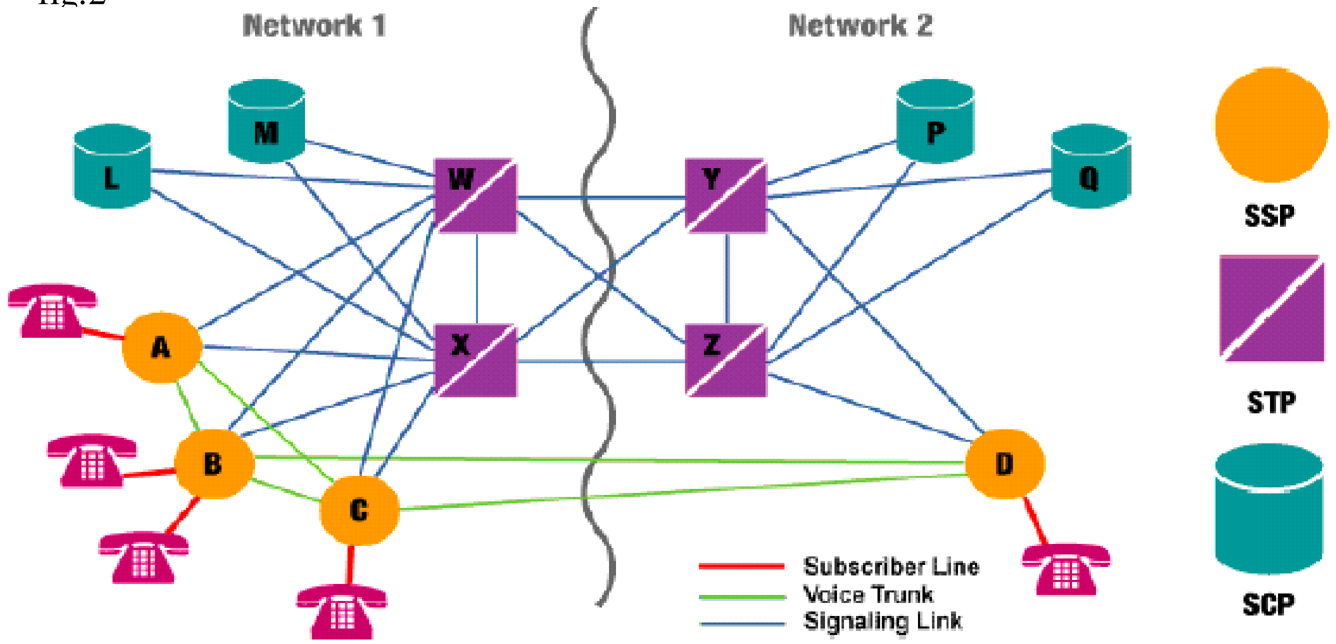

Fig. 2 The general structure a digital telephone network using SS7 signaling

- STPs W and X perform identical functions; they are redundant and together they are referred to as mated pair of STPs; similarly, STPs Y and Z form a mated pair.
- each SSP has two links (or set of links), one to each STP of a mated pair; because the STPs of the mated pair are redundant, messages sent over either link (to either STP) will be treated equivalently.
- the STPs of a mated pair are joined by a link (or set of links).
- two mated pairs of STPs are interconnected by four links (or set of links) – these links are referred to as a quad.

- SCPs are usually (not always) deployed in redundant pairs – they are not directly joined by links.
- signaling architectures such as the presented one, which provide indirect signaling paths between network elements, are referred to as providing quasi-associated signaling.

- **SS7 signaling link types**
- The SS7 network structure allows different types of connections between SPs. These links are logically organized by types (A to F), according to their use in the network;
  - all links are identical (56 or 64 kbps bidirectional data links) and support the same lower layers of the protocol.
  - one time slot of the T1 or E1 carriers can be used for transmission of the SS7 messages.
  - in fig. 3 are presented the signaling link types.

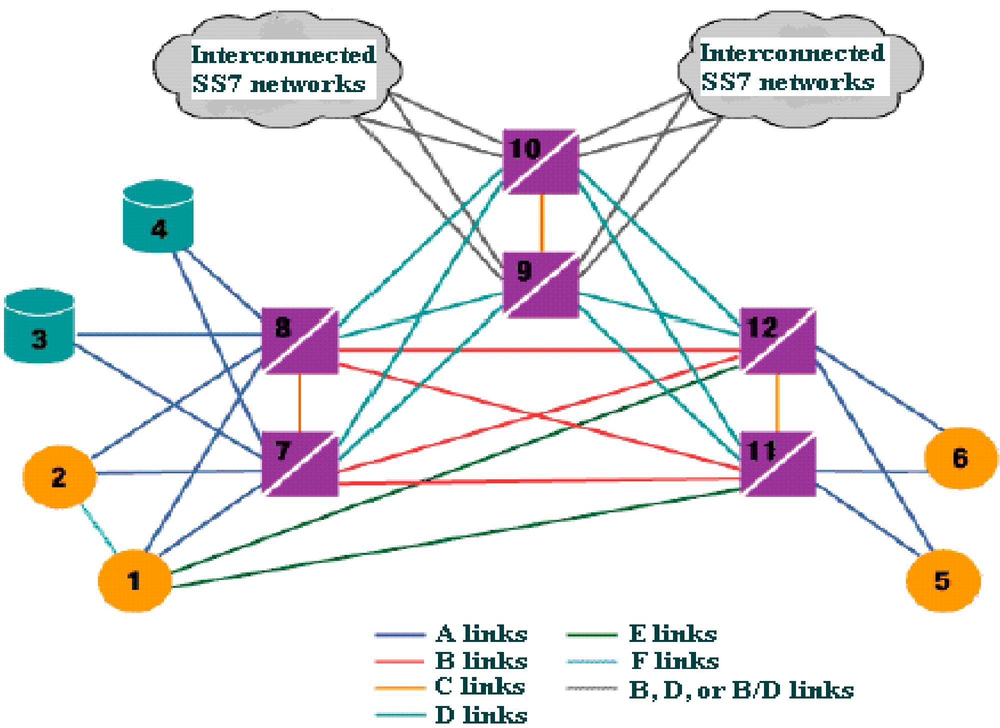

Fig. 3 Signaling link types used in SS7 networks

- **A link** – access link – connects a signaling end point or source point (for ex. SSPs or SCPs) to an STP; only messages originating from or destined to the signaling end points are transmitted on a A link.
- **B link** – bridge link – connects STPs; typically, quads of B links interconnect primary STPs of one network to primary STPs of another network; the function of these links is to carry signaling messages beyond their initial point of entry in the signaling network toward their destination; the interconnected pairs of STPs are on the same hierarchy levels.
- **C link –** cross link – connects STPs performing identical functions into a mated pair; they are used to enhance the reliability of the signaling network; a C link is used only when an STP has no other route available to a destination signaling point due to link failures; C links are not used between mated SCPs.

- **D link –** diagonal link – connects pairs of STPs at different hierarchical levels (for example a secondary -local or regional- STP pair to a primary -inter-network-gateway STP pair in a quad-link configuration); there is no clear hierarchy associated with a connection between networks and interconnecting links are referred to as either B, D or B/D links.
- **E link –** extended link – connects an SSP to an alternate STP to provide an alternate signaling path; E links are not provisioned usually, unless the benefit of a higher degree of reliability justifies the added expenses; these links provide backup connectivity to the SS7 network in the event that the home STPs cannot be reached via the A links.
- **F link** – fully associated link – directly connects two signaling end points (SSPs or SCPs); these links allows associated signaling only; these links are not usually deployed in networks with STPs, because they bypass the security features provided by the STPs.

- **Basic call setup based on SS7 signaling** – see figure 4 and 5 for a simple example

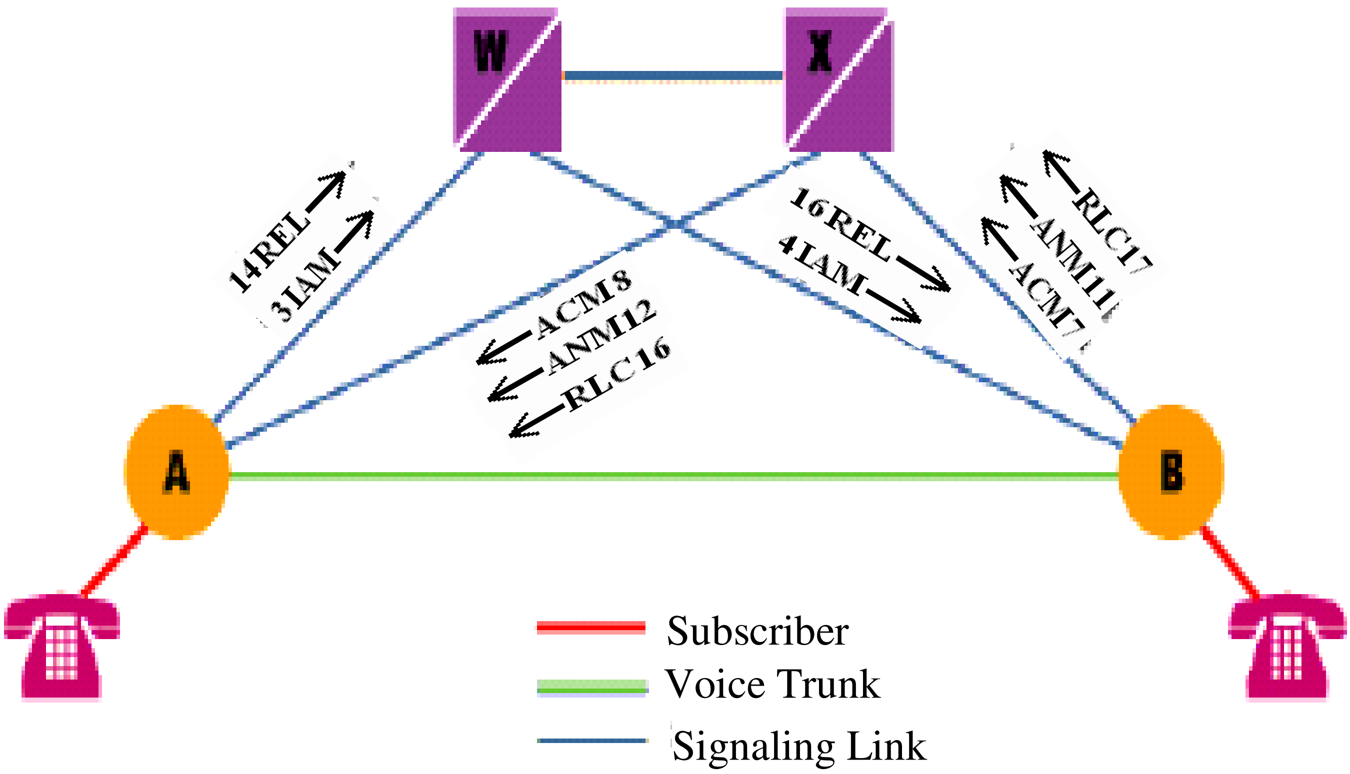

Fig. 4 Basic call setup based on SS7 signaling

- A subscriber of switch A places a call to a subscriber of switch B – the steps of call establishment, maintenance and release are the following:

1. Switch A analyzes the dialed digits and determines that the call is intended to switch B.
2. Switch A selects an idle trunk between switches A and B and formulates an initial address message (IAM) – the basic message necessary to initiate a call; the IAM message is addressed to switch B.
3. Switch A accesses one of its access links (for ex. A-W) and transmits the message over the link for routing to switch B.
4. STP W receives the message, inspects its routing label, and determines that it is to be routed to switch B; it transmits the message on link B-W.

5. Switch B receives the message, analyzes it and determines that it serves the called number and that this number is idle.
6. Switch B formulates an address complete message (ACM), which indicates that the IAM message has reached the proper destination; the message identifies the recipient switch (A), the sending switch (B), and the selected trunk.
7. Switch B accesses one of its A links (B-X) and transmits the ACM over the link for routing to switch A and at the same time, it completes the call path in the backward direction, sends the ring back signal over the seized trunk toward switch A and rings the line of the called subscriber.
8. STP X receives the message, inspects its routing label and determines that it is to be routed to switch A; it transmit the message on link A-X.
9. On receiving the ACM message, switch A connects the calling subscriber line to the selected trunk in the backward direction – the caller can hear the ring back signal sent by switch B.
10. When the called subscriber picks up the phone, switch B formulates an answer message (ANM), identifying the intended recipient switch (A), the sending switch (B), and the selected trunk.
11. Switch B selects the same A link it used to transmit the ACM message (link B-X) and sends the ANM message; in this moment the trunk must be connected to the called line in both directions (to allow conversation).
12. STP X recognizes that the ANM message is addressed to switch A and forwards it over link A-X.
13. Switch A ensures that the calling subscriber is connected to the outgoing trunk (in both directions) and that conversation can take place.
14. If the calling subscriber hangs up first (following the conversation), switch A will generate a release message (REL) addressed to switch B, identifying the trunk associated with the call; it sends the message on link A-W.
15. STP W receives the REL message, determines that it is addressed to switch B, and forwards it using link W-B.
16. Switch B receives the REL message, disconnects the trunk from the subscriber line, returns the trunk to idle state, generates a release complete message (RLC) addressed back to switch A, and transmits it on link B-X; the RLC identifies the trunk used to carry the call.
17. STP X receives the RLC message, determines that it is addressed to switch A, and forwards it over link A-X.
18. On receiving the RLC message, switch A places the identified trunk in idle state.

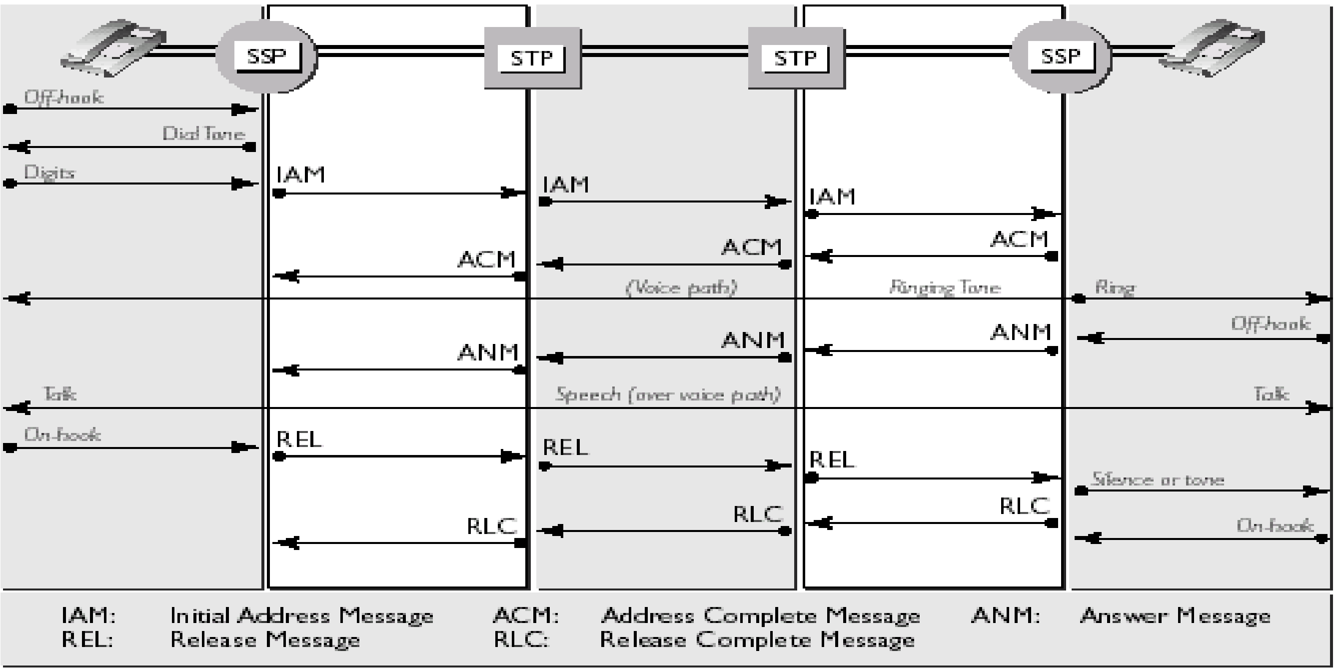

Fig. 5 Basic call setup based on SS7 signaling – alternative representation

- **Basic database query based on SS7 signaling – see figure 6**

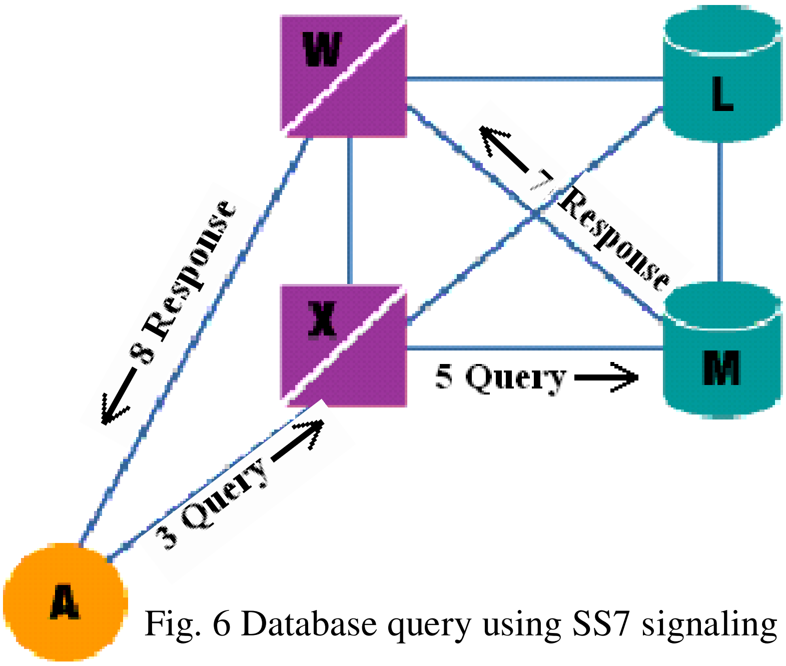

Fig. 6 Database query using SS7 signaling

- a possible example is related to calls addressed to toll-free 800 or 888 number; these numbers are virtual telephone numbers, not assigned to a subscriber line.
- when a subscriber dials an 800 number the switch must seek further instructions from a database – the database provides either a real phone number to which the call should be directed, or it will identify another network (a long-distance carrier) to which the call should be routed for every processing – the response from the database could be the same for every call or it can vary based on the calling number, the time of the day, the day of the week, or several of other factors.
- a simple example, related to a 800 number call is presented in fig.6.

1. A subscriber served by switch A dials a 800 number.

2. When the subscriber has finished dialing, switch A recognizes that this is an 800 call and that it requires assistance to handle it.
3. Switch A formulates an 800 query message including the calling and called number and forwards it to one of its STPs (for example STP X) over its access link (for ex. link A-X).
4. STP X determines that the received query is an 800 query and selects a database suitable to respond to the query (for ex. database or SCP M).
5. STP X forwards the query to SCP M over the appropriate access link (M-X); SCP M receives the query, extracts the passed information, and based on its stored records selects either a real phone number or a network or both to which the call should be routed.
6. SCP M formulates a response message with the information necessary to properly process the call, addresses it to switch A, access an STP and an access link to use (for example M-W) and routes the response.
7. STP W receives the response message, recognizes that it is addressed to switch A, and routes it to A over the A-W link.
8. Switch A receives the response and uses the information to determine where the call should be routed; it seizes a trunk to that destination, generates an IAM message, and proceeds to set up the call – see the previous example.

- **<u>Layers of the SS7 protocol</u>**
- the SS7 network is composed of an interconnected set of network elements that are used to exchange messages in support of telecommunications functions – the SS7 protocol is designed both to facilitate these functions and to ensure the maintenance of the network over which the mentioned functions are provided.
- the SS7 protocol is divided into several functional layers - initially the SS7 architecture was designed for circuit-related telephony, but it evolved as new requirements have emerged and now it allows also the transfer of non-circuit related information.
- the layers of the SS7 protocols are presented in fig. 7

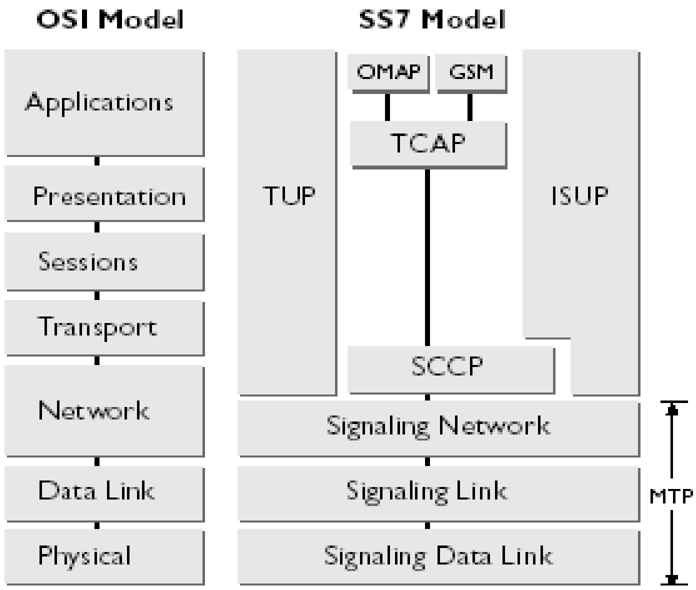

Fig. 7 The layers of SS7 protocols and comparison with the OSI layers
OSI – Open System Interconnection

- **Message Transfer Part (MTP)**
  - **Signaling Data Link** functions: define the physical, electrical, and functional characteristics of the digital signaling link;
    - defined physical interfaces include: DS1 (one slot of the T1 frame having a bit rate of 1.544Mbps), E1 (one time slot of the E1 frame having a bit rate of 2.048Mbps, usually the time slot 16), V.35 (synchronous serial interface at 64kbps or 56kbps), DS0 (64kbps), DS0A (56kbps) – 56kbps is the more common implementation.
  - **Signaling Link** functions: define the functions and procedures to ensure that messages are reliably transmitted across a signaling link;
    - the mentioned functions implement flow control, message sequence validation, and error checking – when an error occurs on a signaling link, the messages are retransmitted.
  - **Signaling Network** functions: define those transport functions and procedures that are common to and independent of individual signaling links;
    - provides node addressing and message routing between signaling points in the SS7 network.
    - re-routes traffic away from failed links and signaling points, and control traffic when congestion occurs.
    - ensures that the messages can be delivered between signaling points across the SS7 network regardless of whether they are directly connected.
- **Signaling Connection Control Part (SCCP)** – provides additional functions to the MTP, to support connectionless and connection-oriented network services and Global Title Translation (GTT) – it is used as an end to end transport layer.
  - SCCP provides subsystem numbers to allow messages to be addressed to specific applications or subsystems at specified signaling points.
  - GTT adds the ability to perform incremental routing and frees the originating signaling point of having to know every possible destination; a global title is an address (an 800 number, calling card number, or mobile subscriber identification number) which is translated by the SCCP into a destination point code and subsystem number – such a subsystem number uniquely identifies an application at the destination signaling point.
  - SCCP is used as transport layer for TCAP – based services.
- **Telephone User Part (TUP)** – defines the international call control signaling functions for basic call setup and release – represents an earlier implementation of SS7 and does not allow data type applications.
- **ISDN User Part (ISUP)** – defines the protocols used to setup, manage, and release trunk circuits that carry voice and data between SSPs (located in PSTN) – is used for both ISDN and non-ISDN calls – calls that originate and terminate at the same switch do not use ISUP signaling.
- **Transaction Capabilities (TC)** – provides the means to establish non-circuit related communications between two SPs
  - **Transaction Capabilities Application Part (TCAP)** – supports the exchange of non-circuit related data between applications across the SS7 network using the

SCCP connectionless service as a transport layer. It defines the messages and protocols used to communicate between applications in nodes.

- ♦ queries and responses sent between SSPs and SCPs are carried in TCAP messages.
- ♦ in mobile networks, TCAP carries the Mobile Application Part (MAP) messages sent between mobile switches and databases to support user authentication, equipment identification, and roaming.

- o **Operations, Maintenance and Administration Part (OMAP)**
  - OMAP defines messages and protocols used in the administration of the SS7 networks – services provided by OMAP may be used to verify network routing databases and diagnose link problems – OMAP includes messages that use both the MTP and SCCP for routing.

- **Packet transmission over the Signaling Links**
- signaling information is transmitted over the signaling link in messages, which are called signal units (SUs)
- there are three types of signal units defined in the SS7 protocol:
  - Fill-In Signal Units (FISUs)
  - Link Status Signal Units (LSSUs)
  - Message Signal Units (MSUs)
- SUs are transmitted continuously in both directions on any link that is in service; a signaling point that does not have messages or status signals to transmit will send FISUs over the link – FISUs facilitate link transmission monitoring and acknowledgement of other SUs; all messages are composed of bytes.
  - o Fill-In Signal Units (FISU) – operate when there is no other SU traffic present; FISU are transmitted continuously on a signaling link in both directions to keep the link alive and aligned; these units carry CRC and in this way the link quality is continually checked by the SPs at each end of the link – see fig. 8 for the structure of the FISU messages.
  - o Link Status Signal Units (LSSU) – are used to exchange link status information between the SPs at each end of the link; they are used to control link alignment and to give status of a signaling point to a remote signaling point - see fig. 9 for the structure of the LSSU messages. Before an SS7 link is able to convey information from the higher layers, the layer 2 entities at each end of the link follow a handshaking procedure known as the proving period, lasting for 0.5 to 8.2 seconds (depending on the availability of routes served by the link in question). During this time, Link Status Signal Units (LSSU) are exchanged between the layer 2 entities of the protocol, the number of the errors received during this time being monitored. If the detected number of errors is less than a threshold, the link enters the in service state, and may carry MSU packets containing information from the upper layers. The layer 2 entities also monitor the state of the link and communicate this link state information to their peers in LSSU messages. These messages are transmitted, for example, when links become congested or are taken out of service.
  - o Message Signal Units (MSU) – are the containers that carry TUP, ISUP, and SCCP protocol messages within the information field; they carry all call control, database query and response, network management, and network maintenance data; there are additional specialized functions pertaining to mobile cellular applications; these units have a routing label that allows an originating signaling point to send information to a

destination signaling point across the network - see fig. 10 for the structure of the MSSU messages.

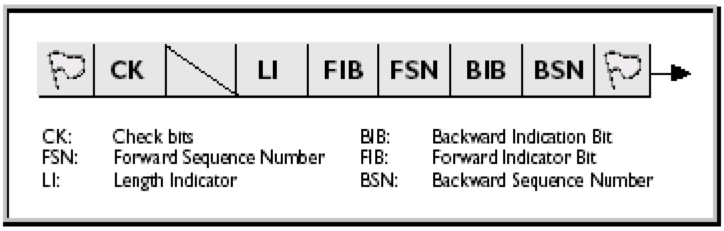

Fig. 8 FISU message structure

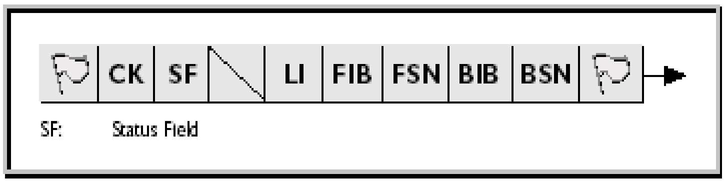

Fig. 9 LSSU message structure

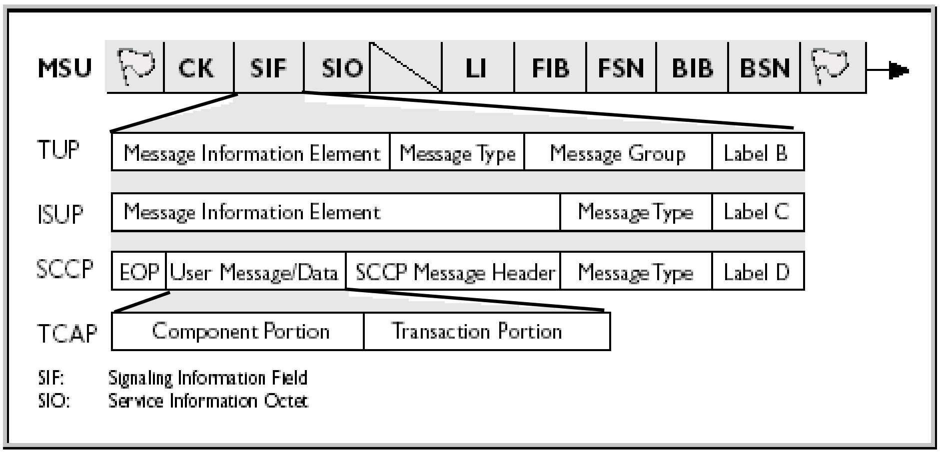

Fig. 10 MSU message structure

- Flag (0 1 1 1 1 1 1 0) – indicates the beginning of a new signal unit and the end of the previous signal unit; bit manipulation techniques are used to ensure that this pattern does not occur within the message transmitted on the link – the SU is reconstructed once it has been taken off the link and any bit manipulation is reversed - a possible bit manipulation consists in insertion of a zero after any sequence of five ones; any occurrence of the flag on the link indicates the end of one SU and beginning of another – in theory two flags could be placed between SUs, one to mark the end of the current message and one to mark the start of the next message, but in practice just one flag is used.
- BSN (Backward Sequence Number) – acknowledges the receipt of signal units by the remote signaling point; contains the sequence number of the signal unit being acknowledged; every single message needs to be acknowledged by means of BSN.
- BIB (Backward Indicator Bit) is used for error recovery and indicates a negative acknowledgement by the remote signaling point when inverted.
- FSN (Forward Sequence Number) – contains the sequence number of the signal unit.
- FIB (Forward Indicator Bit) is used in error recovery; when a negative acknowledgement is received all forward messages are retransmitted beginning with the corrupted message – in these messages FIB is inverted.
  - BSN+BIB and FSN+FIB are used to confirm the receipt of SUs and to ensure that they are received in the order in which they are transmitted; they are used also to provide flow control; the sequence numbers of the transmitted messages are stored until these messages are acknowledged by the receiving signaling point.

- seven bits are allocated to the forward sequence number and in this way is possible to store 128 distinct values – a signaling point is restricted to sending 128 unacknowledged SUs before it must await an acknowledged SU – by acknowledging an SU, the receiving node frees that SUs sequence number at the transmitting point, making it available for a new outgoing SU.

o Remark: There are two error control methods used on SS7 links: the basic method, when a message is only retransmitted on receipt of a negative acknowledgement, method which uses the BSN+BIB, FSN+FIB and CK fields, and Preventative Cyclic Retransmission (PCR), when a message is repeatedly sent when the upper layers have no information to be sent to the network. PCR is generally only used over transmission paths where the transmission delay is large, such as satellite links.

- ♦ SIO (Service Information Octet) – contains the subservice field and service indicator – see the presentation of the MTP3 level.
- ♦ SIF (Signaling Information Field) – contains the routing label and signaling information, i.e. SCCP, TCAP and ISUP message data – see the presentation of the level 4; LSSUs and FISUs have no routing label and SIO as they are sent between two directly connected signaling points.
- ♦ Length Indicator (LI) – indicates the number of octets between itself and the checksum; it serves both as a check on the integrity of the SU and as a means of discriminating between different types of SUs – FISUs have a length indicator of 0, LSSUs have a length indicator of 1 or 2 (in general LSSUs have a LI=1), and MSUs have a length indicator greater than 2; only 6 of the 8 bits of the length indicator field are used to store the mentioned length – thus the largest value that can be accommodated in the length indicator is 63 – MSUs with more than 63 octets after the LI field use a value of 63.
- ♦ CK (Check bits) – is a CRC value used to detect transmission errors.

- **MTP Layer 3 (MTP3)**
  - ♦ Layer 3 provides message routing and failure handling capabilities for the message transport. Each SS7 node, which could be a classic switch or a node containing 800 number translation records, is uniquely identified within a network using an SS7 address called a *Point Code*. European networks use 14 bit point codes, North American 24 bit point codes.
    - individual signaling points belongs to a cluster of signaling points and within that cluster, each signaling point has a member number; similarly, a cluster is part of a network – the routing addresses are three level numbers defined by the network, cluster and member number – each of these numbers is an 8 bit number; the complete address number is known as the point code of the signaling point, code which uniquely identifies a signaling point.
  - ♦ A single SS7 link is able to carry traffic for thousands of circuits; depending on traffic a single SS7 link is normally engineered to control 1000 to 2000 circuits. Failure of this single link would disable all of the circuits that are controlled, hence for resilience and also to increase traffic capacity, more than one signaling channel is normally provisioned between any two nodes communicating using SS7. The collection of *signaling links* between two adjacent nodes is known as a *link set*, each link set can contain up to 16 signaling links – see fig. 11

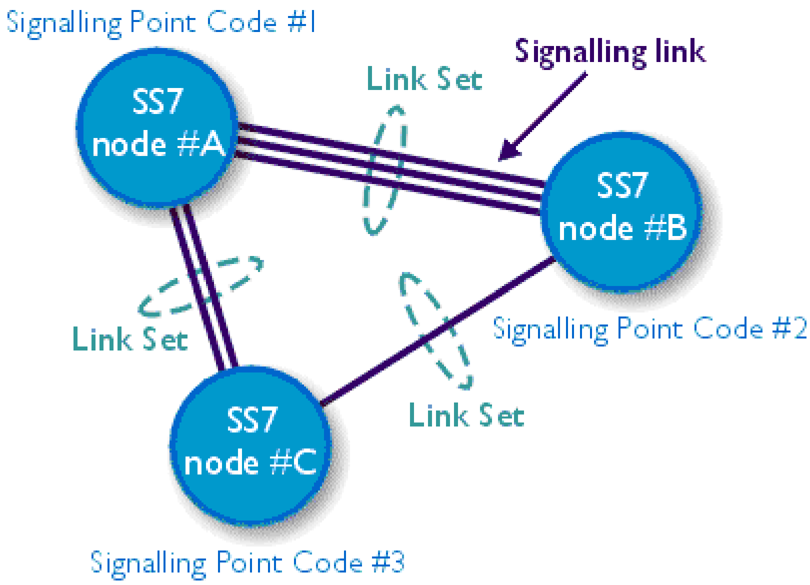

Fig. 11 Example of SS7 network and signaling link sets

- MTP3 adds information into the Signaling Information Field (SIF) of the MSU packets. This includes a Destination Point Code (DPC) identifying the destination for a message, an Originating Point Code (OPC) identifying the originator of a message and a Signaling Link Selection (sls) value used by MTP3 to load share messages between links in a link set – see fig. 12 and 14.

Fig. 12 Structure of the MTP3 header

- The MTP automatically load shares between the links within a link set, and re-routes traffic from failed links to a working link within the same link set on detection of failure. MTP3 layer also attempts to automatically restore failed links and returns traffic to a recovered link, these two procedures being termed *Changeover* and *Changeback*. MTP3 is also able to load share between two link sets that serve the same destination, by the use of intermediate nodes, the link sets in discussion being contained within a *route set*.

- MTP3 provides a reliable message transport service to the higher layer protocols, which use MTP as a message transport service – the protocols located at higher layer are generically called *User Parts*. In order to deliver a received message to the correct user part, MTP3 examines the *Service Indicator* (SI) which forms part of the *Service Information Octet* (SIO) – see fig. 13 and 14

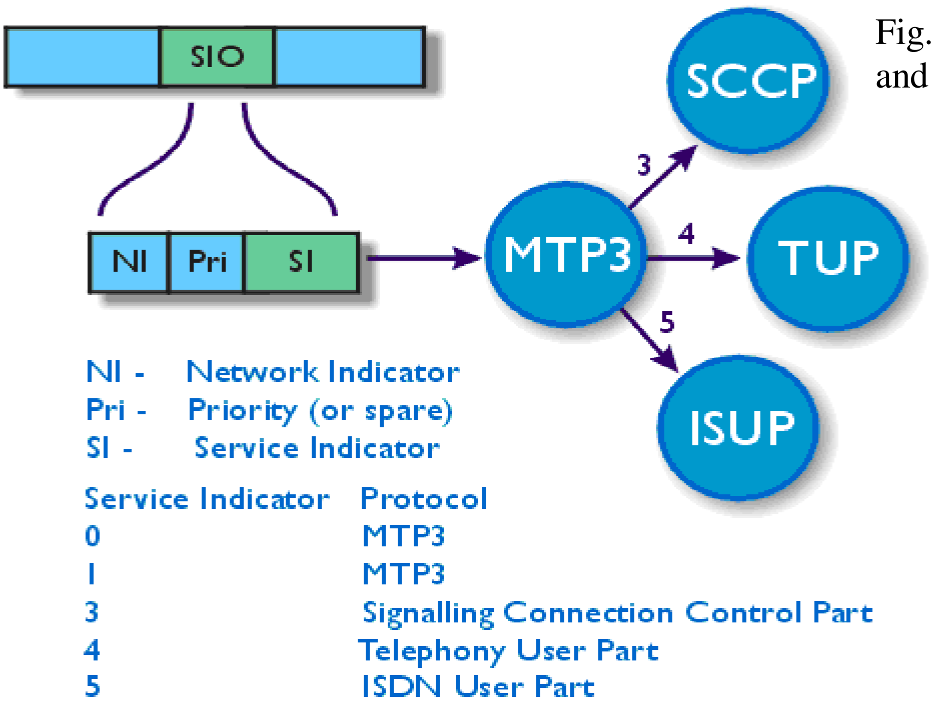

Fig. 13 Structure of the SIO octet and MTP3 message distribution

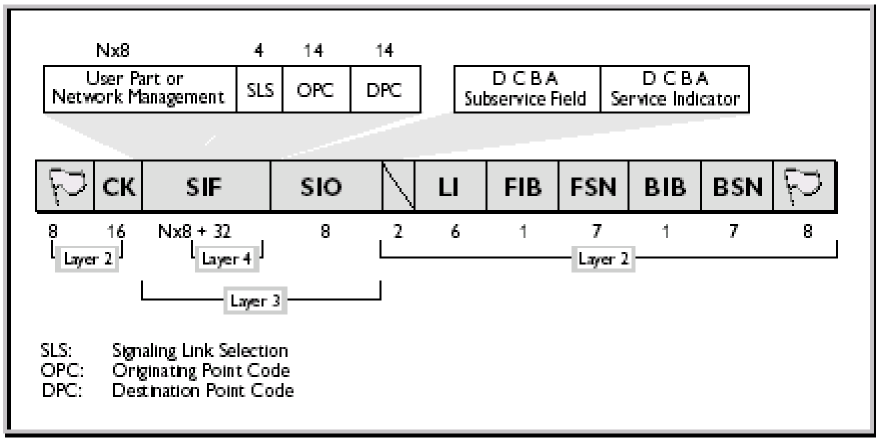

Fig. 14 SIF and SIO field structures

- ♦ SIO (Service Information Octet) – contains the subservice field and service indicator
  - ▪ Subservice Field contains the network indicator (national or international) and the message priority – low priority messages may be discarded during periods of congestion; signaling link test messages have a higher priority than call setup messages.
  - ▪ Service Indicator specifies the MTP user, that are TUP, ISUP, SCCP or other – see fig. 13.

- **Layer 4 protocols**
  - ♦ The layer 4 protocols define the contents of the messages and sequences of messages sent to MTP3 in order to control network resources, such as circuits and databases.

- **Telephony User Part (TUP)**
  - ♦ Telephony User Part (TUP) provides conventional PSTN telephony services across the SS7 network. TUP was the first of the standardized layer 4 protocols and did not provision for ISDN services.

♦ The message (signal) sequence used for establishment – control – release of a normal telephone call is similar with the message sequence characteristic to the ISUP protocol.

- **ISUP – ISDN user part** – defines the protocol and procedures used to set up, manage, and release trunk circuits that carry voice and data calls over the public switched telephone network – it is used for both ISDN and non-ISDN calls; calls that originate and terminate at the same switch do not use ISUP signaling. Offers a greater variety of messages and parameters in order to implement ISDN type services within the network.
- ISUP and TUP both provide additional messaging and management for circuit state control. It is possible to reset a circuit or a group of circuits. Circuits are normally reset on system initialization or following a failure. Similar procedures exist for blocking circuits, making a circuit temporarily unavailable for calls. Any call received for a blocked circuit is automatically rejected. Blocking may either wait for any active calls to terminate before taking effect, this is know as either maintenance blocking or blocking without release and is used prior to maintenance action. Hardware blocking or blocking with release is used on detection of failure of physical equipment or trunks that disrupt a voice circuit, and causes instant release of associated circuits and calls.
- the ISUP message structure is presented in fig. 15.

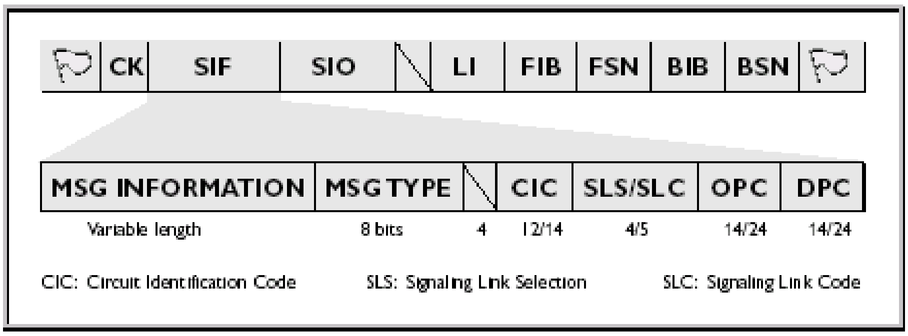

Fig. 15 ISUP message structure

- the SIF field contains the routing labels: DPC and OPC.
- the CIC code identifies the trunk circuit reserved by the originating switch to carry the call; a trunk is uniquely identified by the CIC code and point codes of the interconnected SSPs.
- the MSGTYPE field specifies the type of the message, that are: IAM, ACM, ANM, REL, and RLC – see fig. 4 and fig. 5 and the related explanations; this field defines the content of the message field – MSG INFORMATION
- short presentation of the messages:

♦ IAM – Initial Address Message – this contains call setup information and is sent when the switch needs to complete the circuit between the calling party and the called party.

- the IAM contains the called number in the mandatory variable part and may contain the calling party name and number in the optional part – it is about the MSG INFORMATION field.

♦ ACM – Address Complete Message – indicates that the called party is available and a remote end of a trunk circuit has been reserved;

- the originating switch responds to an ACM message by connecting the calling party's line to the trunk – the voice circuit is completed from the calling party to the called party – the ringing signal is transmitted to the called station and the ring back signal is transmitted to the calling party.

- ♦ ANM – Answer Message – when the called party answers, the destination switch terminates the ringing and ring back signals and sends the ANM message to the originating switch;
    - the originating switch initiates billing after verifying that the calling party's line is connected to reserved trunk.
- ♦ REL – Release Message – indicates that the circuit was released and indicates the release cause; a REL message is sent when either the calling or called party "hangs up" the call; a REL is also sent in the backward direction if the called party line is busy or if no channel is available.
- ♦ RLC – Release Complete Message – acknowledges the reception of REL from the remote end of the trunk circuit and ends the call and billing cycle.

- **Signaling Connection Control Part (SCCP)**
- The Signaling Connection Control Part (SCCP) enhances the routing and addressing capabilities of MTP to enable the addressing of individual processing components or *sub-systems* at each signaling point.
- Basic SCCP addressing routes messages through the network using a sub-system number and point code to identify a destination. Each sub-system could be a number translation database; an SS7 point code can potentially have many sub-systems attached.
- SCCP provides four classes of services, numbered 0 to 3 – see table 1
- The most commonly used class of SCCP is 0 and 1, used by TCAP and higher layers in the control of mobile/wireless and intelligent networks. Class 2 and 3 can be used by mobile networks in the communication between radio base-stations and the base-station controller.

Class	Properties
0	Connectionless, data is sent to a destination without negotiation of a session
1	Connectionless with sequence control. Messages are guaranteed to be delivered to a destination in sequence.
2	Connection oriented. A session (SCCP connection) is negotiated prior to the exchange of data.
3	Connection orientated with flow control.

Table 1 SCCP service classes

- SCCP maintains a state of every sub-system that it is aware of; sub-systems may be on-line (*Allowed*) or off-line (*Prohibited*). A message or connection session can only be delivered to an allowed destination sub-system.
- The basic message of connectionless SCCP is the SCCP UNITDATA (also called UDT). When SCCP detects that a destination for a message is prohibited, the UDT can either be discarded or returned to the originator as a UNITDATA SERVICE (UDTS), if a return option parameter is set in the quality of service field of the message.
- In order to track and report the status of sub-systems, SCCP transmits management messages, encapsulated in UDT message, sent between the entities of each SCCP.

- Subsystems state verification messages are generated and sent periodically (approximately every 30 seconds) to all prohibited sub-systems in order to determine when routing to those destinations becomes available. SCCP also provides an option to make sub-systems *concerned* about the state of other sub-systems so that any changes in the routing process are reported immediately.
- SCCP also provides an advanced addressing capability where a sub-system is represented as an array of digits known as a *Global Title*. A Global Title is a method of hiding the SS7 point code and sub-system number from the originator of a message, for example inter-working between different networks where no common allocations of SS7 point codes are provided. Such a method is used in GSM mobile roaming between countries.
- Depending on network topology, Global Titles are translated either at a STP or at a gateway exchange where a network has an inter-working function with an adjacent network.
- The addressing information delivered to SCCP for message routing may therefore contain a destination point code, a sub-system number and optionally a global title. For successful message transmission, the minimum requirement is for a destination point code in order for the message to leave the SCCP node. If none is present, the called address information is submitted for Global Title Translation. This will hopefully produce as a minimum a destination point code and optionally a sub-system number or new global title. The called address information in a received message contains a routing indicator to instruct SCCP to route on either point code and sub-system number or Global Title (if present). If set to route on Global Title, the called address is submitted for translation to produce a new destination address, which may be the local node or a different SCCP node in the network, which may itself translate the address information again.

- Figure 16 shows how Global Titles are used in GSM-mobile operation to locate subscriber account information (stored in a *Home Location Register* sub-system, HLR) from other networks as used for international roaming. The subscribers account information is held in a database in the home network, which has to be interrogated in order for the subscriber to obtain service from the visited network. The database query is sent through SCCP, with a called address Global Title constructed from information within the subscribers handset (generally either the Equipment Identity or Mobile Subscriber Number), this giving sufficient information to route the message to the correct outgoing gateway using global title translation. Subsequent translation within the home network routes the query to the correct database.

- Global title translation can also be used to determine the location of a free-phone number translation database (held at a SCP), by using the 800 number as a Global Title which is translated at an STP to give the database containing the entry for a range of 800 numbers. For example, 800-1xxxxx could match to database A and 800-2xxxxx could match to database B. This is illustrated in fig. 17.

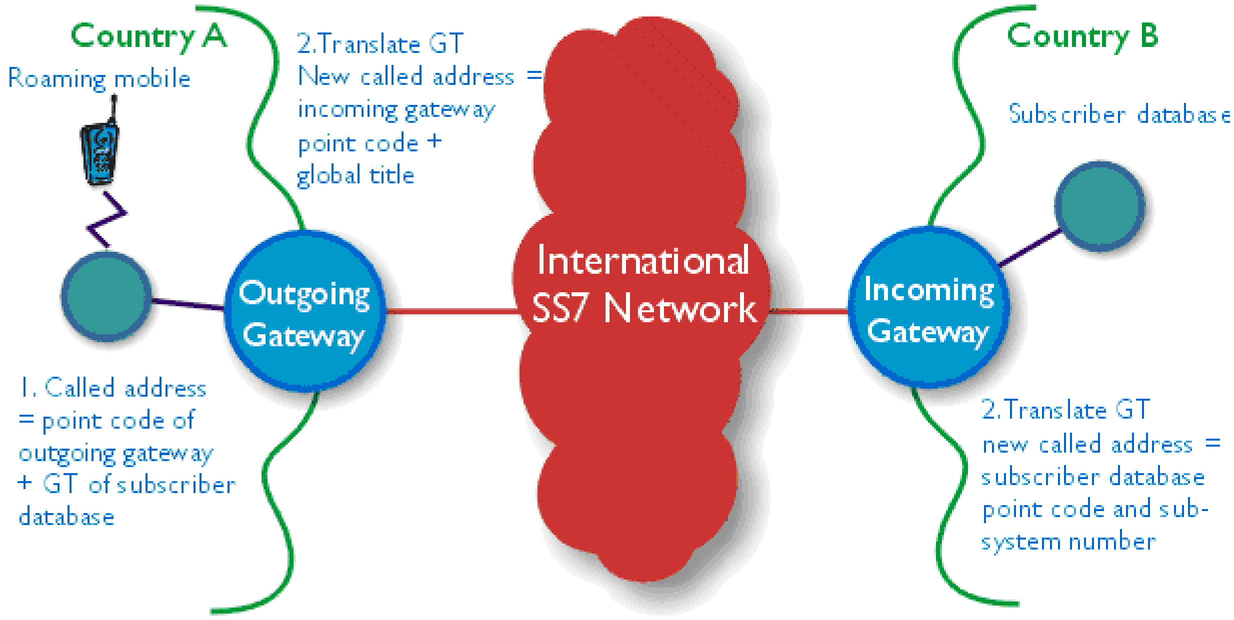

Fig. 16 Use of global title translation (GTT) in mobile roaming

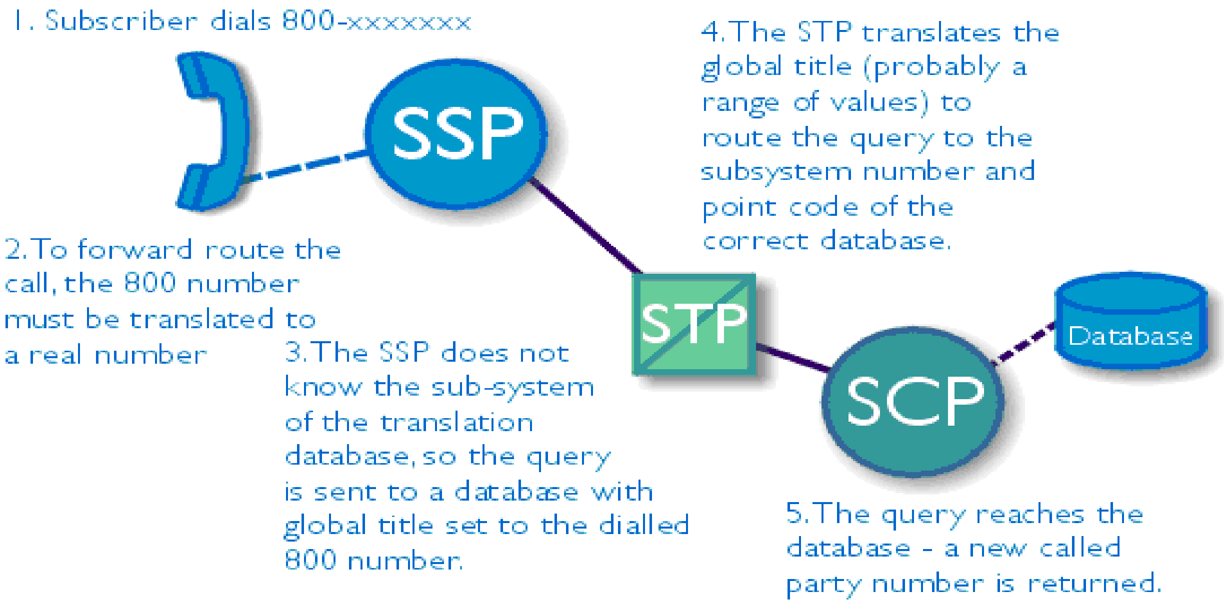

Fig. 17. Use of global title translation (GTT) to locate 800 number translation data base

- **Transaction Capabilities (TCAP or TC)**
- The Transaction Capabilities Application Part provides a structured method to request processing of an operation at a remote node, defining the information flow to control the operation and the reporting of its result.
- Operations and their results are carried out within a session known as a dialogue (at the ‘top’ of TCAP) or a transaction (at the ‘bottom’ of TCAP). Within a dialogue, many operations may be active, and at different stages of processing. The operations and their results are conveyed in information elements known as *components*. The operation of TCAP is to store components for transmission received form the higher layers until a dialogue handling information element is received, at which time all stored components are formatted into a single TCAP message and sent through SCCP to the peer TCAP.

- In the receive direction, TCAP unpacks components from messages received from SCCP and delivers each as a separate information element to the upper protocol layer. Figure 18 shows a general TCAP information flow.

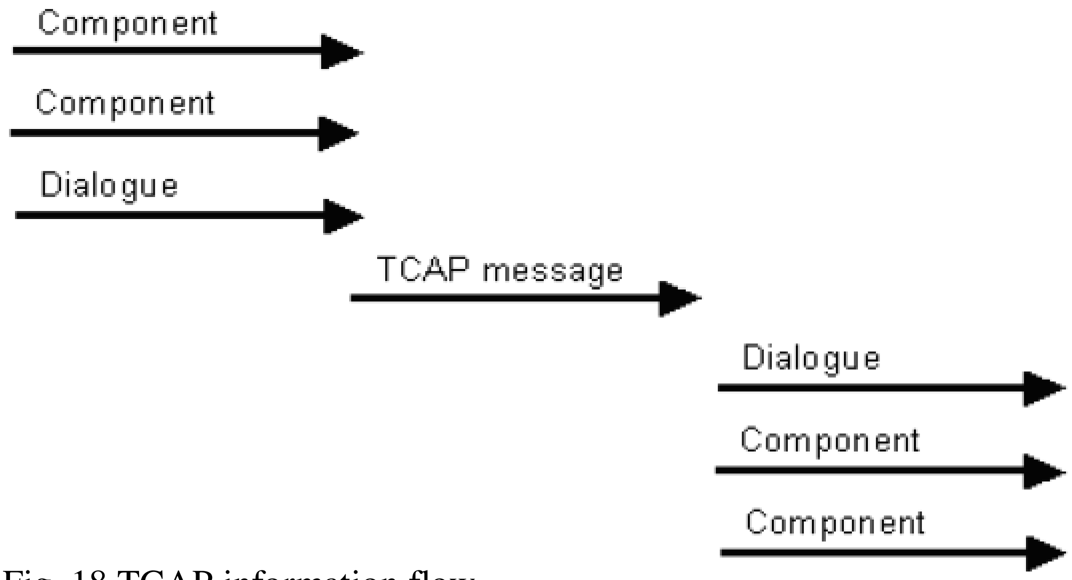

Fig. 18 TCAP information flow

- Typical applications of TCAP are mobile services (e.g. registration of roamers), Intelligent Network services (e.g. free-phone and "calling card" services), and operation, administration and maintenance (OA&M) services.

- **Mobile Application Part (MAP)**
- The Mobile Application Part (MAP) is used within mobile/wireless networks to access roaming information, control terminal hand-over and provide short message services (SMS). It typically uses TCAP over SCCP and MTP as a transport mechanism.
- Mobile networks are database intensive; the point of subscription of a subscriber is a database known as a *Home Location Register* (HLR). When a subscriber roams to a cell and registers with the network, information regarding the subscriber is temporarily stored at the visited equipment in a second database type known as Visitor Location Register (VLR). MAP specifies a set of services and the information flows that implement these services to enable information to be transferred from these databases, in order to register, locate and deliver calls to a roaming subscriber (the roaming terms refers also to the change of the MSC (Mobile Switching Center) and not only to international calls).

- Figure 19 and table 2 show how a mobile terminated call is routed.

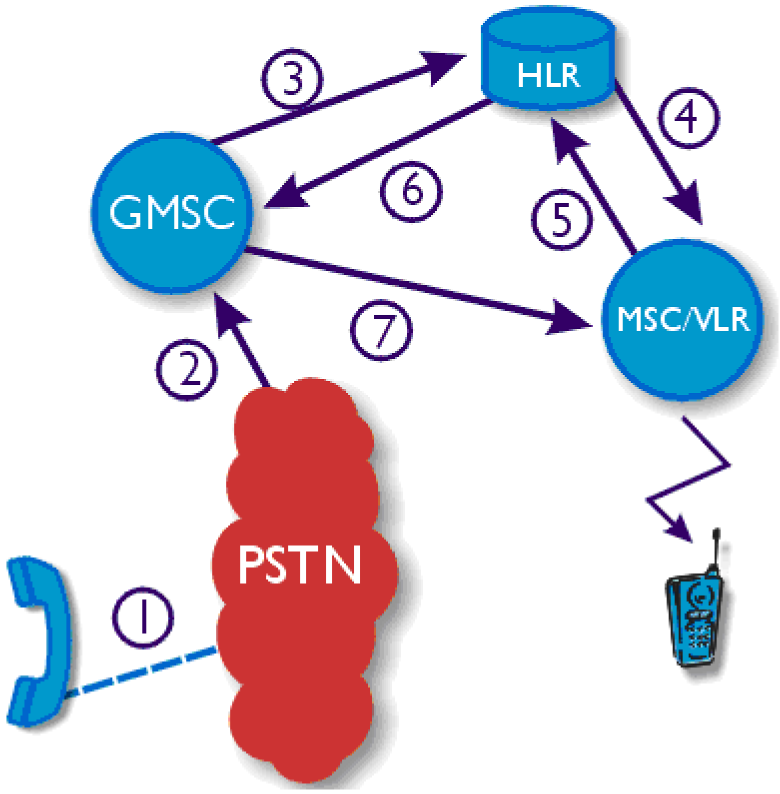

Fig. 19 Mobile terminal call

1 The calling subscriber dials the mobile subscriber.
2 The mobile network prefix digits cause the call to be routed to the mobile network gateway MSC.
3 The gateway MSC uses information in the called address digits to locate the mobile subscribers HLR.
4 The HLR has already been informed of the location (VLR address) for the mobile subscriber and requests a temporary routing number to allow the call to be routed to the correct MSC.
5 The MSC/VLR responds with a temporary routing number that will be valid only for the duration of this call.
6 The routing number is returned to the GMSC
7 The call is performed using standard ISUP (or similar) signaling between the GMSC and the visited MSC.

Tab. 2 Stages of a mobile terminal call