ISDN

Integrated Services Digital Network

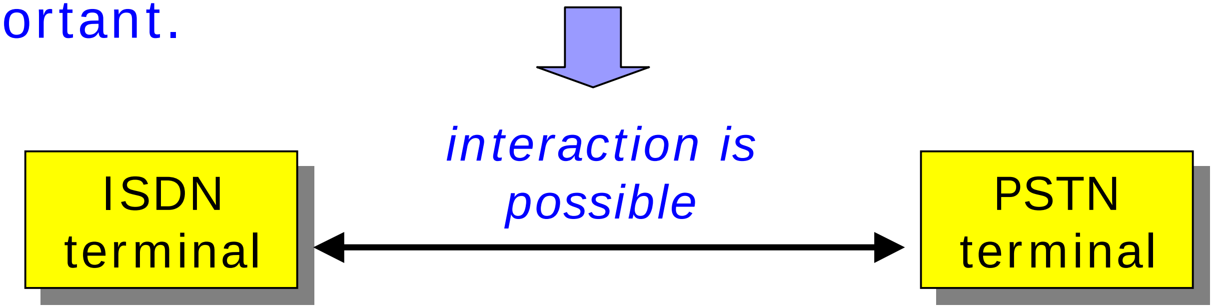
- definition of ISDN
- evolution to ISDN and beyond
- ISDN services
- basic BRA / PRA architecture
- protocols & signalling

What is ISDN ?

1. End-to-end digital connectivity
2. Enhanced subscriber signaling
3. A wide variety of new services (due to 1 and 2)
4. Standardized access interfaces and terminals

Idea originated in the 1980's

ISDN is not a “new” network separated from the PSTN. Interworking with “normal” PSTN equipment is very important.



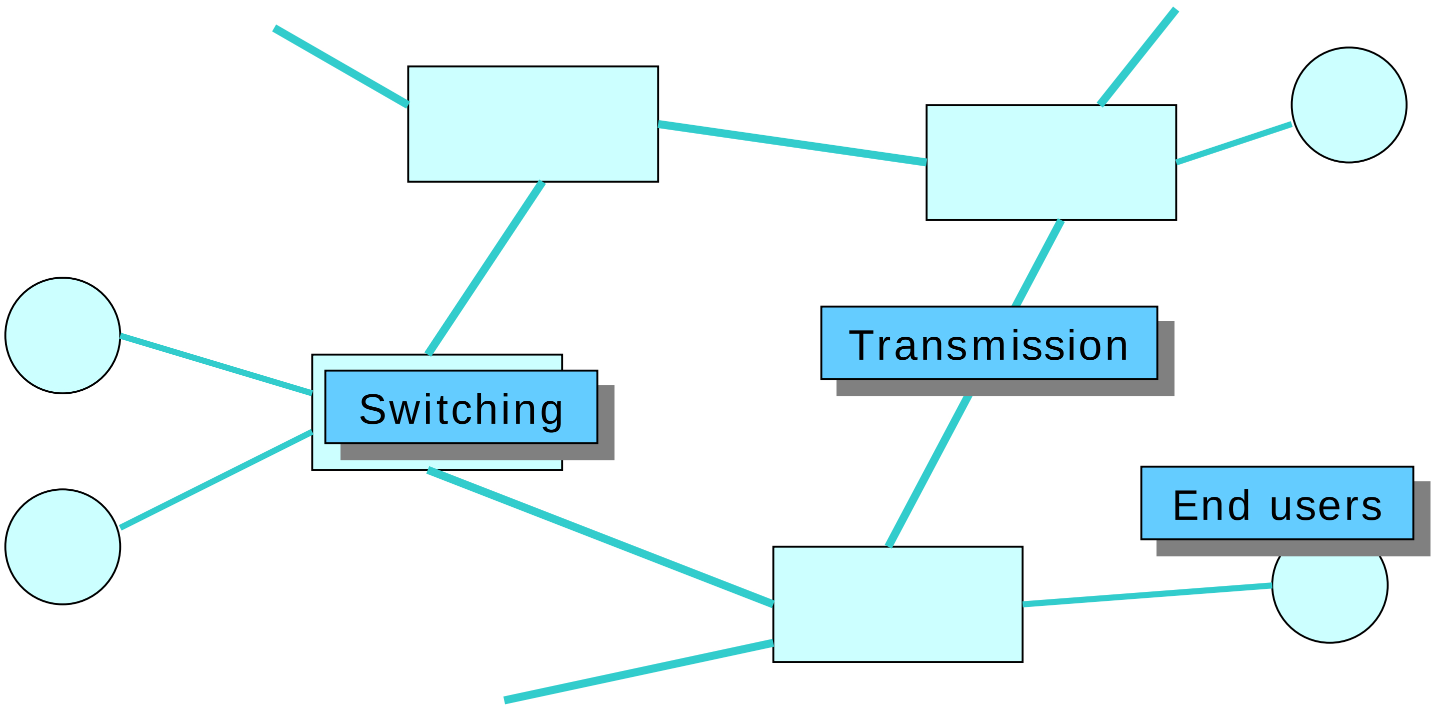
Evolution towards ISDN and beyond

How does ISDN fit into the telecom network evolution in general?

1. First the network was all-analogue and voice-centric
2. Digital transmission (PDH) in the core network
3. Digital switching at 64 kbit/s
4. SS7 replaces channel associated signalling systems
5. SDH replaces PDH
6. ISDN offers digital technology to end users
7. DSL (primarily ADSL) technology takes over

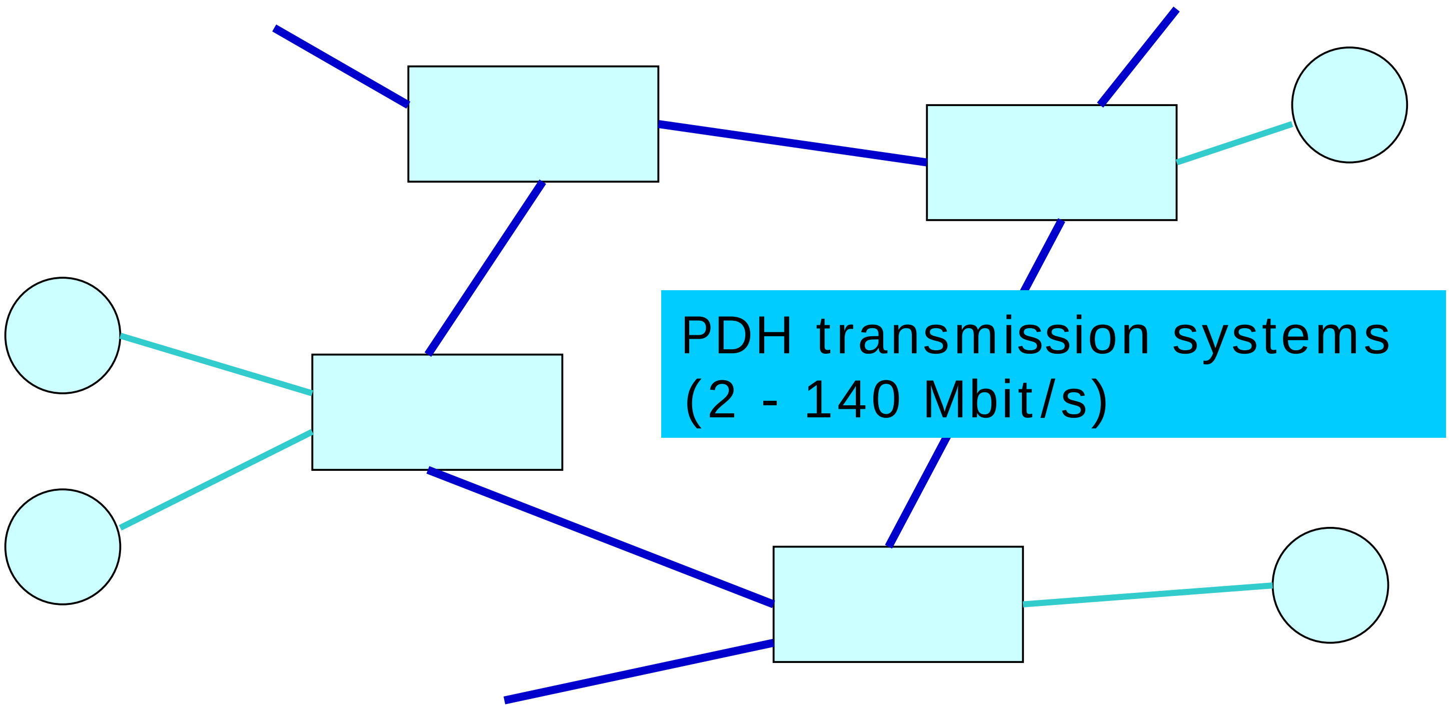
Evolution history

Step 1: All-analogue network (before 1960)



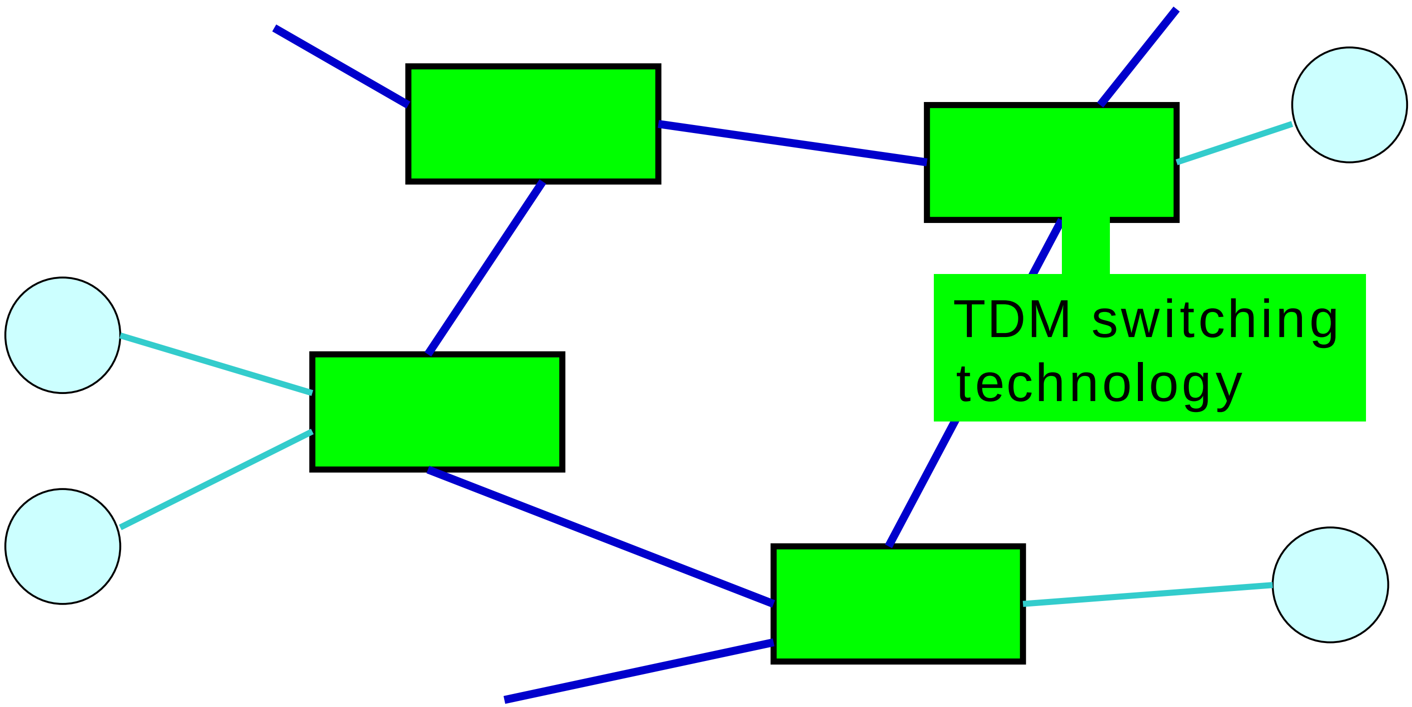
Evolution history

Step 2: Digital transmission in the core network
(1960 - 1980)



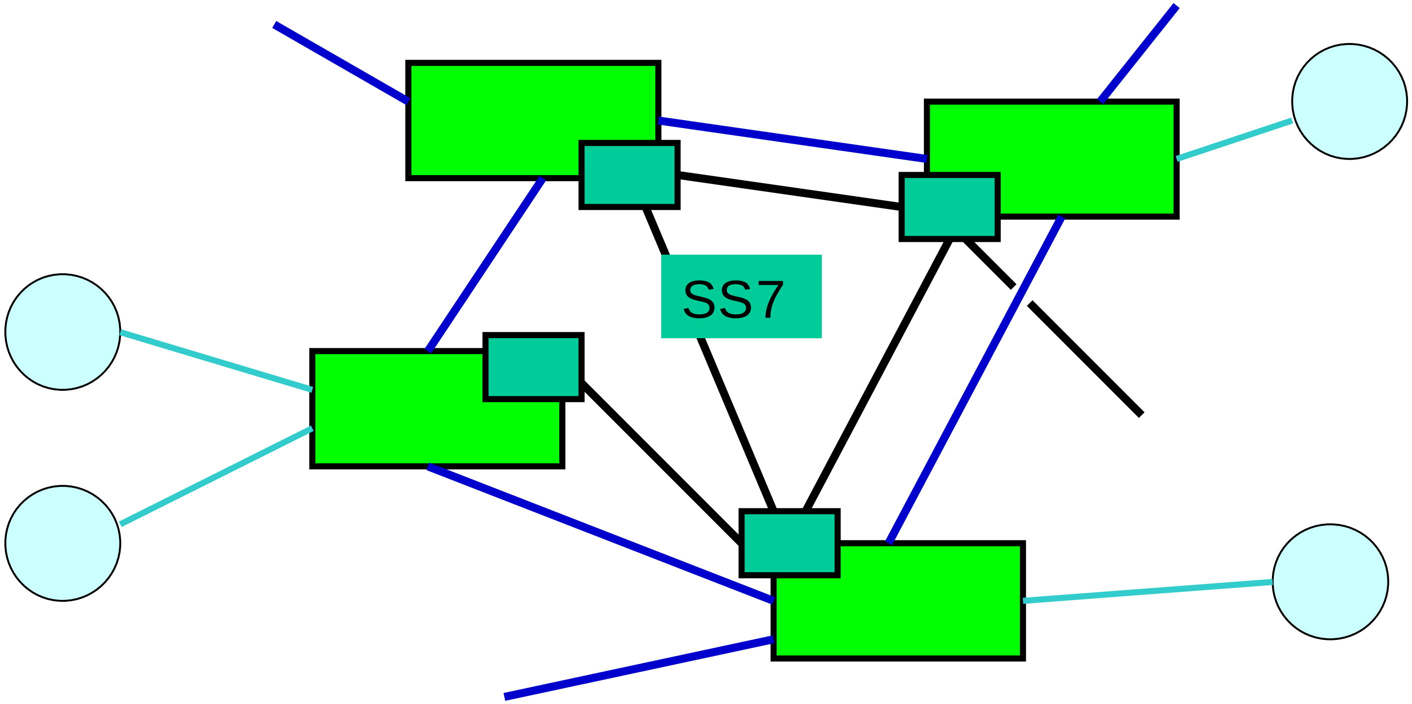
Evolution history

Step 3: Digital switching at 64 kbit/s (1970 - 1990)



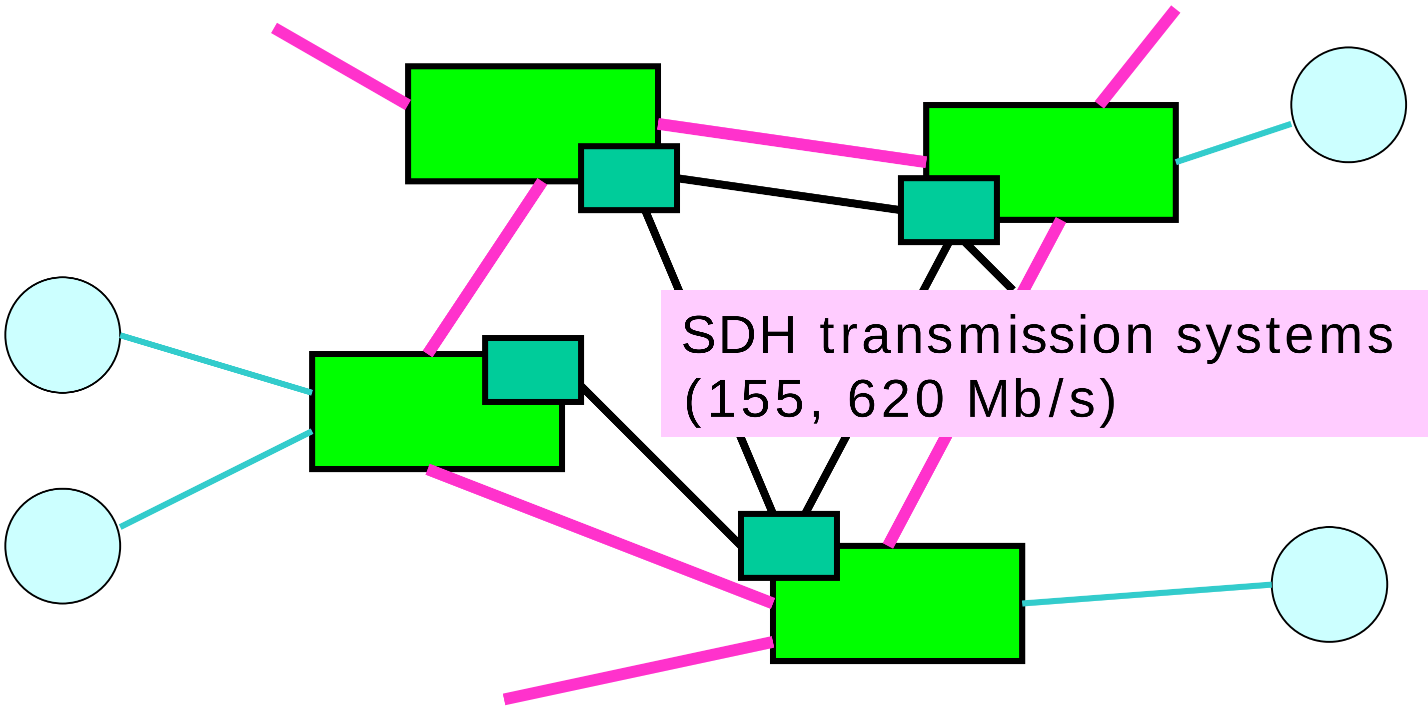
Evolution history

Step 4: Common Channel Signalling in the core network (1980 - 2000)



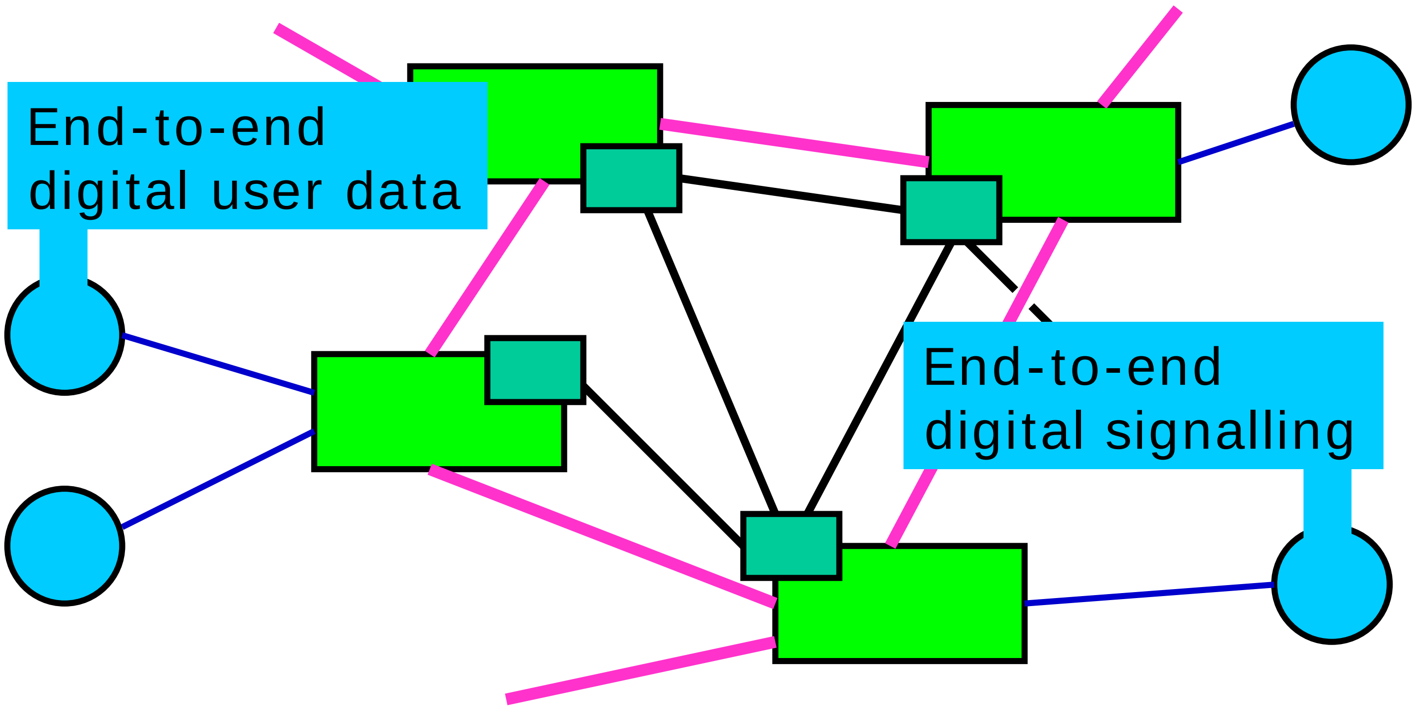
Evolution history

Step 5: PDH systems are replaced by SDH systems (1990 ...)



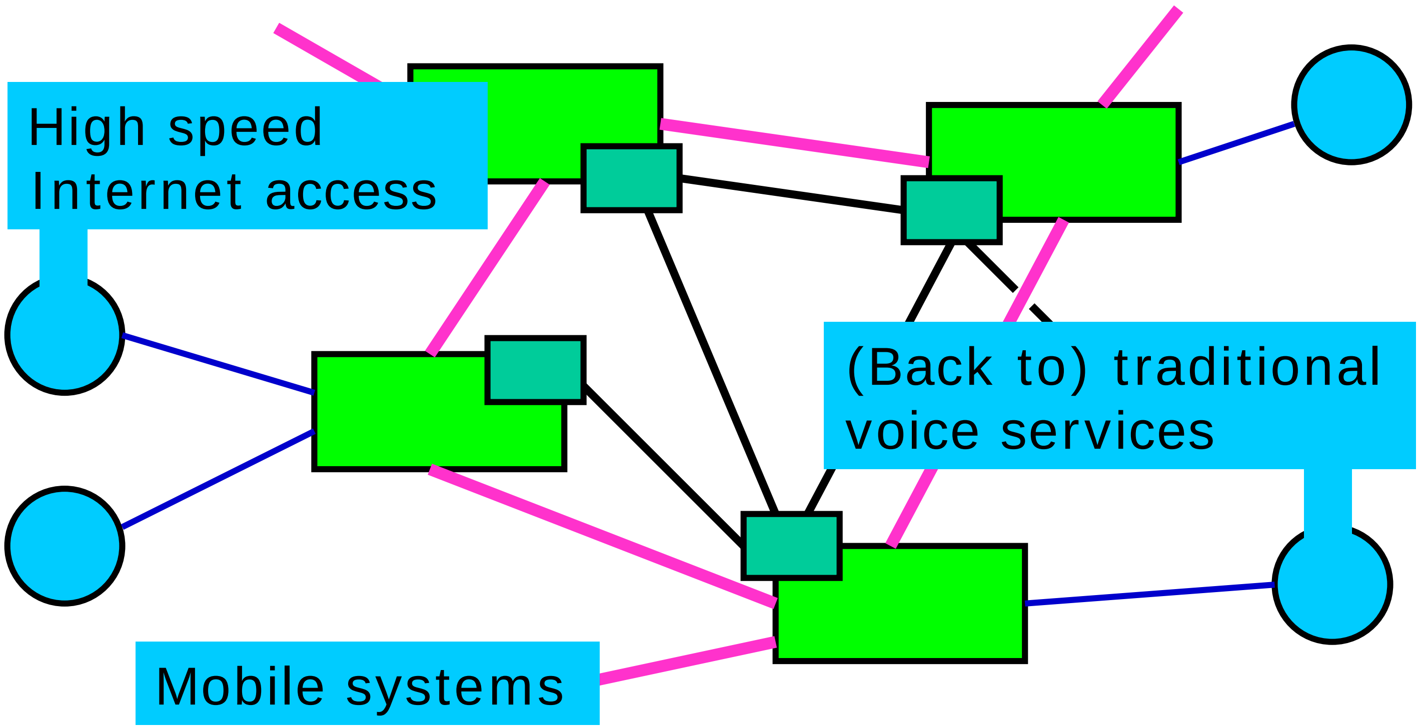
Evolution history

Step 6: ISDN => digital technology to end users



Evolution history

Step 7: ADSL (for Internet services) + analogue voice



PSTN vs. ISDN user access

PSTN	300 ... 3400 Hz analogue transmission band “poor-performance” subscriber signaling
Basic Rate Access ISDN	2 x 64 kbit/s digital channels (B channels) 16 kbit/s channel for signaling (D channel)
Primary Rate Access ISDN	30 x 64 kbit/s digital channels (B channels) 64 kbit/s channel for signaling (D channel)

Digital access: several alternatives

	ISDN BRA	modem	ADSL
Bit rate (kb/s)	2 x 64	max. 50	much larger
Connection setup time	fast	slow	fast
Popularity	little	decreasing	great

However, large impact on signalling protocols

Telecommunication services

... as defined in ISDN standards

Basic telecommunication services

Bearer services provide the capability of transmitting signals between network access points. Higher-level functionality of user terminals is not specified.

Teleservices provide the full communication capability by means of network functions, terminals, dedicated network elements, etc.

Supplementary services

A supplementary service modifies or supplements a basic telecommunication service. It cannot be offered to a customer as a stand-alone service.

Services examples

Some typical teleservices

- Telephony (normal, high quality)
- Telefax (Group 3, Group 4)
- Video-telephony

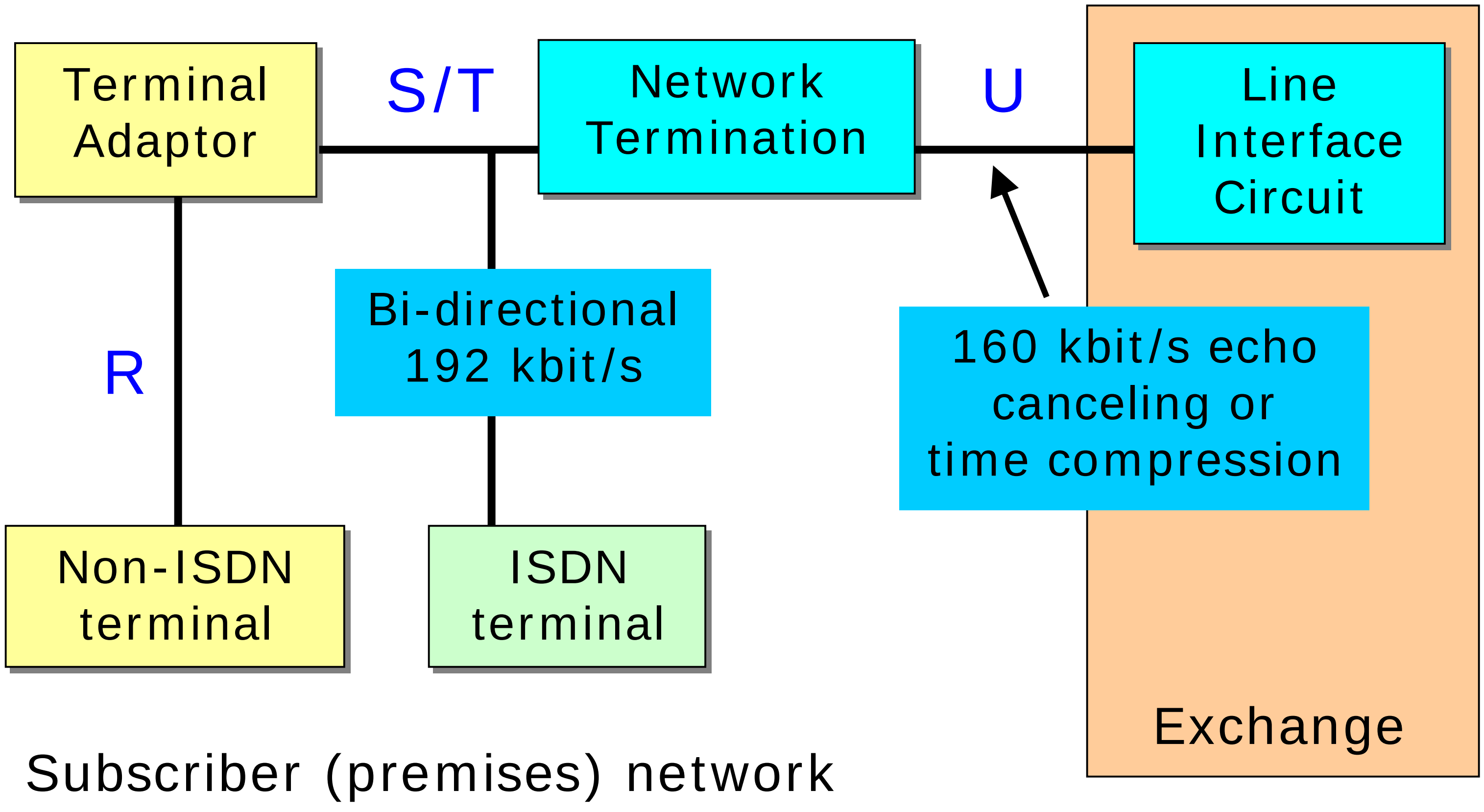
Some typical bearer services

- Speech (transparency not guaranteed)
- 64 kbit/s unrestricted
- 3.1 kHz audio (non-ISDN interworking)

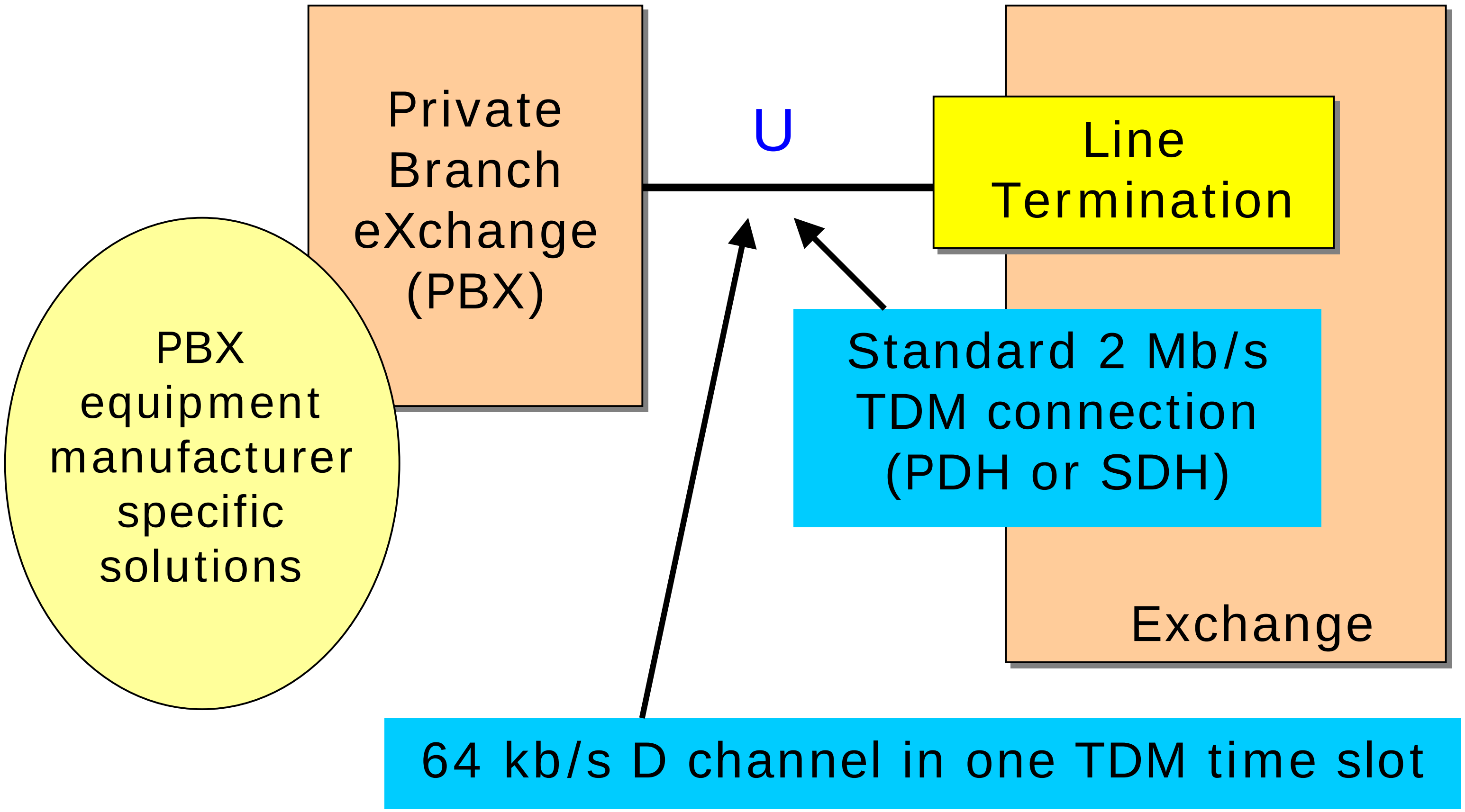
Some typical supplementary services

- CLIP / CLIR
- Call forwarding / waiting / hold
- Charging supplementary services

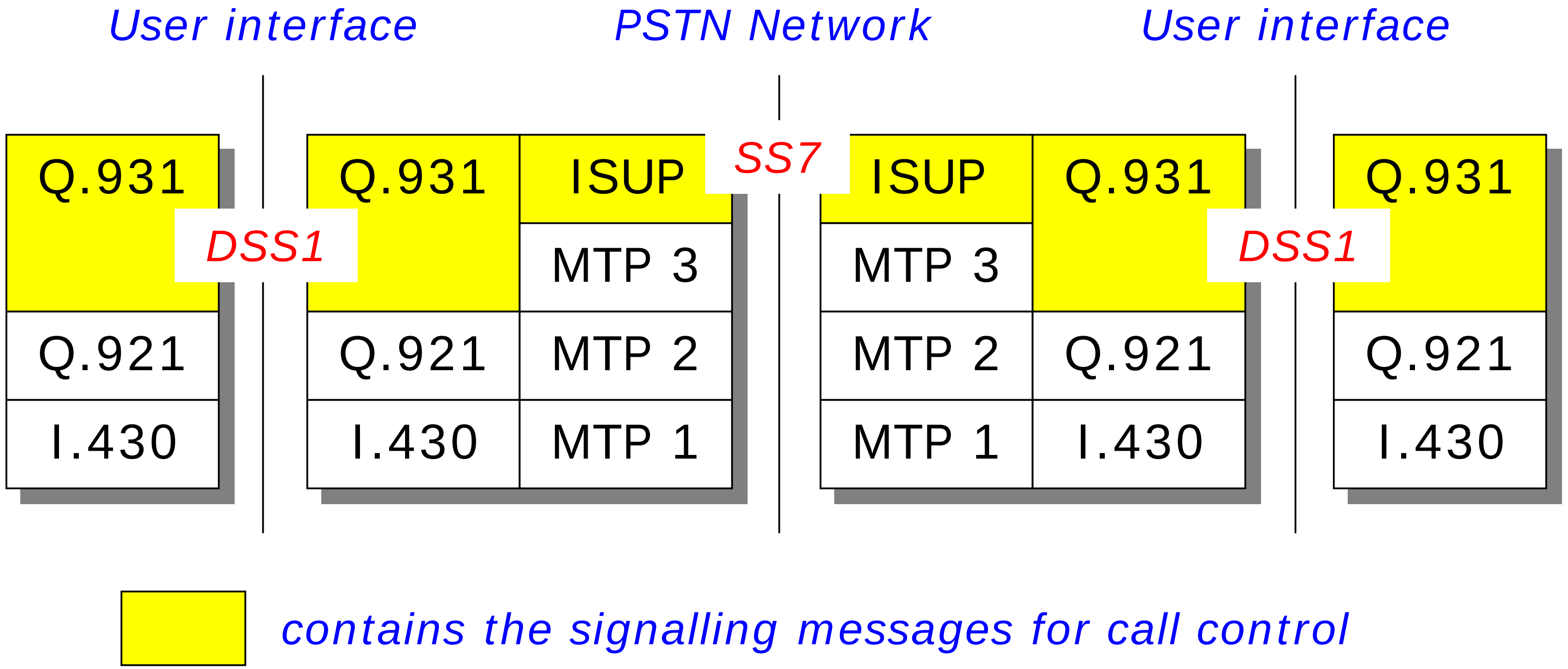
Basic rate access – user interface



Primary rate access – user interface



Signalling protocols for end-to-end circuit-switched digital connection



Layered DSS1 signaling structure

DSS1 = Digital Subscriber Signalling system no.1

Layer 1: I.430

Bit sequence structure, framing & multiplexing

Layer 2: Q.921

Link control (HDLC-type protocol called LAPD)

Layer 3: Q.931

Signaling messages (application layer)

Q.931 Call-related messages

Similar functions as ISUP in SS7

Call establishment messages:

ALERTING
CALL PROCEEDING
CONNECT
CONNECT ACKNOWLEDGE
PROGRESS
SETUP
SETUP ACKNOWLEDGE

Call clearing messages:

DISCONNECT
RELEASE
RELEASE COMPLETE

Typical content of ISDN Set-up message

Called party (user B) number & numbering plan

Calling party (user A) number (+ CLIP/CLIR) Show to B?

Bearer capability (64 kbit/s unrestricted, speech, 3.1 kHz audio, packet mode B-channel, packet mode D-channel)

Channel identification (B1, B2, or D channel request)

Low-layer compatibility (type of bit rate adaptation, type of modem ...)

High-layer compatibility (teleservice-related issues)

Keypad facility

Example: Structure of Release message

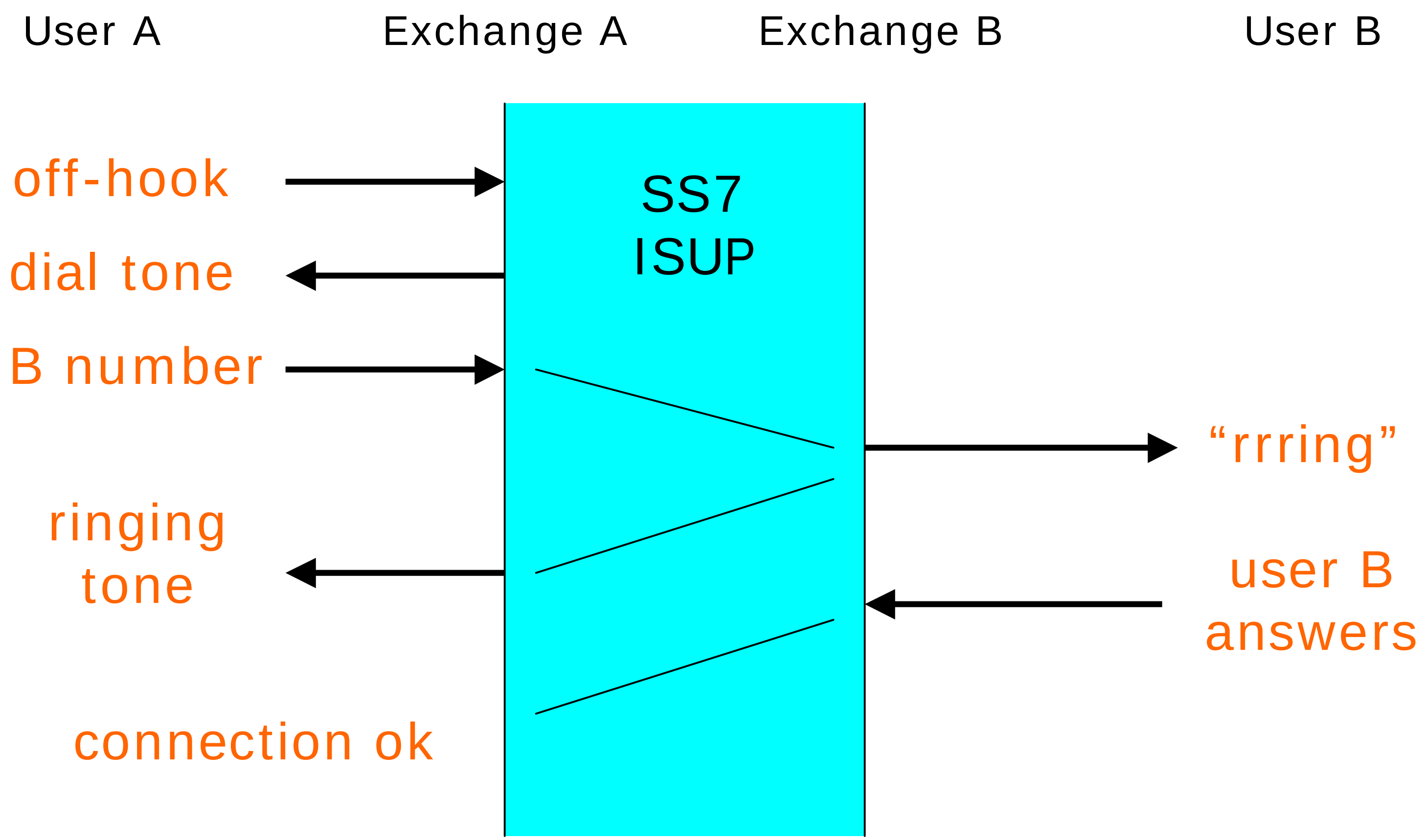
Message type: RELEASE
Significance: Local
Direction: Both

Common header part of message

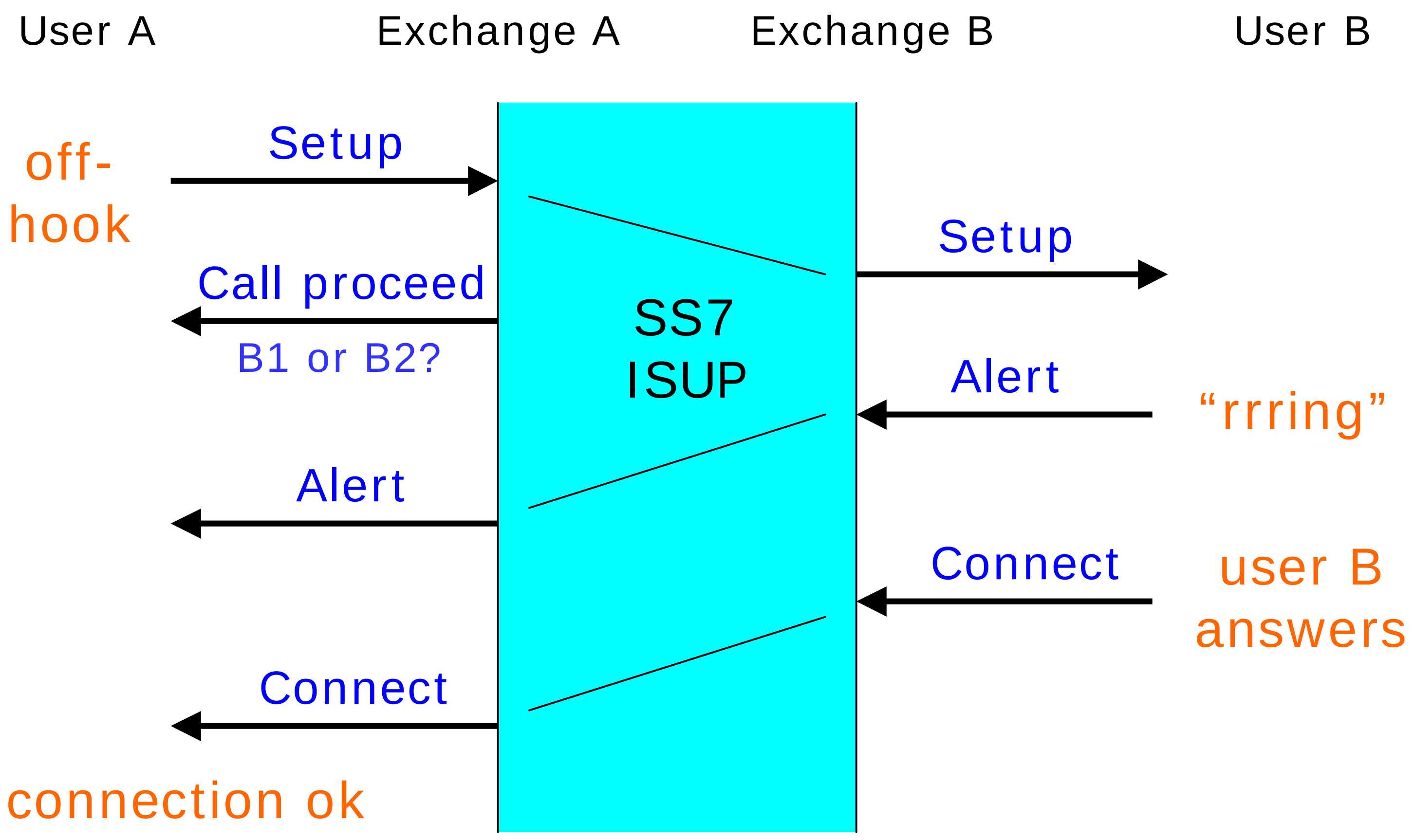
Info Element	*Direction*	*Type*	*Length*
Protocol discriminator	Both	M	1
Call reference	Both	M	2-
Message type	Both	M	1
Cause	Both	O	2-32
Display	n $\rightarrow$ u	O	
Signal	n $\rightarrow$ u	O	2-3

Cause description may require many bytes

Setup of an “old-fashioned” PSTN call



Setup of an ISDN call using Q.931



SS7

Common Channel Signalling System Nr. 7

Bhatnagar, Chapter 4

- CCS vs. CAS
- SS7 protocol structure
- basic signalling examples
- MTP, ISUP and SCCP

History of inter-exchange signalling

CAS

Before 1970, only channel-associated signalling (CAS) was used. In CAS systems, signalling always occurs in-band (i.e. over voice channels).

CCIS

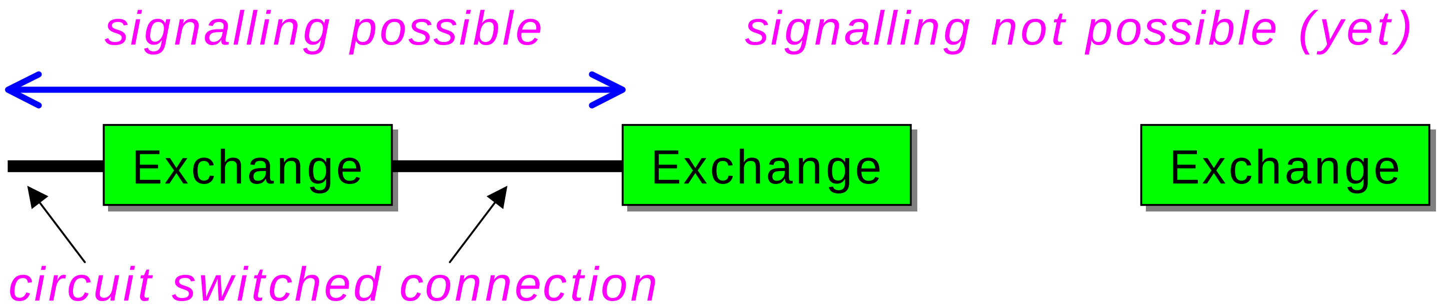
SS6 = CCIS (common channel interoffice signaling) was widely deployed in North America, but not in Europe (=> concentrating on SS7 instead).

SS7

Starting from 1980 (mainly in Europe), CAS was being replaced by SS7. The use of stored program control (SPC) exchanges made this possible. Like CCIS, signalling messages are transmitted over separate signalling channels. Unlike CCIS, SS7 technology is based on protocol stacks.

Channel-associated signalling (CAS)

CAS means in-band signalling over voice channels.



CAS has two serious draw-backs:

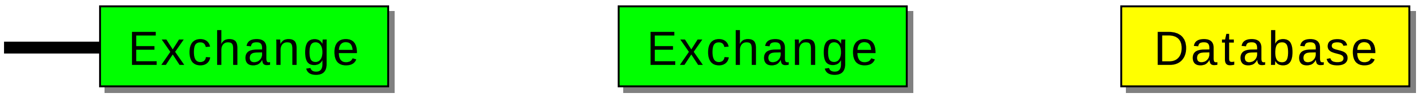
1) Setting up a circuit switched connection is very slow.

2) Signalling to/from databases is not feasible in practice (setting up a circuit switched connection to the database and then releasing it would be extremely inconvenient).

Common channel signalling (CCS)

In practice, CCS = SS7
In Finnish: CCS = yhteiskanavamerkinanto (YKM)

signalling possible anywhere anytime

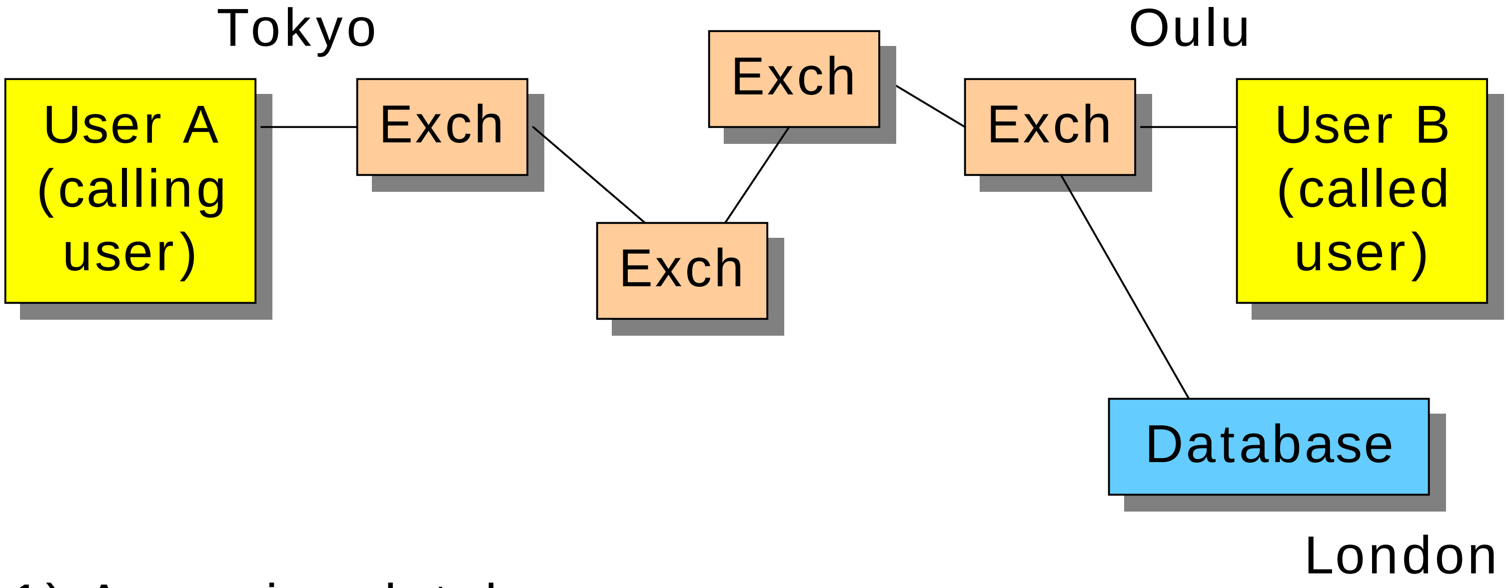


The packet-switched signalling network is separated from circuit switched connections. Consequently:

1) Signalling to/from databases is possible anytime.

2) End-to-end signalling is possible before call setup and also during the conversation phase of a call.

CAS vs. CCS example

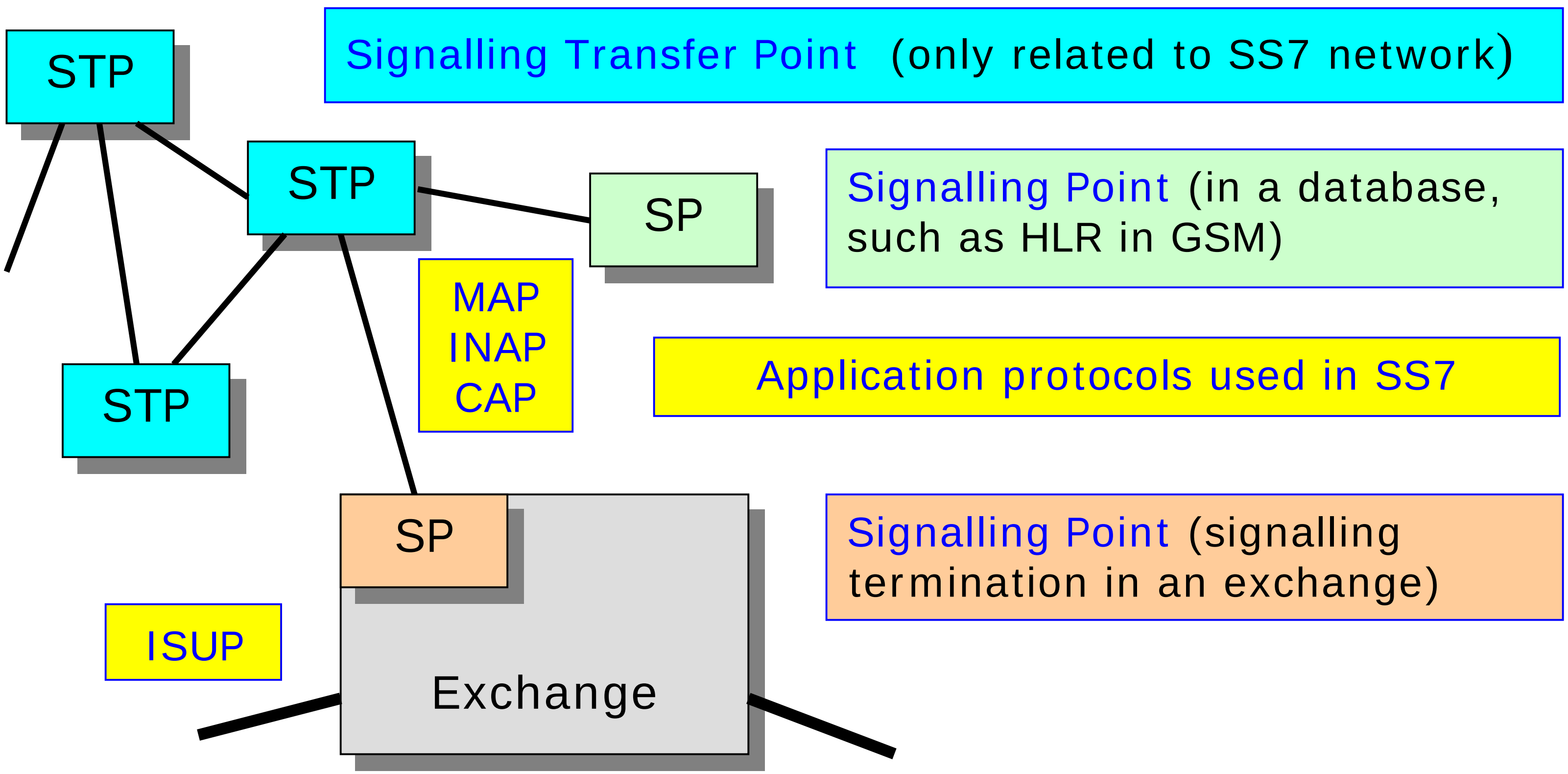


1) Accessing database

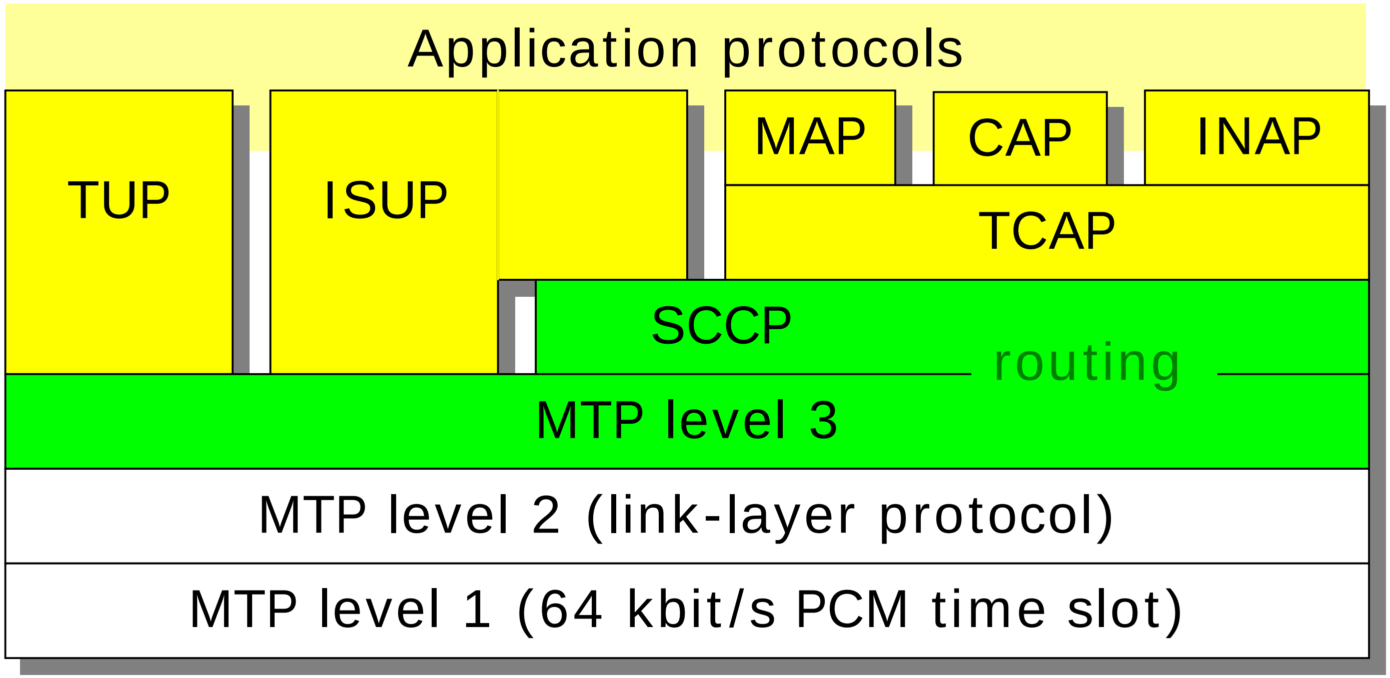
2) End-to-end signalling before call setup

Signalling points (SP) in SS7

Every SP is identified by a unique signalling point code



Protocol layers ("levels") of SS7



MTP - Message Transfer Part

SCCP - Signalling Connection Control Part

UP - User Part   AP - Application Part

Application protocols in SS7

TUP (Telephone User Part) – is being replaced by ISUP

ISUP (ISDN User Part) – for all signalling related to setting up, maintaining, and releasing circuit switched connections

MAP (Mobile User Part) – for transactions between exchanges (MSC, GMSC) and databases (HLR, EIR, AuC) in mobile networks

INAP (Intelligent Network Application Part) for IN applications in fixed networks

CAP (CAMEL Application Part) for extended IN functionality in mobile networks (where MAP is not sufficient ...)

MTP functions

MTP level 1 (signalling data link level):

Digital transmission channel (64 kbit/s TDM time slot)

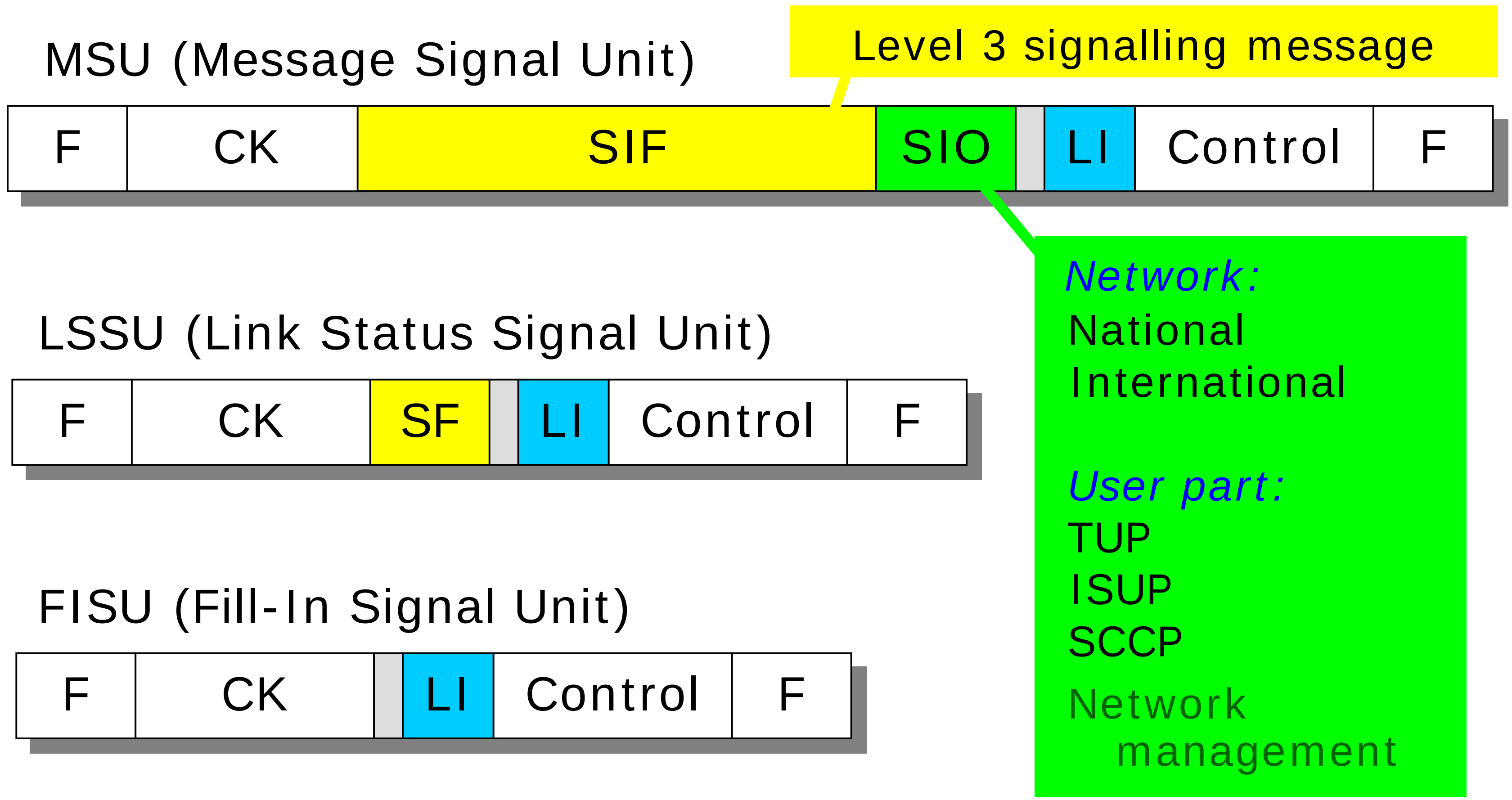
MTP level 2 (signalling link level):

HDLC-type frame-based protocol for flow control, error control (using ARQ), and signalling network supervision and maintenance functions.

MTP level 3 (signalling network level):

Routing in the signalling network (using OPC, DPC) between SPs with level 4 users (see SIO at level 2).

MTP level 2 frame formats



MTP level 2 frames

MSU (Message Signal Unit):

- Contains signalling messages (User Part ⇦ SIO)
- The received frame is MSU if LI > 2 (number of payload octets, payload = SIF or SF)

LSSU (Link Status Signal Unit):

- Contains signalling messages for link supervision
- The received frame is LSSU if LI = 1 or 2

FISU (Fill-In Signal Unit):

- Can be used to monitor quality of signalling link
- The received frame is FISU if LI = 0

Routing information in SS7 message

Level 3 signalling message in SIF (Signalling Information Field)

Routing label

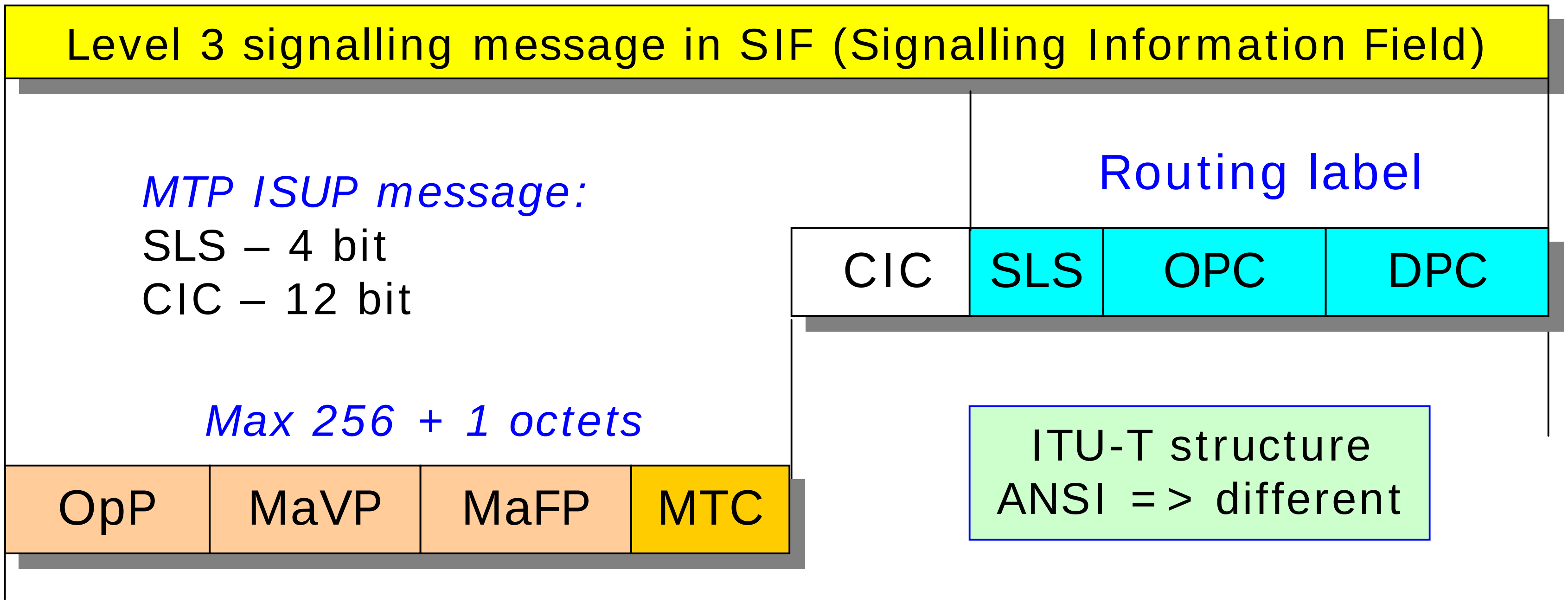
MTP management message:
SLC – 4 bit signalling link code

SLC	OPC	DPC

MTP SCCP message:
SLS – 4 bit signalling link selection

SLS	OPC	DPC

Structure of SS7 ISUP message

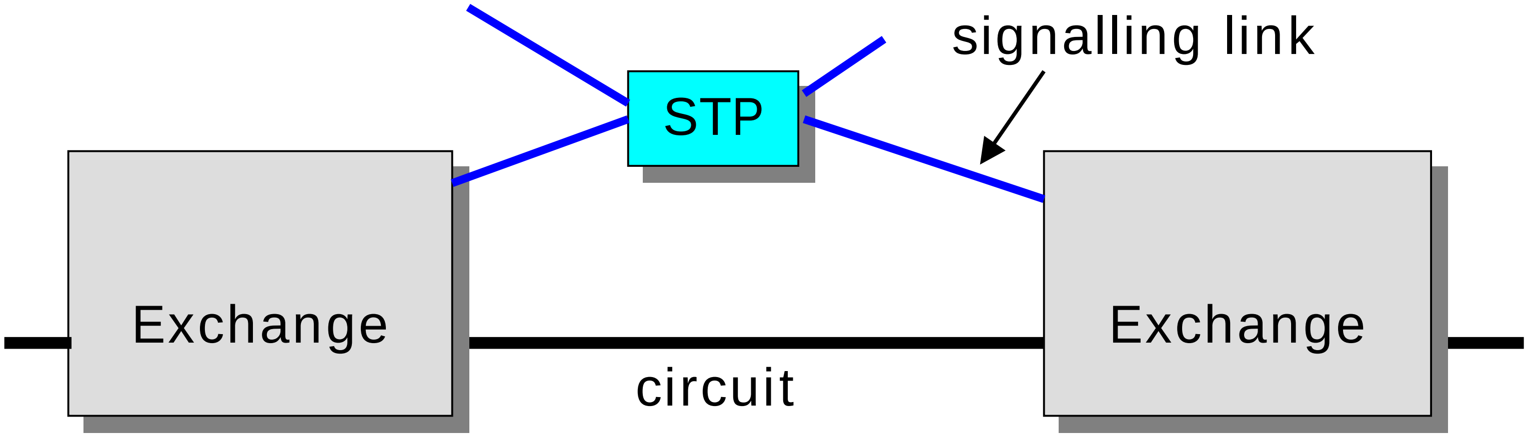


MTC: Message Type Code (name of ISUP message)
MaFP: Mandatory Fixed Part (no LI, no parameter names required)
MaVP: Mandatory Variable Part (LI, no parameter names required)
OpP: Optional Part (LI and parameter names required)

Difference between SLS and CIC

SLS defines the signalling link which is used for transfer of signalling information (SLS enables load sharing).

CIC defines the circuit (used for a certain circuit switched connection) with which the ISUP message is associated.



Identification of signalling points (SP)

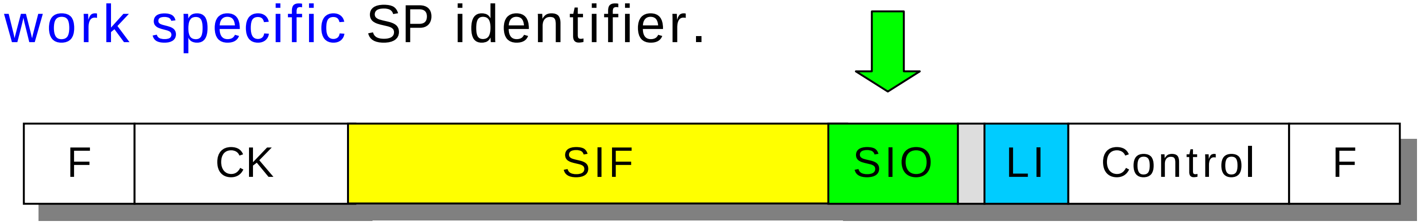
DPC – Destination Point Code (14 bit ⇔ 16384 SPs)

- Termination point of application transaction
- Key information for routing within SS7 network
- DPC is inserted by the originating MTP ”user”.

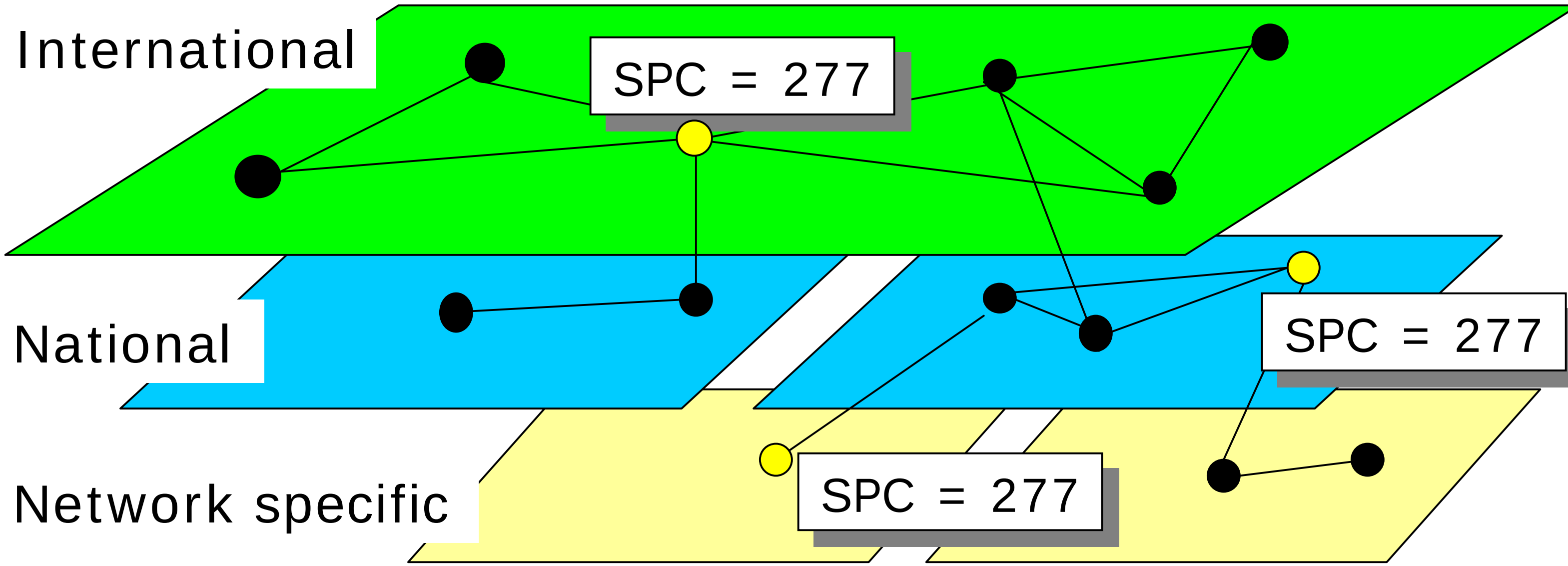
OPC – Originating Point Code (14 bit)

- Originating point of application transaction

The ”network indicator” in the SIO octet indicates whether the DPC or OPC is an international, national, or network specific SP identifier.

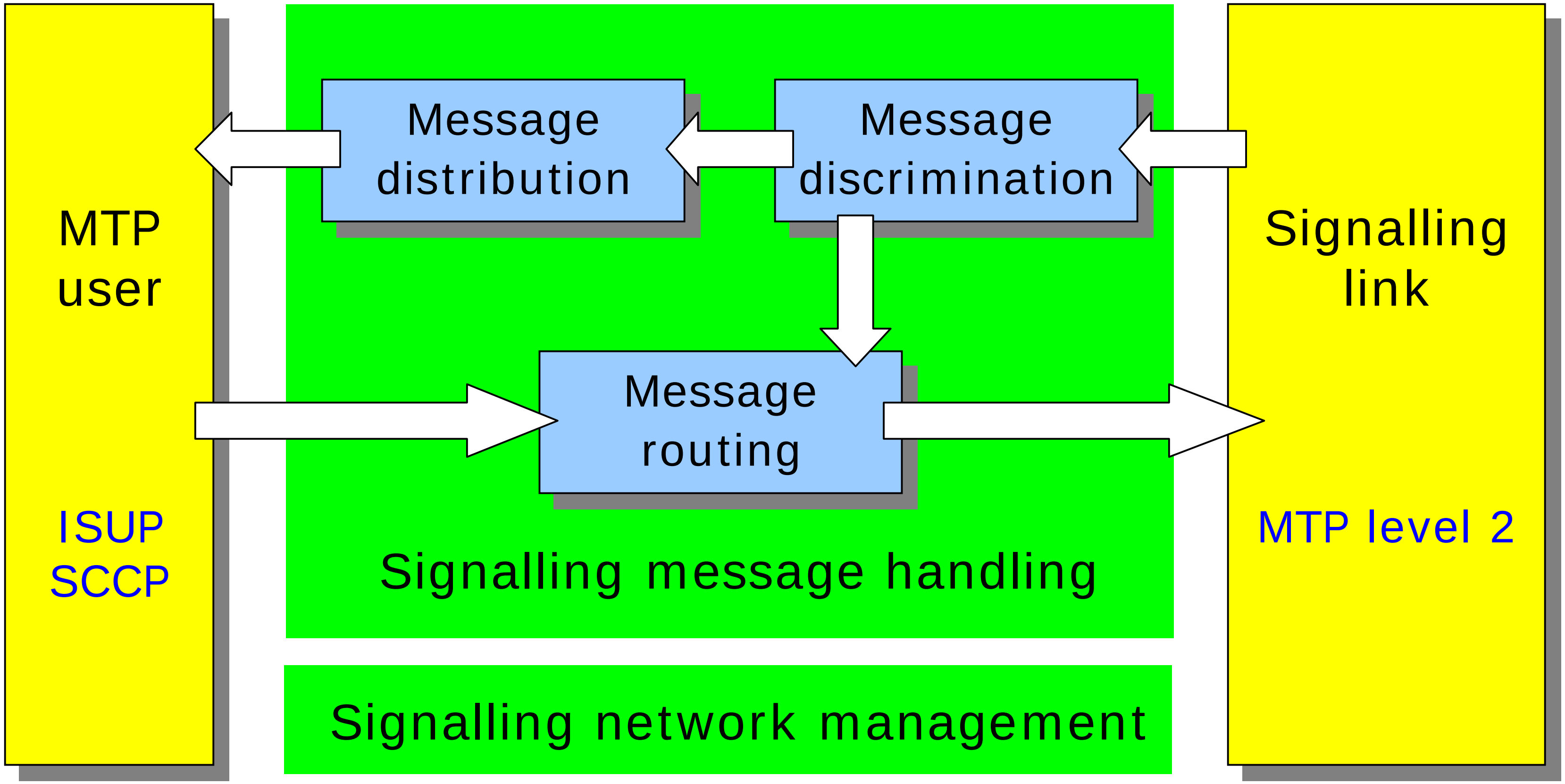


Same signalling point codes can be reused at different network levels



SPC = 277 means different SPs at different network levels

Basic MTP level 3 functions



ISUP (Integrated Services User Part)

Essential for circuit-switching related signalling

Generally used in PSTN (i.e., not only for ISDN)

Features:

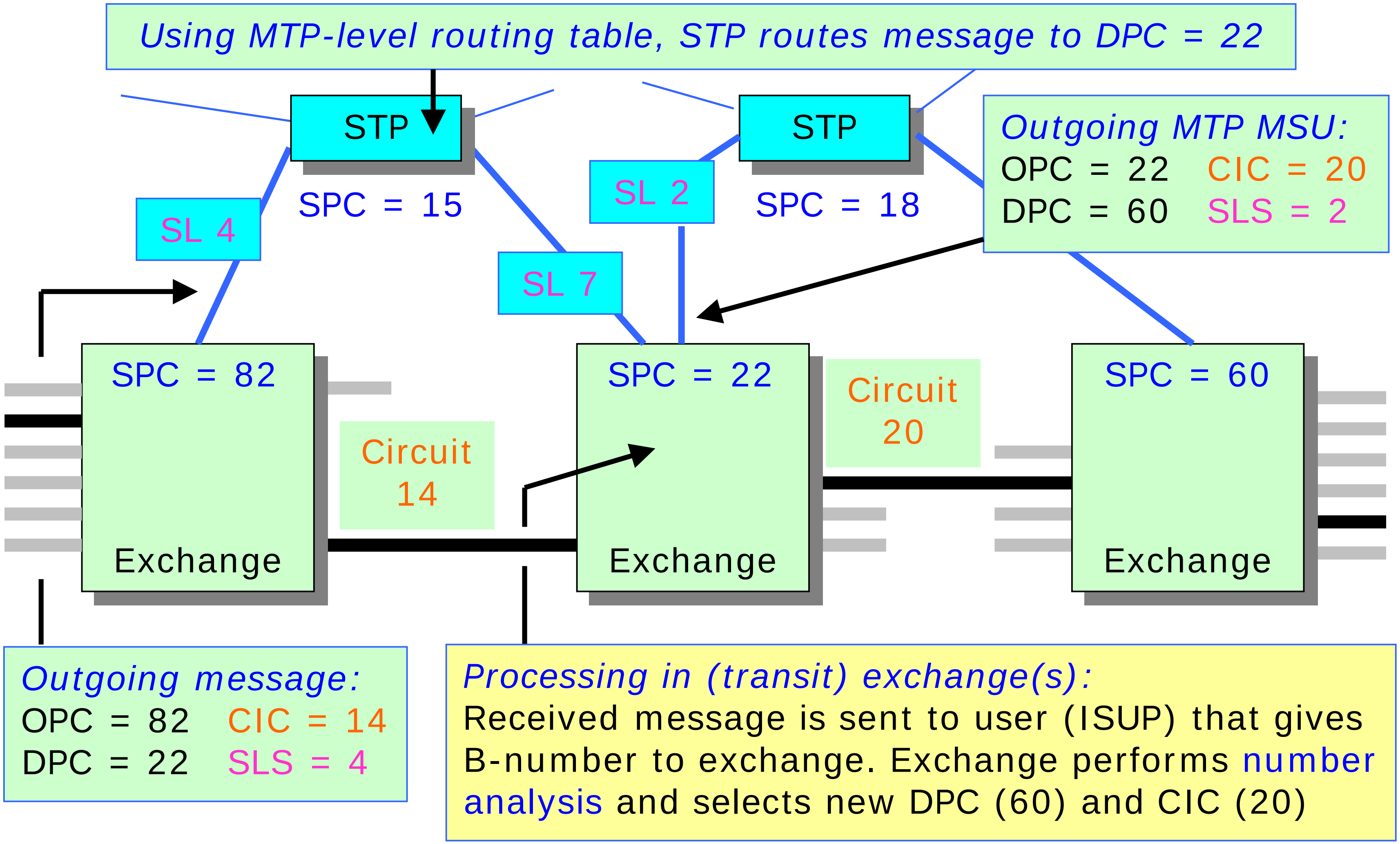
Establishment / release of circuit switched connections (basic call control) using link-by-link signalling

End-to-end signalling between two exchanges (for this purpose SCCP + ISUP is used)

see Bhatnagar, p.77

Only for signalling between exchanges (never to/from a stand-alone database).

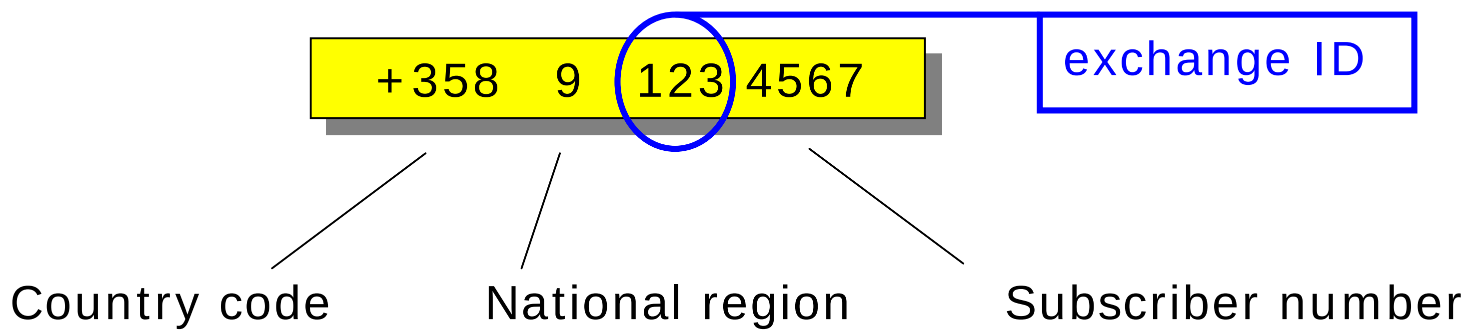
Example: link-by-link signalling (IAM)



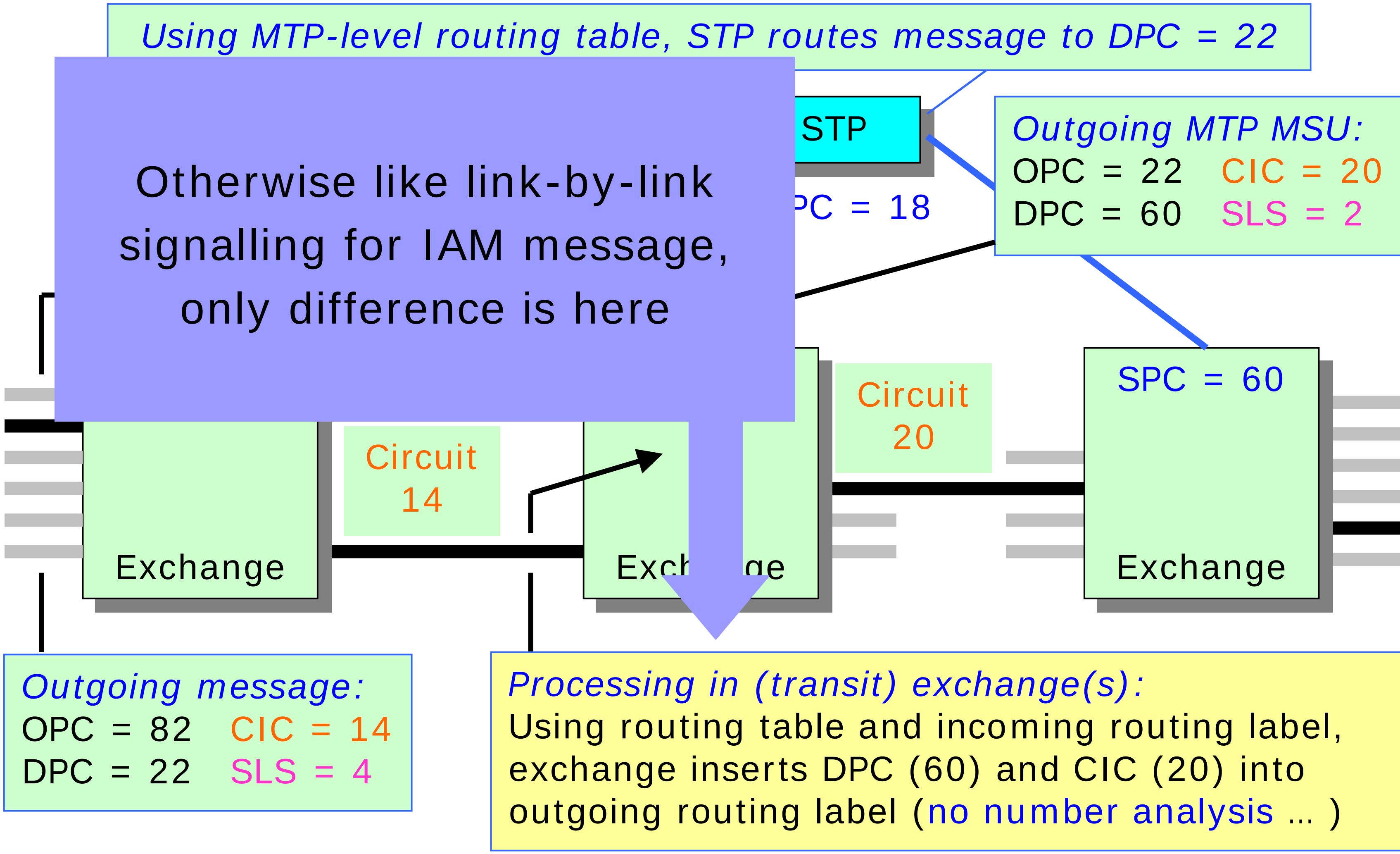
MTP + ISUP in SS7

The routing capability of MTP is rather limited (routing tables are entirely based on signalling point codes).

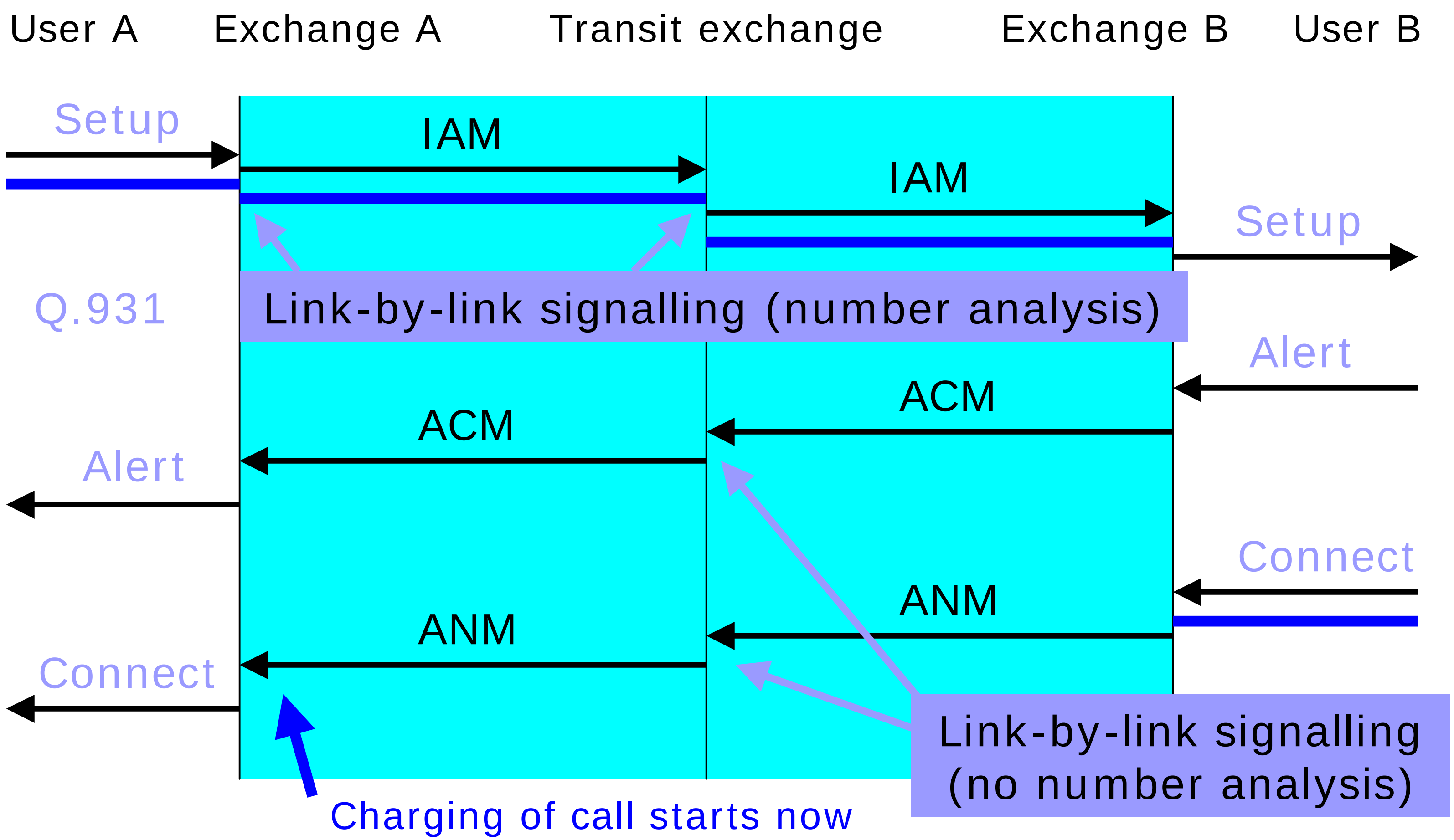
The ”real” routing through the network(s) during call setup is performed by exchanges on an exchange-to-exchange basis, using the dialed digits and routing tables.



Example: link-by-link signalling (non-IAM)



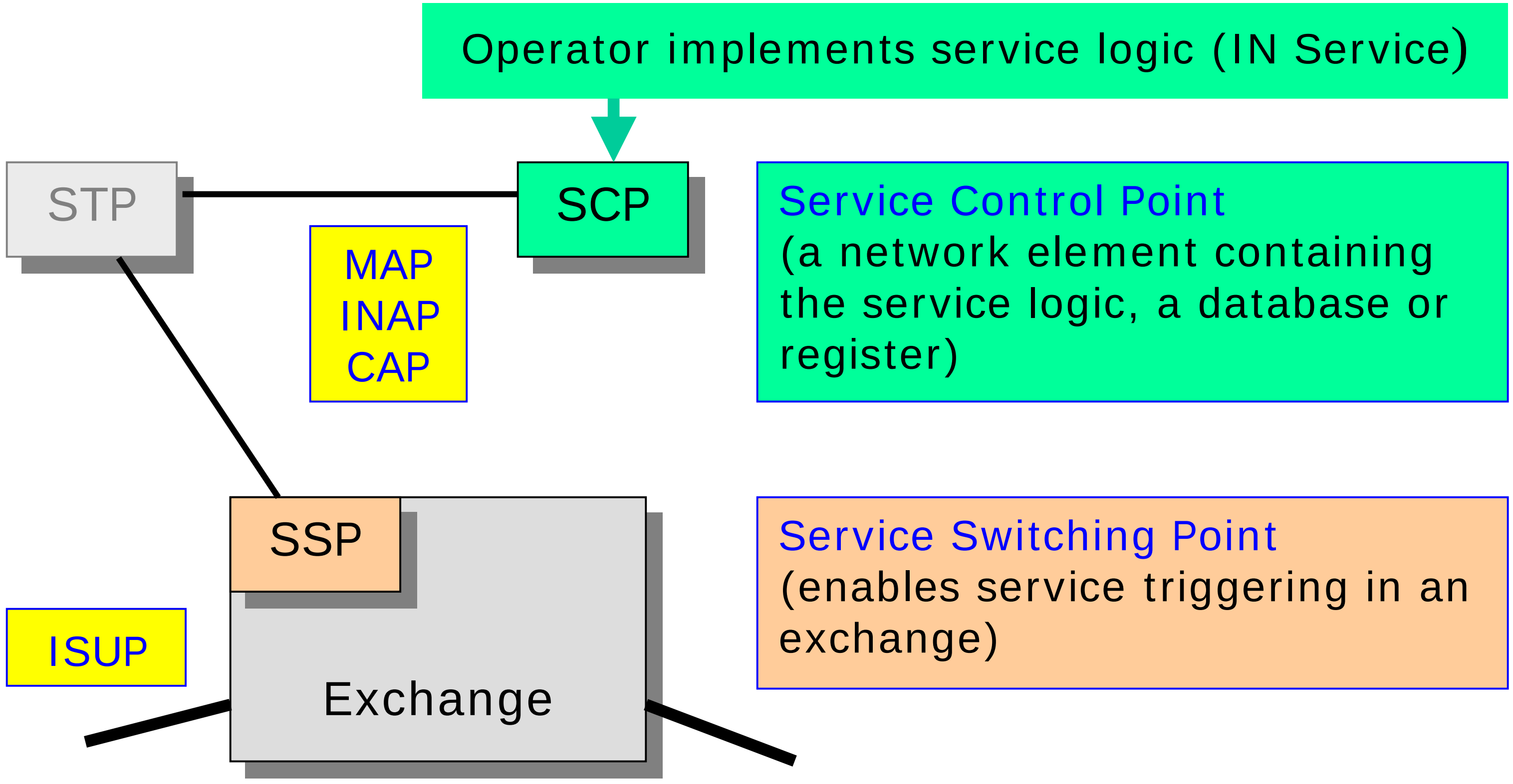
Setup of a call using ISUP



Some basic ISUP messages

	user A user B
IAM – Initial Address Message	→
ACM – Address Complete Message	←
ANM – Answer Message	←
REL – Release Message	⇄
RLC – Release Complete	⇆

Intelligent Network (IN) Concept



SCCP (Signalling Connection Control Part)

Essential for non-circuit-switching related signalling

Features:

OSI Layer 3 functionality

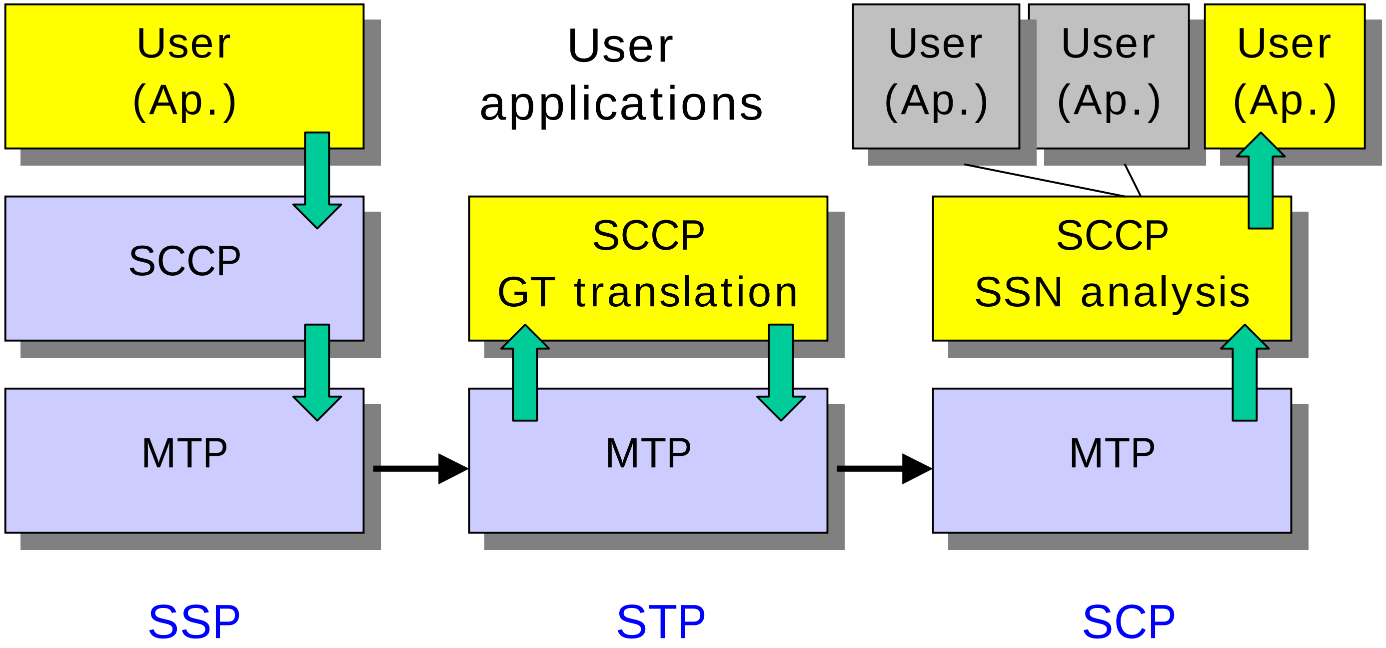
- Essential for end-to-end signalling & database access
- Global Title Translation (GTT) for enhanced routing
- SubSystem Number (SSN) analysis at destination
- 4 Transport Service Classes

OSI Layer 4 functionality

SS7 connection setup using SCCP

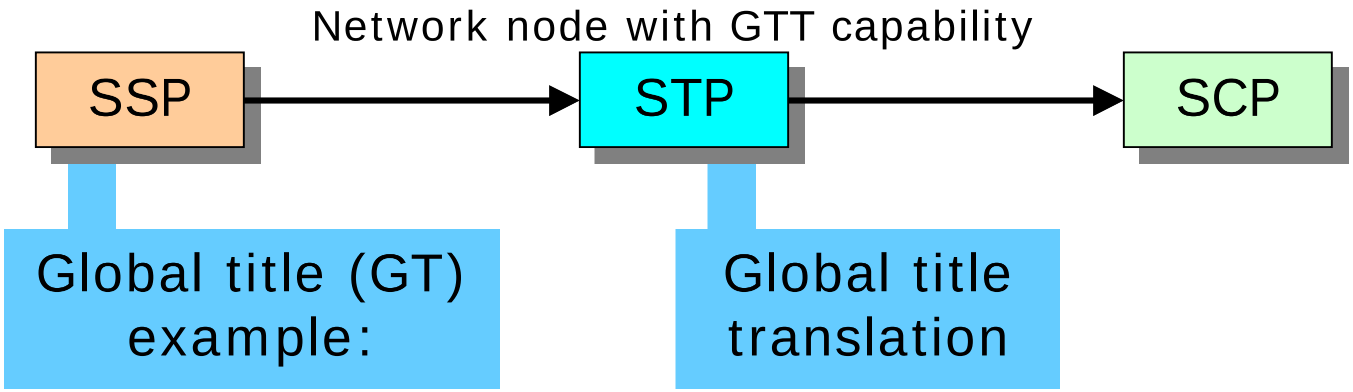
Signalling connection, *not* circuit switched connection (= call),

”setup” => several higher level signalling transactions over the same connection possible



Global title translation (GTT)

Global title translation (GTT) is required when the originating exchange (SSP) knows the ”global title” instead of the point code of the database (SCP).



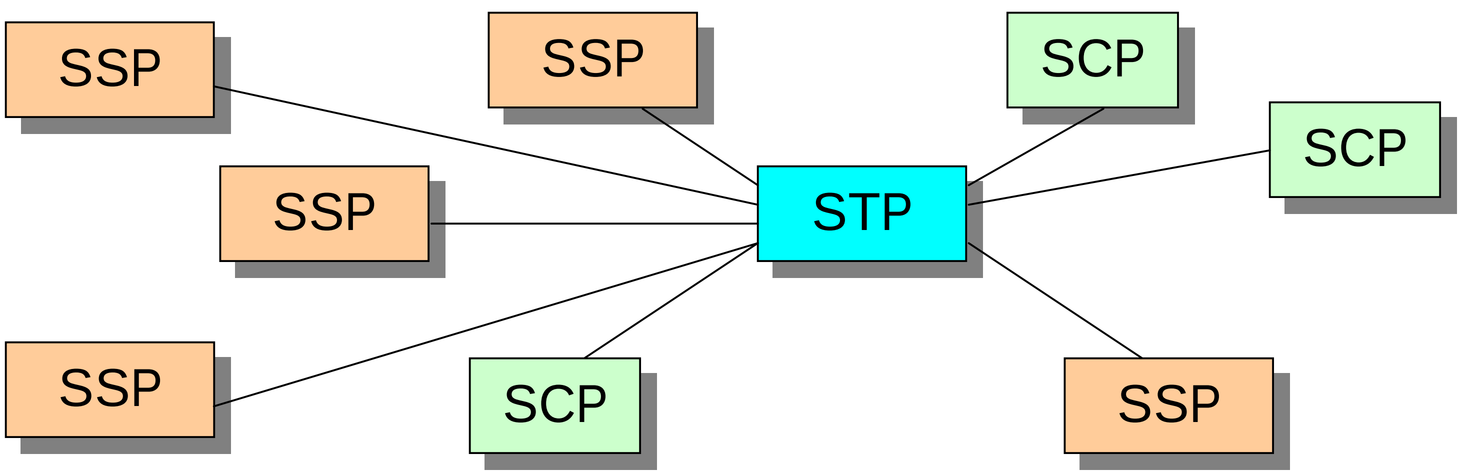
Find SCP using GT (0800 number)

Change GT (0800 number) into DPC + SSN

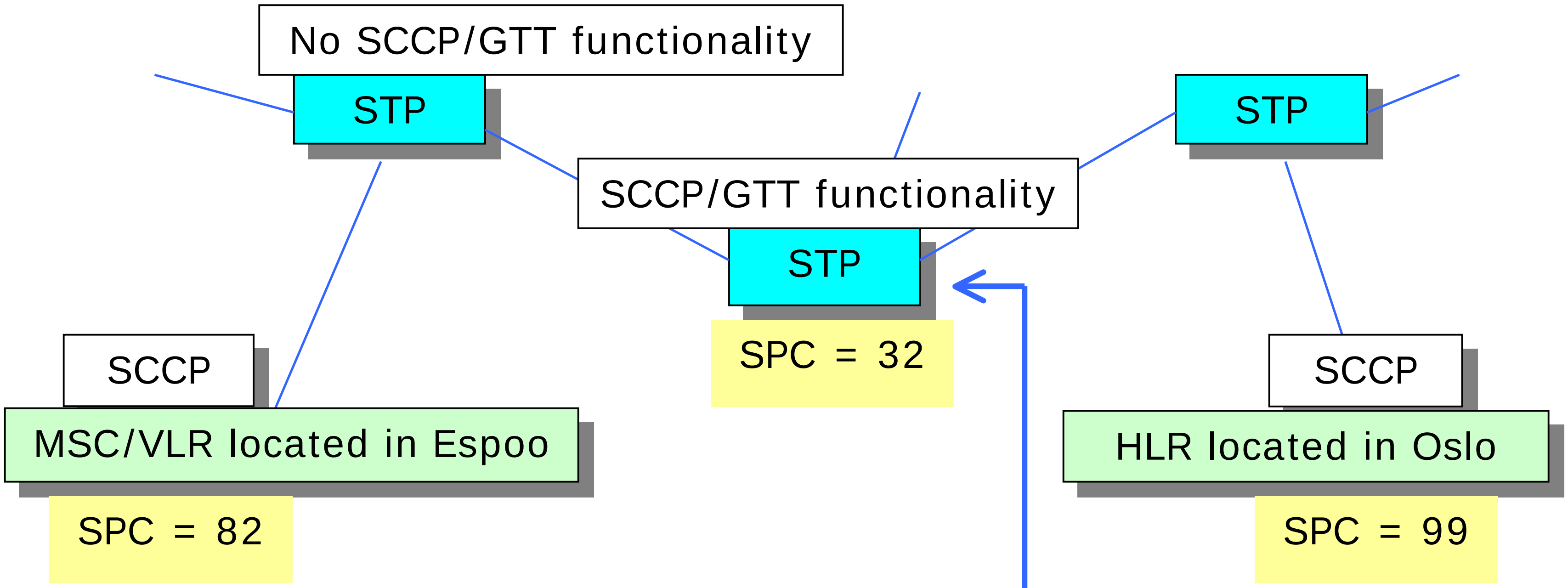
Why GTT in STP network node?

Global title translation (GTT) is usually done in an STP.

Advantage: Advanced routing functionality (= GTT) needed only in a few STPs with large packet handling capacity, instead of many exchanges.



Example: SCCP connection with GTT



Outgoing message:
OPC = 82 DPC = 32
SCCP: Global title (IMSI)

Processing in STP:
Received message is given to SCCP for GTT.
SCCP finds the DPC of the HLR: DPC = 99

Four classes of service in SCCP

Class 0: Basic connectionless class. Each information block (SCCP message) is transmitted from one SCCP user to another SCCP user independently.

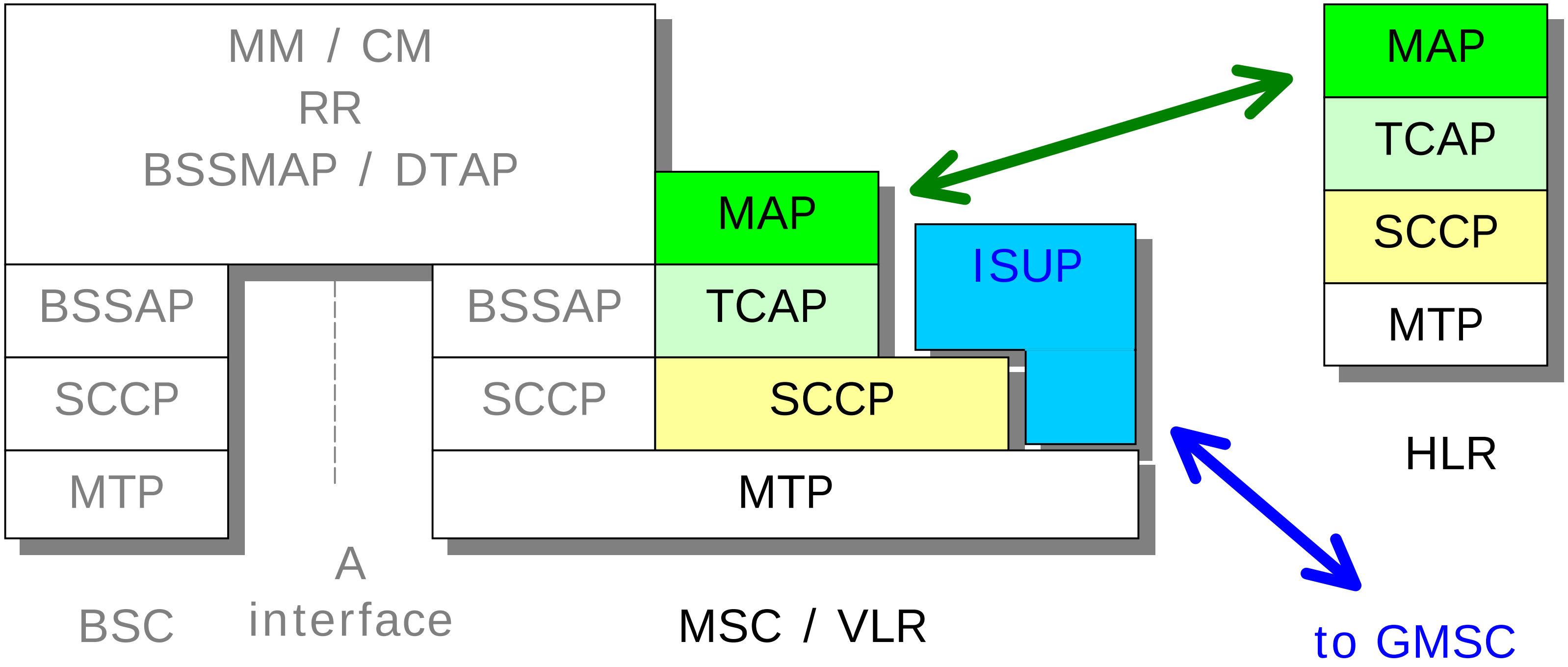
Class 1: Sequenced (MTP) connectionless class. All messages use the same SLS code.

Class 2: Basic connection-oriented class. Virtual connections are set-up and released + using same SLS code + segmentation & reassembly (SAR)

Class 3: Flow-control connection-oriented class. VC control + same SLS codes + SAR + flow control

Example: Signalling in GSM core network

ISUP for signalling between exchanges (MSC, GMSC)
MAP for signalling to/from databases (VLR, HLR, AuC, EIR)



IN

Intelligent Network

- basic concept
- technology
- IN services

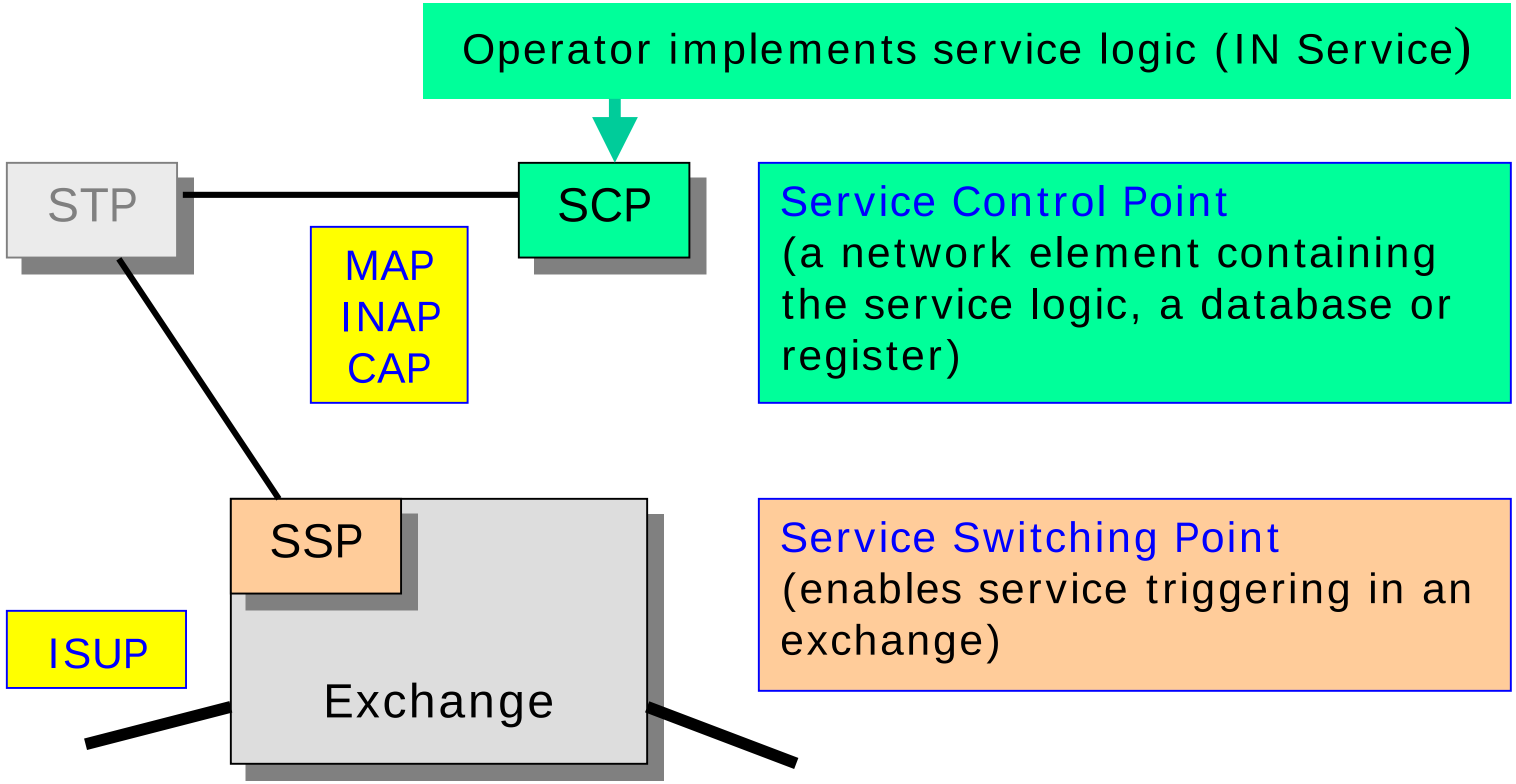
Intelligent Network (IN) Concept

The intelligent network concept: intelligence is taken out of exchanges and placed in computer nodes that are distributed throughout the network.

Intelligence => access to various databases

This provides the network operator with the means to develop and control services more efficiently. New capabilities can be rapidly introduced into the network. Once introduced, services are easily customized to meet individual customer's needs.

Intelligent Network (IN) Concept



IN service subscriber and customer

In a typical IN service scenario, the network operator or a 3rd party service provider implements the service for one or several subscribers, after which customers can use the service.

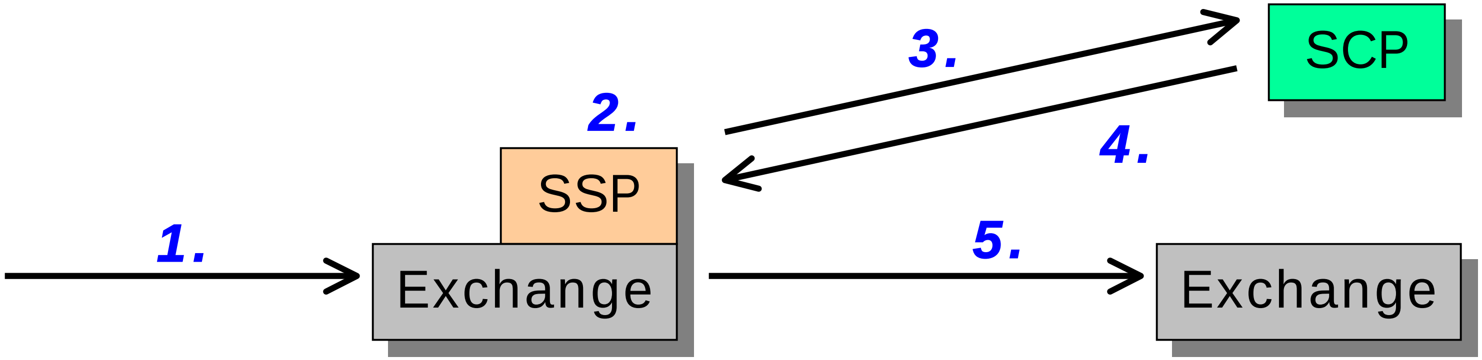
Service subscriber = company offering the service (e.g. the 0800 number that anybody can call)

Customers = those who use the service (e.g. those who call the 0800 number)

Confusion possible:

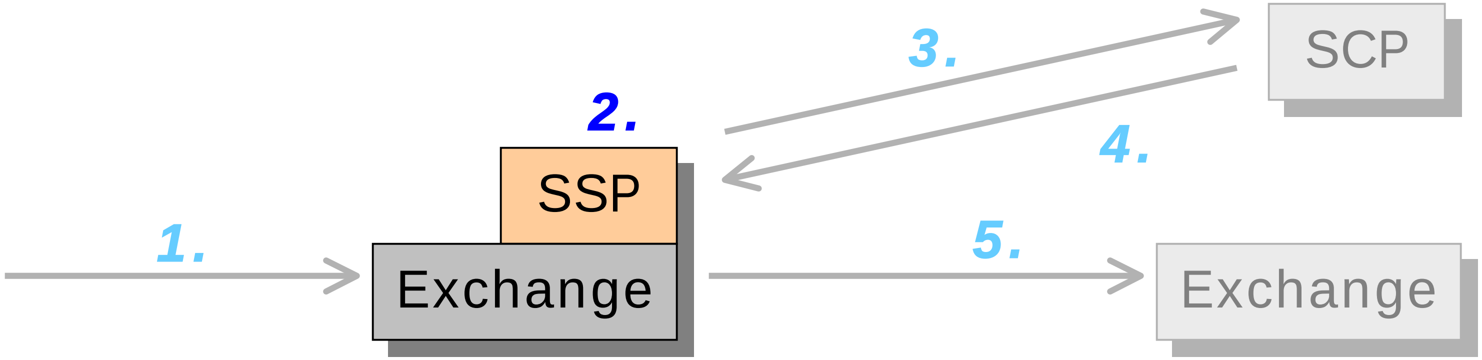
IN service subscriber $\neq$ PSTN subscriber

Typical call-related IN procedure (1)



1. *Call routing proceeds up to Exchange*
2. *Trigger activated in Basic Call State Model at SSP*
3. *SSP requests information from SCP (database)*
4. *SCP provides information*
5. *Call routing continues (routing to next exchange)*

Typical call-related IN procedure (2)

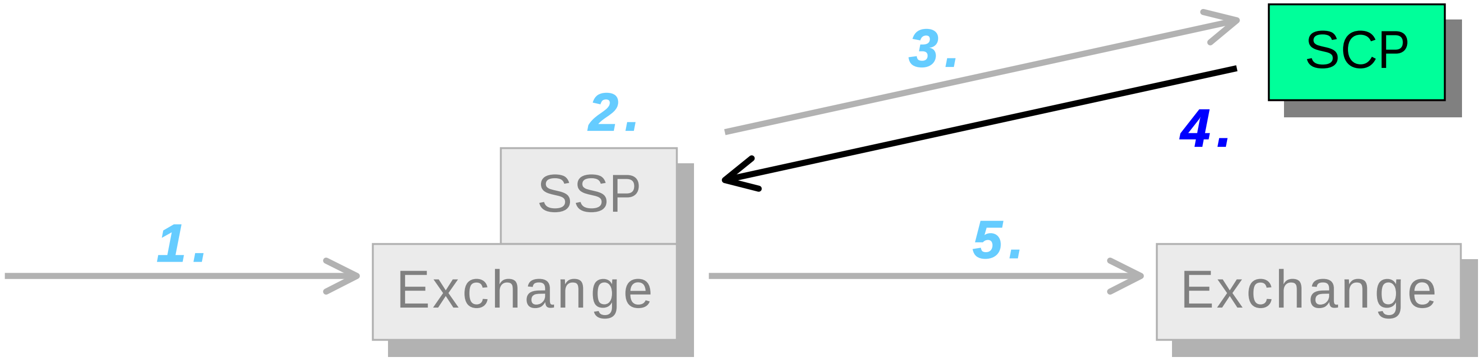


2. *Trigger activated in Basic Call State Model at SSP*

Typical triggers:

Called number (or part of number)
Destination busy
Caller dos not answer in predefined time

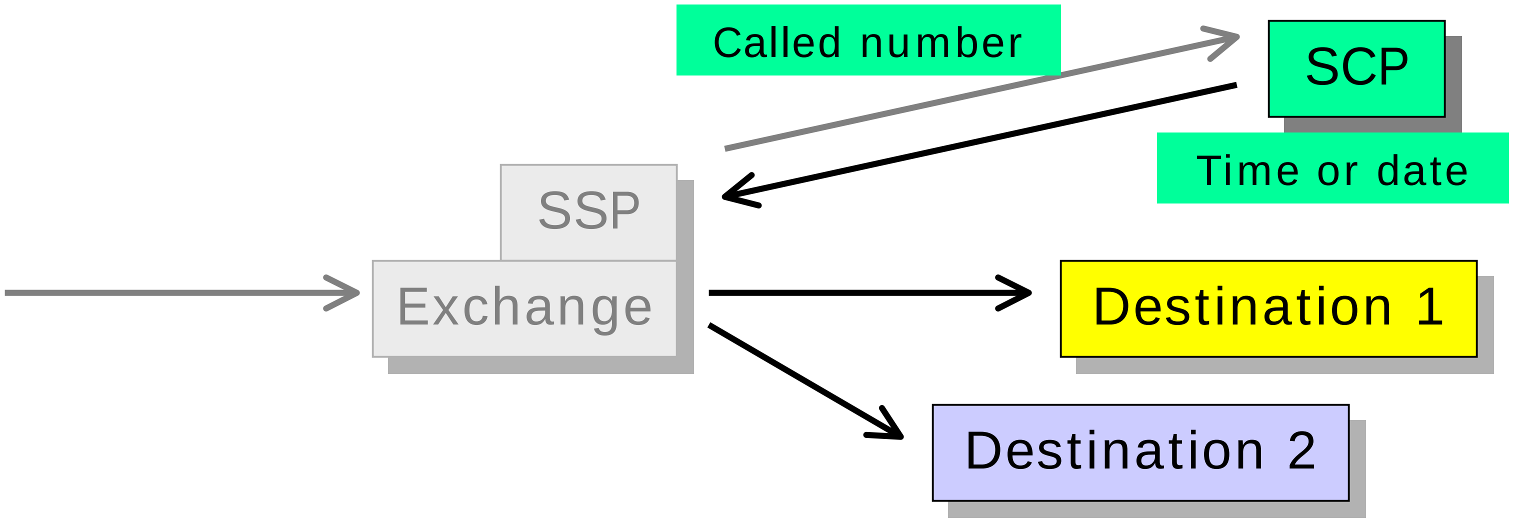
Typical call-related IN procedure (3)



4. *SCP provides information*

Example: Number translation in SCP *SSP sends 800 number (0800 1234)* *SCP translates into "real" number which is used for routing the call (+358 9 1234567)*	translation may be based on several variables

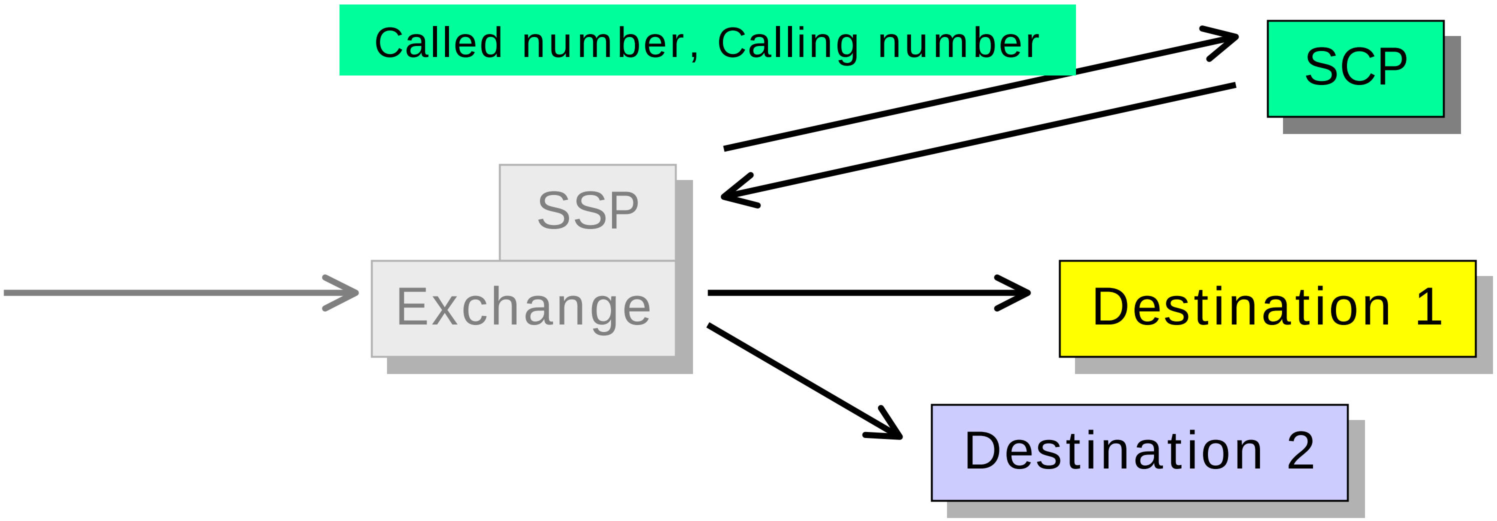
Examples of how SCP can affect call (1)



SCP decides the destination of the call depending on the calling time or date:

9.00 - 17.00 => Destination 1
17.00 - 9.00 => Destination 2

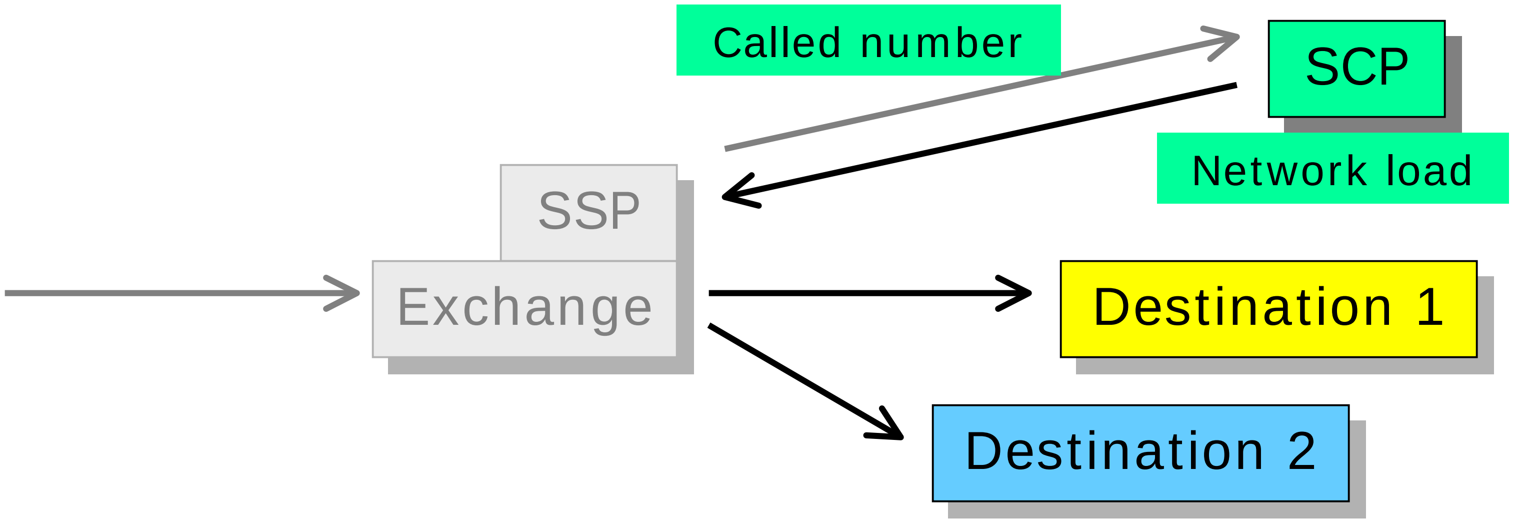
Examples of how SCP can affect call (2)



SCP decides the destination of the call depending on the location of calling user:

Calling user in southern Finland => Destination 1
Calling user in northern Finland => Destination 2

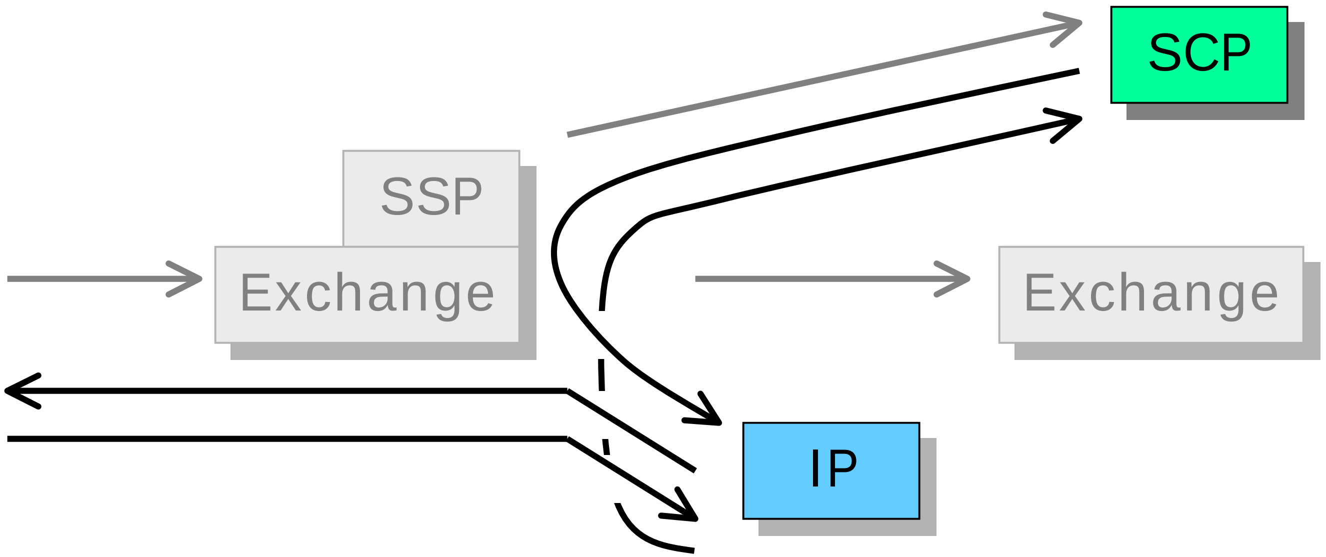
Examples of how SCP can affect call (3)



SCP decides the destination of the call depending on the traffic load in the network:

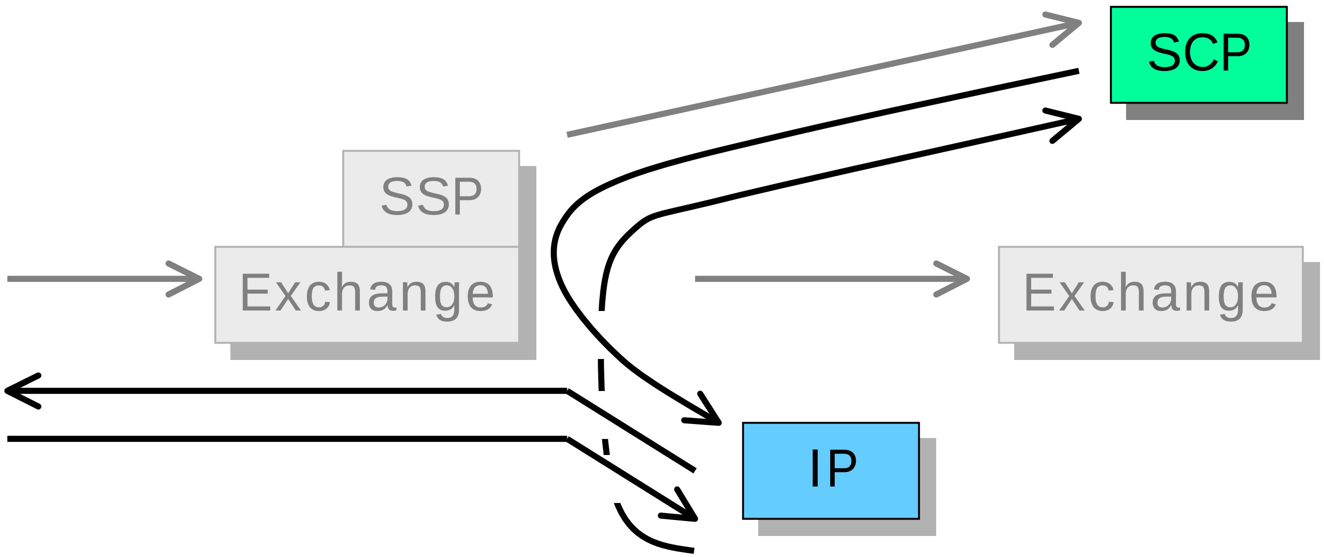
Traffic load situation 1 => Destination 1
Traffic load situation 2 => Destination 2

Additional IN features (1)



Intelligent Peripheral (IP) can (a) send announcements to the user (usually: calling user) and (b) receive DTMF digits from the user. IP is not a database; connection to exchange not via SS7, instead via digital TDM channels.

Additional IN features (2)

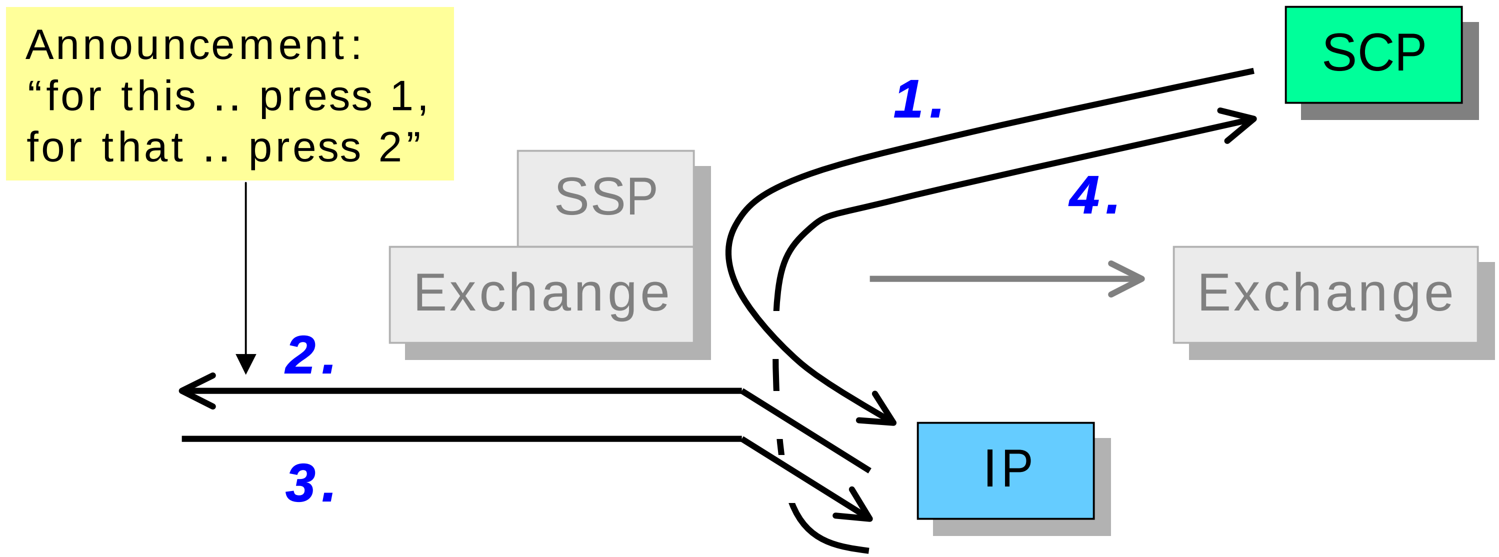


Typical applications:

1) whenever services need user interaction

2) user authentication

User interaction in IN service



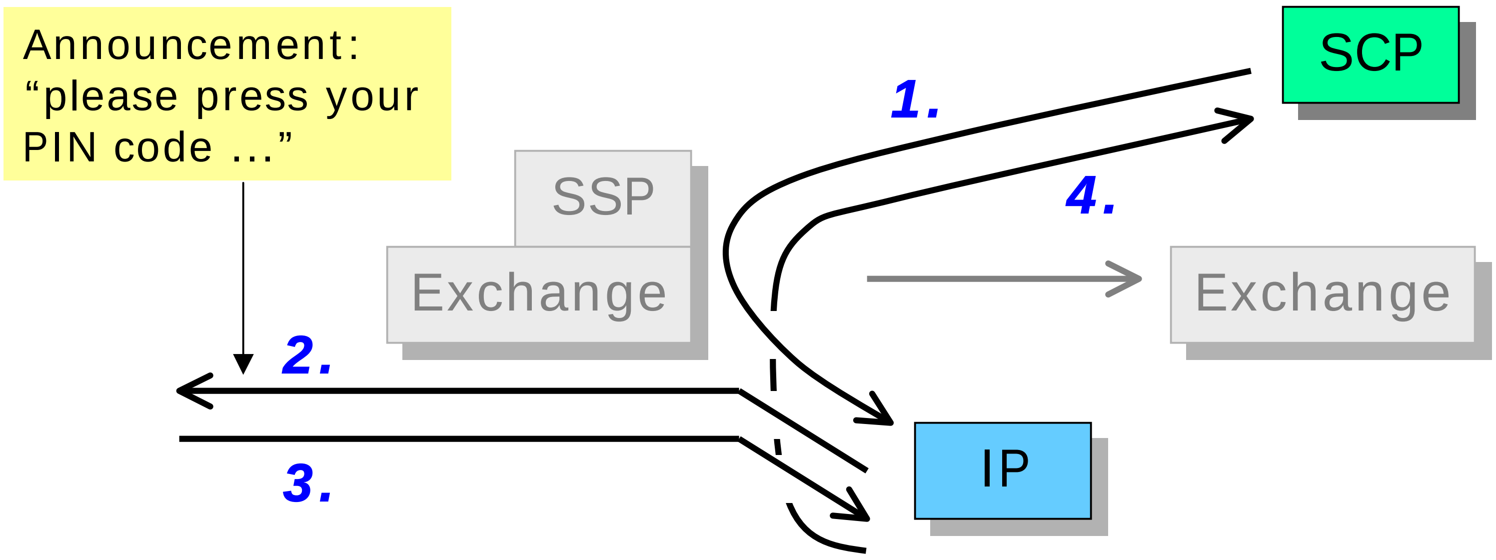
1. *SCP orders IP to select and send announcement*

2. *IP sends announcement to calling user*

3. *User replies by sending DTMF number(s) to IP*

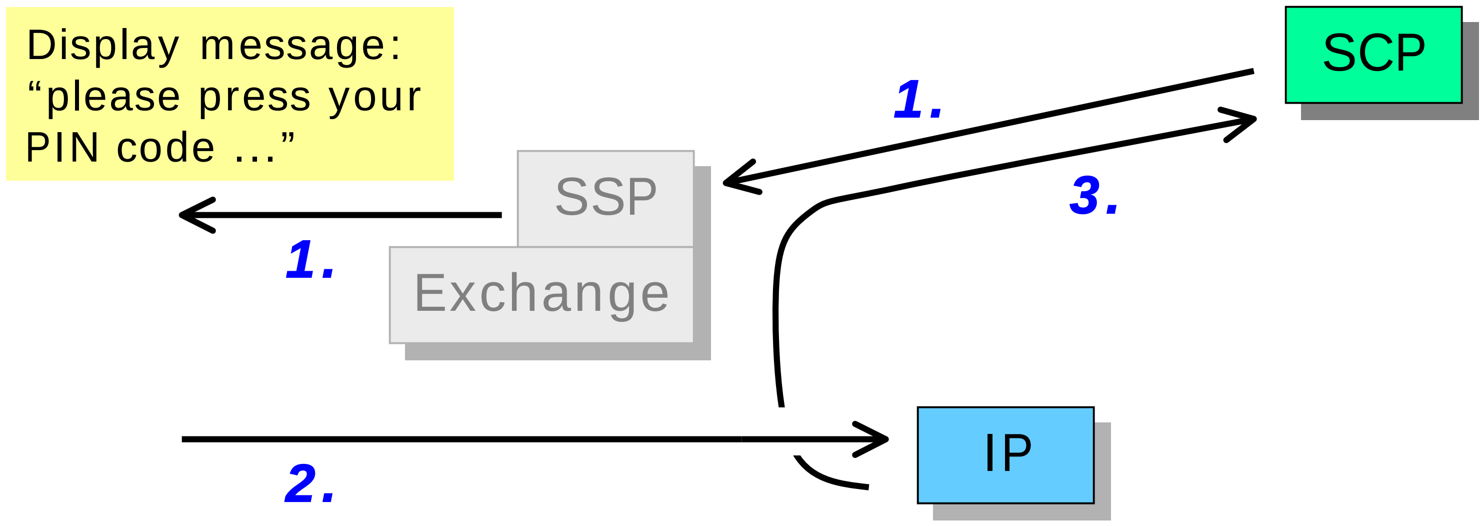
4. *IP sends number information to SCP*

User authentication (1)



1. *SCP orders IP to select and send announcement*
2. *IP sends announcement to calling user*
3. *User sends authentication code (in DTMF form) to IP*
4. *IP sends authentication code to SCP*

User authentication (2)



When connected to the network via a digital subscriber line, the calling user can be notified with a digital message ("please press your PIN code ...") instead of having to use the corresponding voice announcement.

IN services

A large number of IN services can be implemented by combining different “building blocks”:

- called number translation (at SCP)
- routing decision based on calling number, time, date, called user busy, called user alerting timeout, network load ...
- announcements (from IP) or user notification (<= ISDN user signalling)
- DTMF number reception (at IP) and analysis (at SCP)
- customised charging (at exchanges)

IN service examples

“Traditional” IN services:

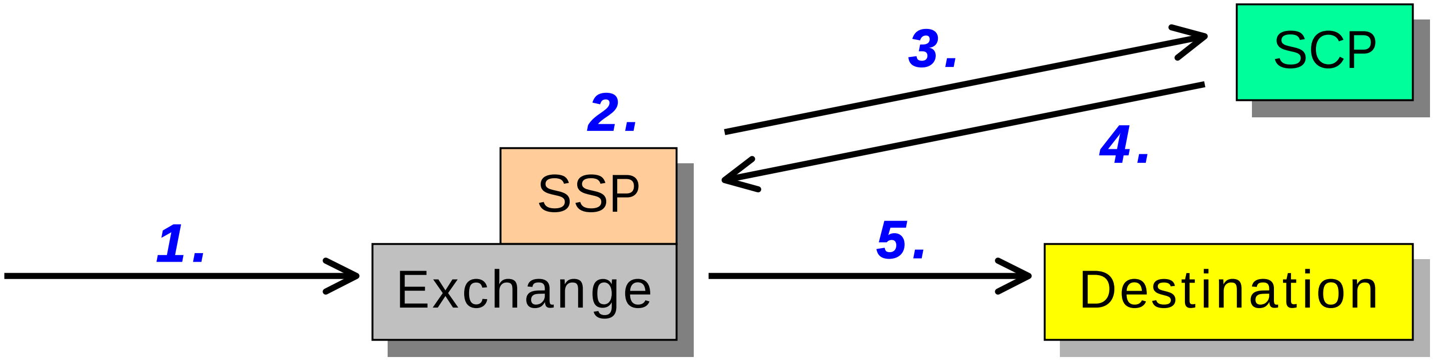
- Freephone / customised charging schemes
- Virtual Privat Network (VPN)
- Number portability
- Televoting

“IN” in mobile networks:

- Mobility management (HLR, VLR = databases)
- Security management (Authentication ...)
- IN in mobile networks ≈ CAMEL (Customised Applications for Mobile networks Enhanced Logic)

Freephone (800) service

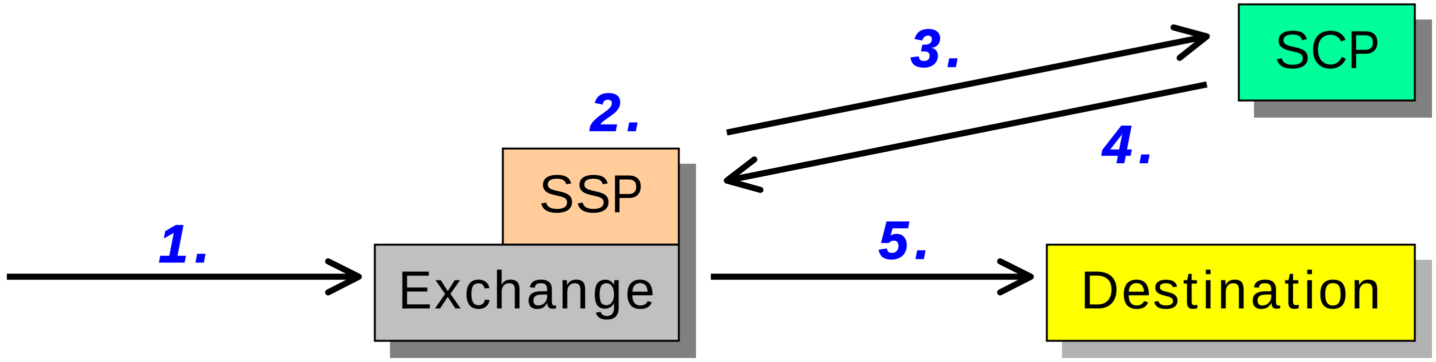
User calls 0800 76543. SSP sends this number to SCP which after number analysis sends back to SSP the real destination address (09 1234567) and call can be routed to the destination. Called party is charged.



Charging: destination (service subscriber) pays the bill

Premium rate service

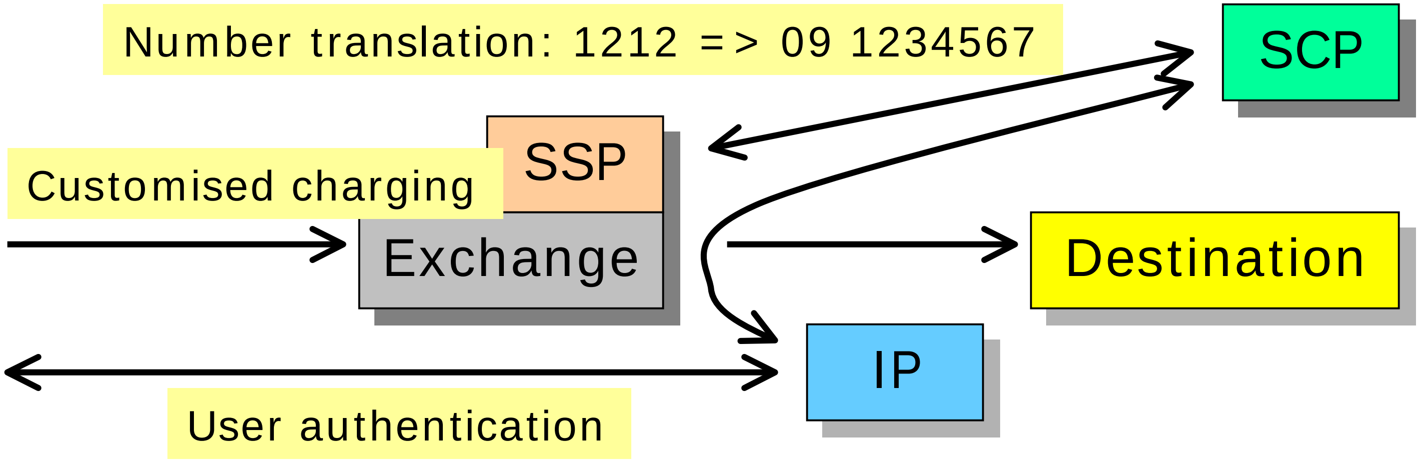
User calls 0200 34343. SSP sends this number to SCP which after number analysis sends back to SSP the real destination address (09 676567) and call can be routed to the destination. Calling party is charged.



Charging: calling user (customer) pays the bill. Both service subscriber and service provider / network operator make profit.

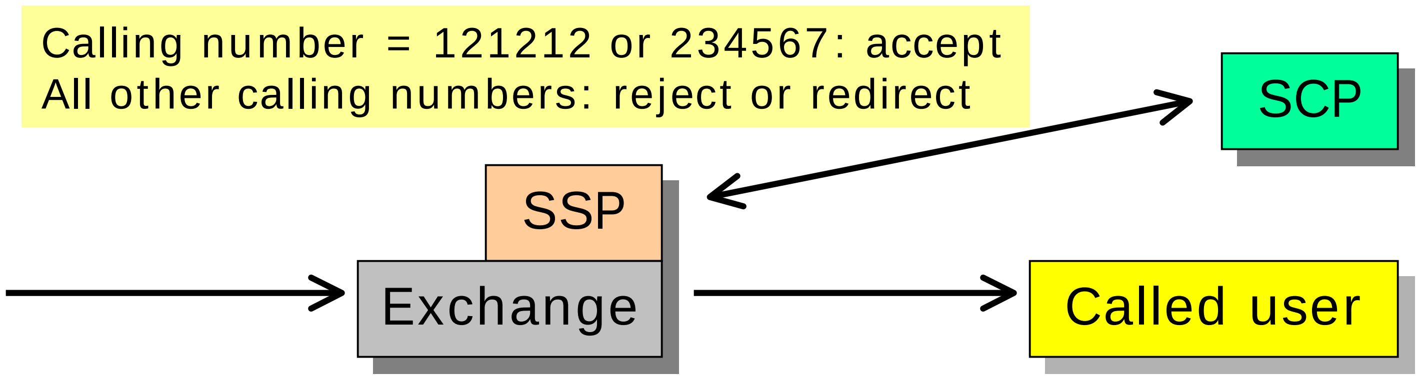
Virtual private network (VPN) service

A VPN provides corporate customers with a private number plan within the PSTN. The customer dials a private (short) number instead of the complete public number in order to contact another user within the VPN. User authentication is usually required.



Screening of incoming calls

This is an example of an IN service related to the call destination end. Alert called user only if calling number is 121212 or 234567, otherwise do something else (e.g. reject call or redirect call to another destination).



Further information on SS7

Tutorial:

Modarressi, Skoog: ”SS7: a tutorial”, *IEEE Comm. Magazine*, July 1990

Books:

Bhatnagar: *Engineering networks for synchronization, CCS7, and ISDN*, IEEE Press, 1997

Van Bosse: *Signaling in telecommunication networks*, Wiley, 1998

SS7 tutorial on the web:

www.iec.org/online/tutorials/ss7