

ISDN User Part - ISUP

✓ ISUP - ISDN User Part

- › ISUP/TUP brief comparison
- › ISUP additional features

✓ Interworking of signaling systems

ISUP is an international and national network signaling system for

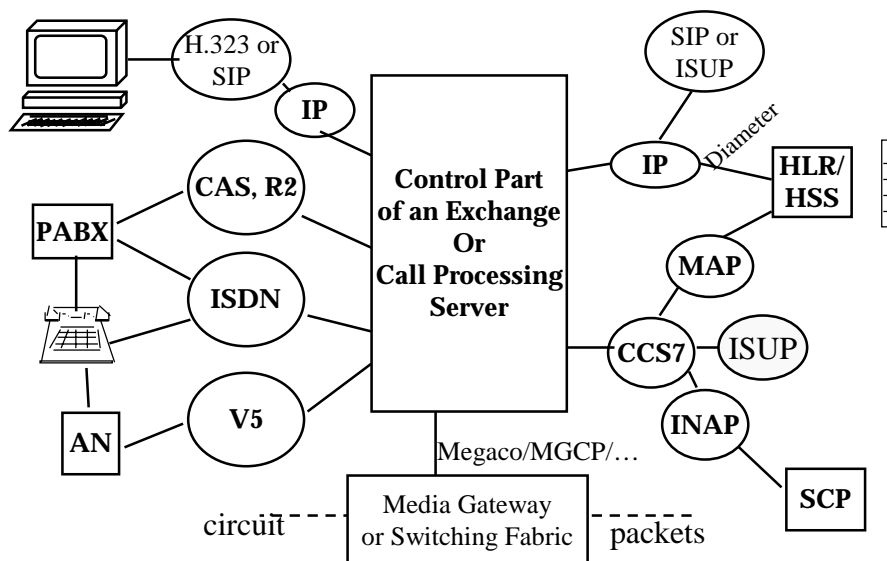
- *call setup,*
- *supervision and*
- *release.*

*In addition it supports a wide range of ISDN supplementary services.
Used also in GSM and 3G.*

Separate versions for

- *International ISUP*
- *National ISUP in many countries (carry legacy of older systems)*

Summary of course scope



ISUP - ISDN User Part milestones

- TUP was specified before DSS1 ISDN user signaling during 1980's.
- ISUP2 specification was released after DSS 1.
- ISUP2 deployment in Finland started 1994.
- Changeover from TUP to ISUP is well under way internationally.
- In network core development path is CAS -> TUP ->ISUP.
- If TUP is already deployed, changing to ISUP is relatively easy by a software upgrade in exchanges, because MTP-infra is already in place.
- Development under way in ISUP:
 - ETSI is adding charging information messages into ISUP.
 - IETF (SIGTRAN) is specifying ISUP over IP

Why does the ISDN need ISUP instead of TUP?

Limitations of TUP compared to ISUP:



DSS 1 terminal compatibility information can not be transported in TUP,



User-to-User information is not specified in TUP signaling messages



ISDN Suspend/Resume is not supported in TUP,



TUP does not support all (Euro-)ISDN supplementary services - call waiting, call hold ...



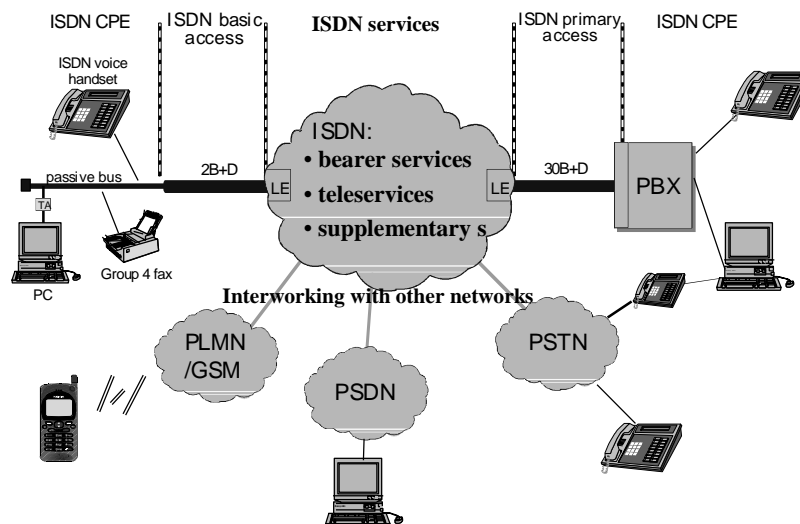
In TUP release is non-symmetric / in ISDN it is symmetric

Bearer services supported by ISUP are

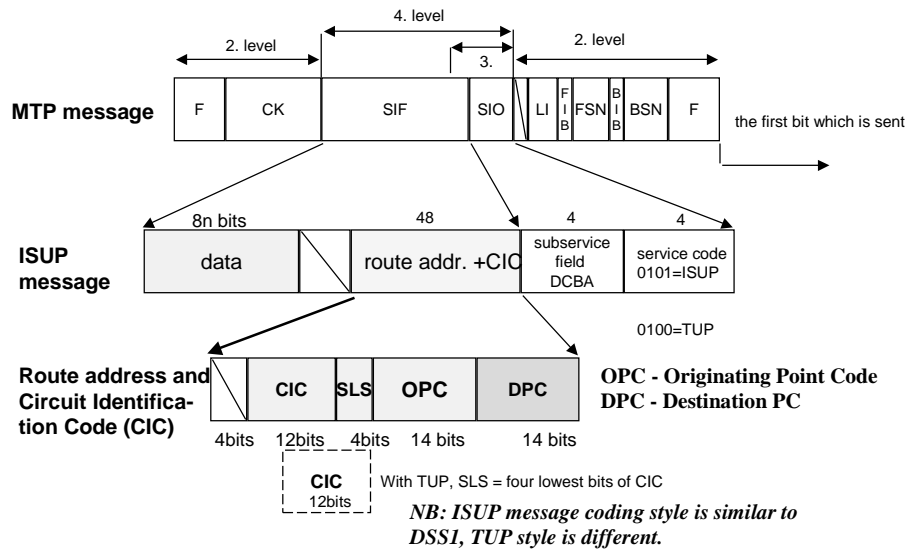
- ✓ speech
- ✓ 64 kbit/s unrestricted (= transparent 64kbit/s)
- ✓ 3.1 and 7 kHz audio
- ✓ alternate speech / 64 kbit/s unrestricted
- ✓ alternate 64 kbit/s unrestricted / speech
- 2 x 64 kbit/s unrestricted
- 384 kbit/s unrestricted
- 1536 kbit/s unrestricted
- 1920 kbit/s unrestricted

In the future ISUP offers extensions compared to TUP, but competition from the Internet is tough!

ISDN basic structures



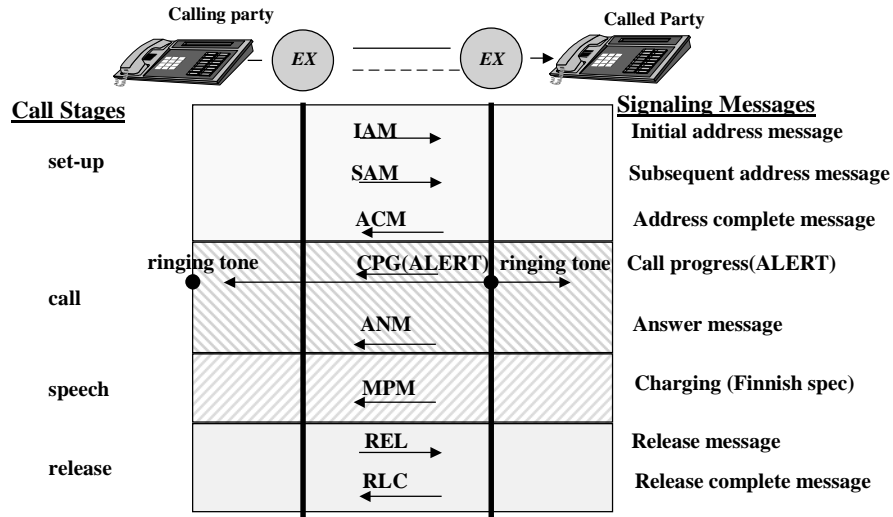
Basic structure of an ISUP message



Call identification is based on a compulsory CIC and an optional (logical) call reference

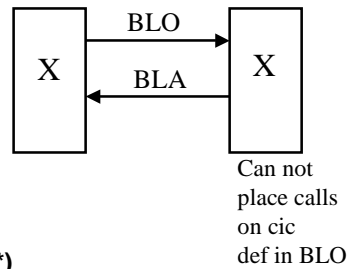
- ✓ Call reference is recommended only for national use.
- ✓ n x 64kbit/s connections are always built using consecutive timeslots, thus one CIC is enough.
- ✓ n x 64kbit/s -connection is identified using the smallest CIC among the time-slots.
- ✓ CIC binds the call channel and signaling together. One can not exist without the other --> one result is that in IN a special standardised Basic Call State Model is needed. The BCSM is used track the state of the resources in an SSP (service switching point) while an SCP (service control point) processes additional features.
- ✓ Binding to CIC is also an issue when Interworking with IP Telephony systems.

A successful ISUP call (calling subscriber initiates release)



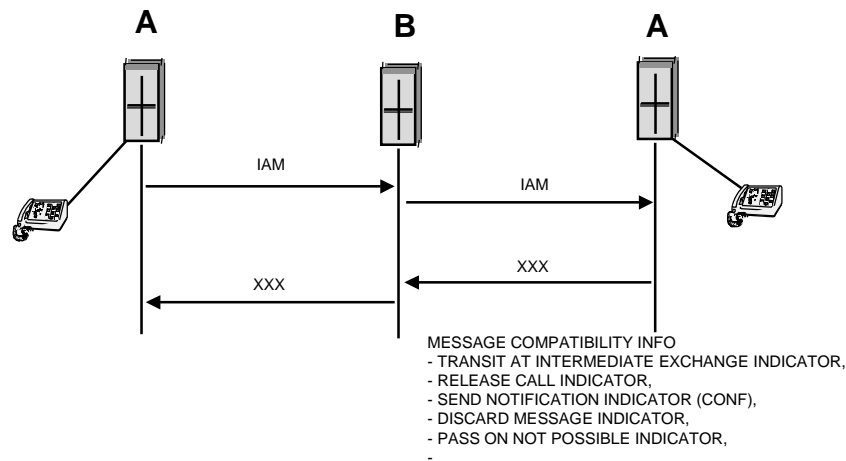
ISUP circuit supervision messages --> circuits and 2M connections can be taken into use and from use in a managed way.

- RSC Reset circuit
- BLO Blocking
- BLA Blocking acknowledgement
- UBL Unblocking
- UBA Unblocking acknowledgement
- EHL End-of-hold (*)
- EHA End-of-hold acknowledgement (*)
- OLM Overload (*)
- UCIC Unequipped circuit identification code (*)



(*) For national use

Forward COMPATIBILITY is ensured from 1992 release



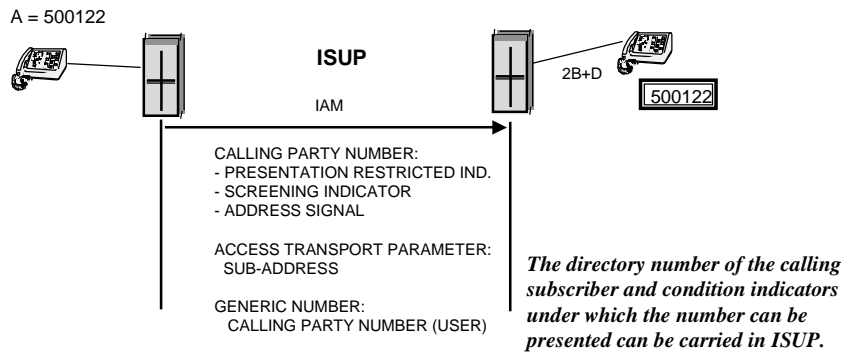
ISUP message coding supports ISUP software upgrades - old and new version can talk to each other!

Version compatibility rules: the following should not be changed:

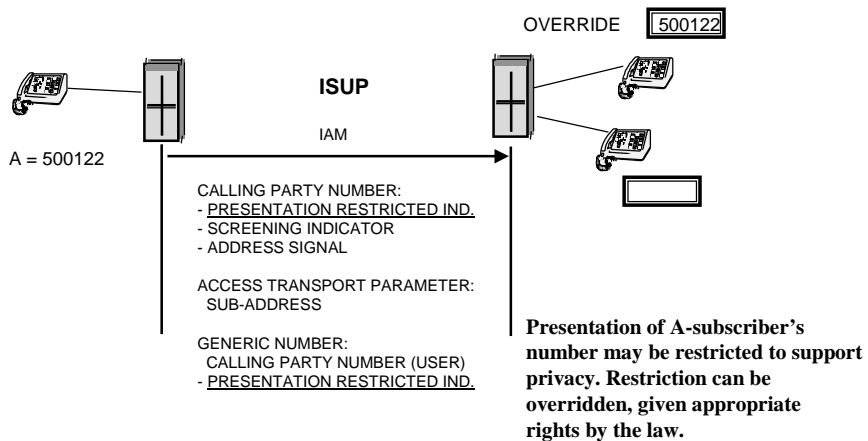
- ✓ Protocol procedures, messages, information elements, coding, except to correct an error in the protocol.
 - ✓ Semantics of existing info elements.
 - ✓ Formatting and coding rules
 - ✓ Adding new parameters into mandatory part of Messages, Optional part can be extended.
 - ✓ Order of information items in an Information Element of variable length, new items can be added to the end of the IE.
 - ✓ Information item value = "all 0" == non-significant value.
- + **Fall-back and other compatibility procedures.**

Rules apply from ISUP'92.

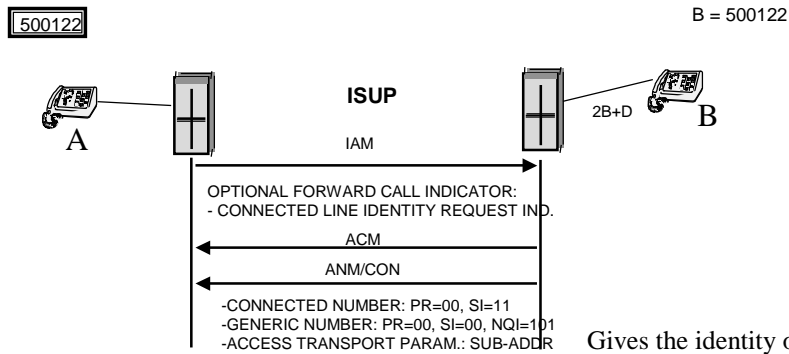
Calling Line Identification Presentation - CLIP - is a supplementary service supported by ISUP



Calling Line Identification Presentation Restriction - CLIR - is a pair to CLIP



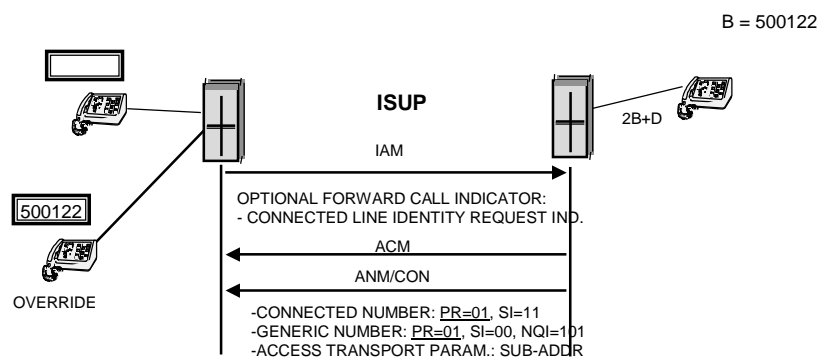
Connected Line Identification Presentation -COLP - is a mirror image to CLIP



PR = 00, presentation allowed
SI = 11, network provided
SI = 00, user provided not verified
NQI = 101, additional connected nbr.

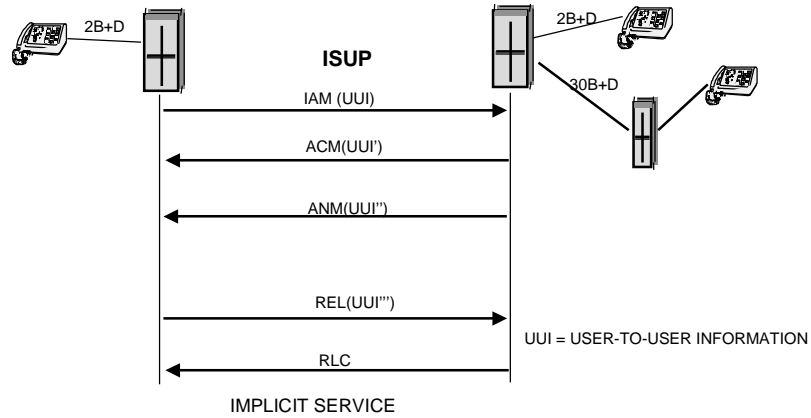
Gives the identity of B to A. NB: due to supplementary services B may be different from dialled nr

Connected Line Identification Presentation Restriction - COLR - is the pair to COLP

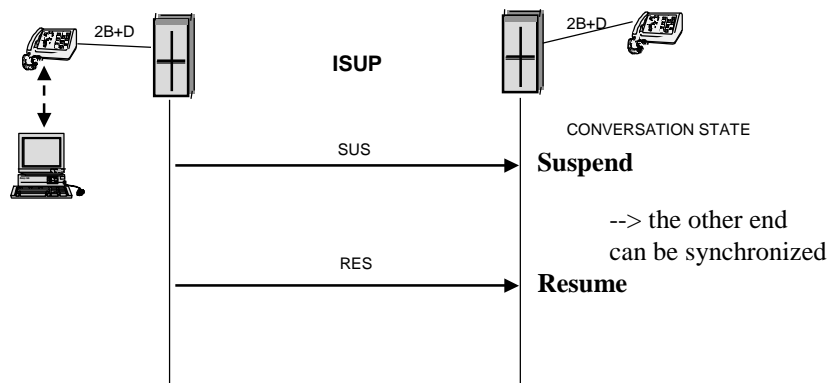


PR = 01, presentation restricted
SI = 11, network provided
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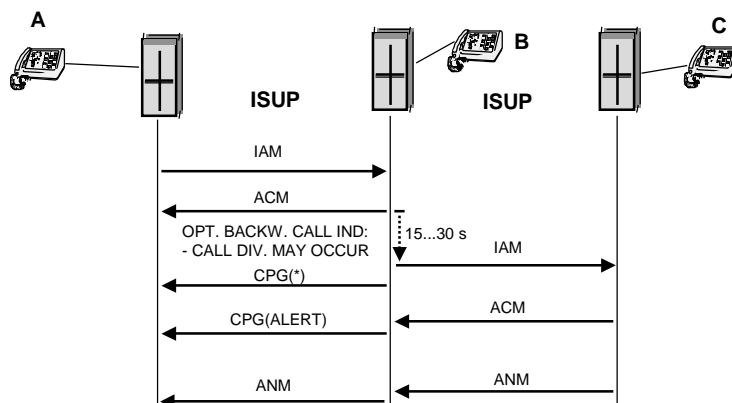
User to User Signalling 1 - UUS1 - allows transporting user provided information over CCS7 network



Terminal Portability (TP) - ISDN allows interrupting a call and resuming it even from a different phone or phone line..

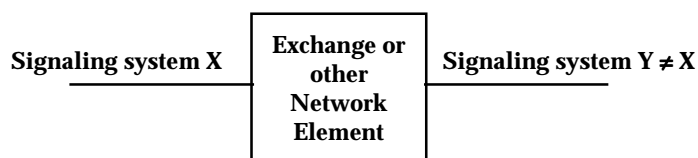


Call Forwarding No Reply - CFNR - automatically forwards an incoming call to C-number



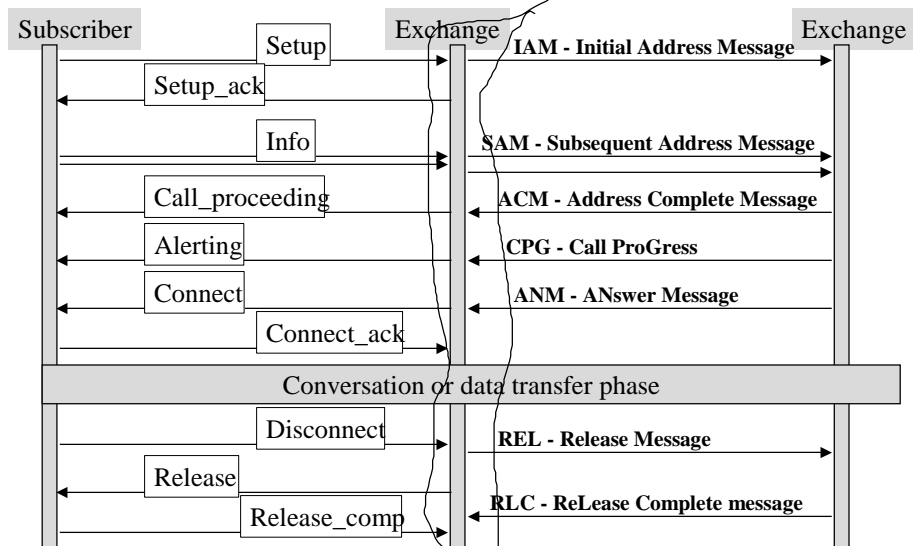
(*) CALL DIV. INFO, REDIR. NBR., GENERIC NOTIF.

Signaling interworking occurs in an exchange if two legs of the call are managed using different signaling systems



- ✓ **Interworking of signalling systems**
- ✓ **Also we talk about signaling interworking if two peer exchanges are manufactured by different vendors (interworking of different implementations)**
- ✓ **cmp. compatibility**

DSS1 / ISUP -interworking



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Signaling Protocols

13 - 21

Each signaling system has its own set of signals of information elements -> in interworking almost always some info is lost.

- To ensure smooth interworking, functioning need to be carefully specified.
If we have n signaling systems, there are n^2 interworking cases!
- Standardization bodies use two methods for the specification of interworking:
 - *For Channel Associated signaling: event based FITE/BITE -method.*
 - *For message based signaling: layer oriented method.*

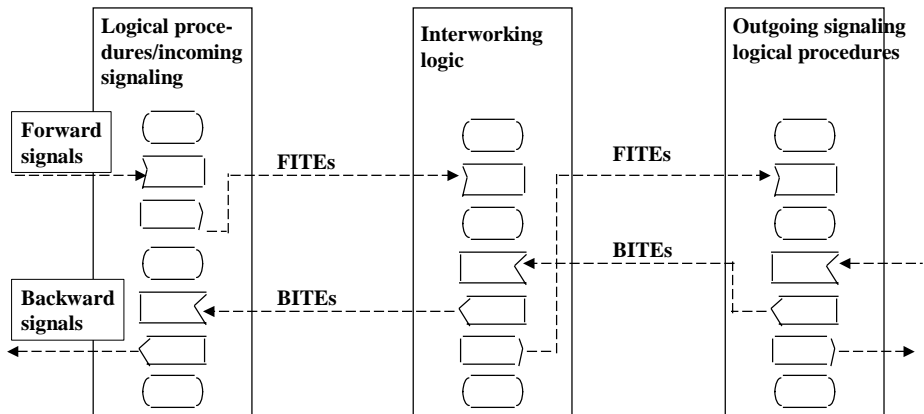
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Signaling Protocols

13 - 22

Event based interworking specification method

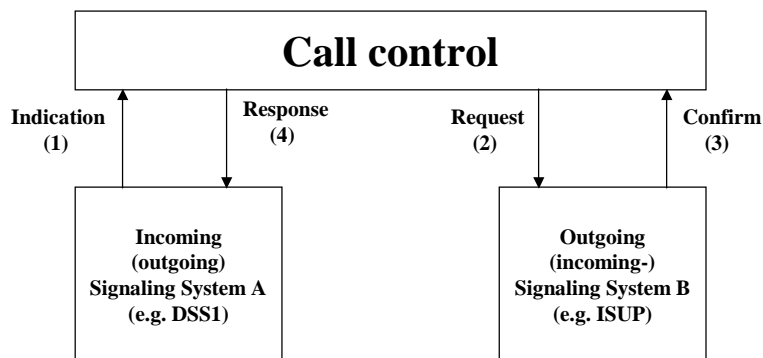
FITE - Forward Interworking Telephone Event } Between signaling systems.
 BITE - Backward Interworking Telephone Event }
 SPITE - Switching Processing Interface Telephone Event - internal to an exchange.



Incoming and outgoing signaling systems are analyzed only to the extent necessary for the specification of interworking. Logic is given using SDL.

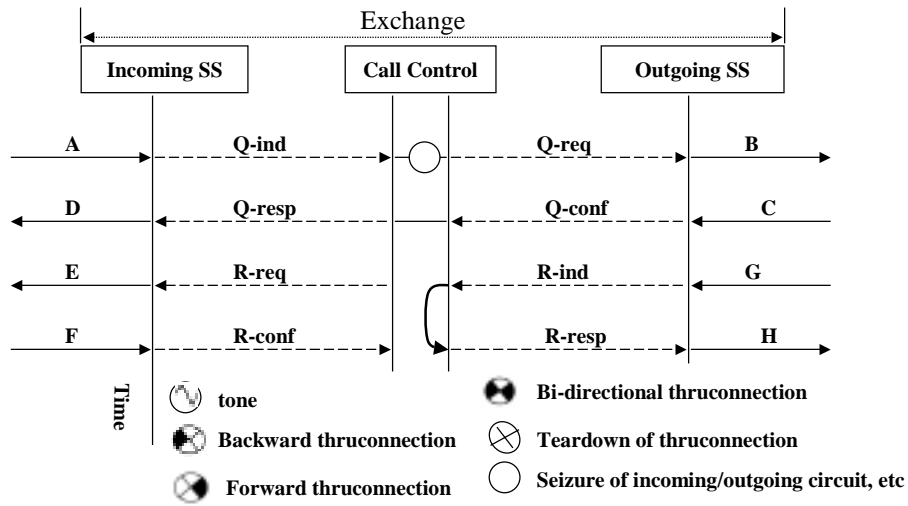
Layer oriented interworking specification

Primitives carry the information between layers



Numerot ilmaisevat primitiivien järjestystä

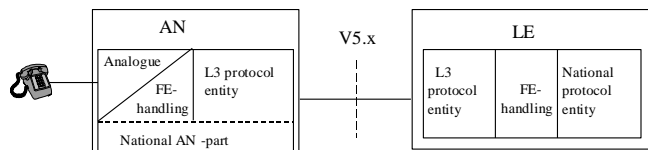
Signaling flow is described in more detail:



Because SDL is not used, specs is never complete -> vendors take care of the details .

In V5 - PSTN protocol, interworking is specified differently: partially in AN, partially in the exchange

PSTN protocol



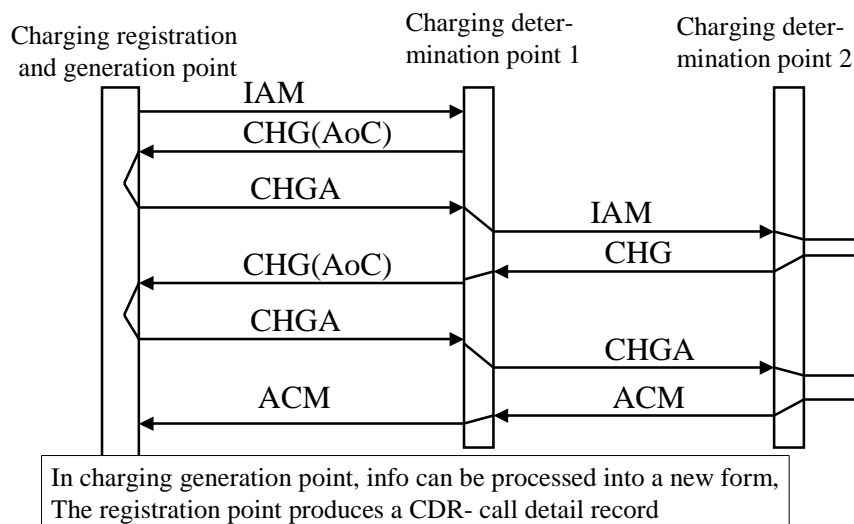
FE - function element primitives
 - primitives describe the state of the analogue circuits
 in AN or in LE describe the interworking with analogue subscriber line signaling

- AN takes care of:
- timing and duration of analogue signals
 - charge indication frequencies and voltages
 - ringing voltage
 - autonomous functions specified in national specs

Latest development of ISUP

1. ISDN charging protocol to transport tariff and billing info
 - The Finnish network has traditionally carried charging messages. In most other countries the originating exchange needs to know all tariffs in the world.
 - Reflect the difference between monopoly and competitive markets
 - ETSI is specifying messages between charging points to transport information about additional tariffs on a call by call basis:
 - Final tariff may be composed of many parts
 - tariff info is maintained by the party, who wants to earn the money.
 - Makes easier to apply dynamic tariffs.

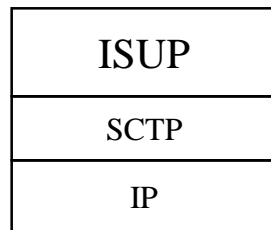
ISUP2 carries charging info



ISUP - more ongoing development

2. ISUP-over-IP for IP-telephony networks

- e.g. In Finland in pilot use although the specs is not ready
- SIGTRAN group in IETF



Signaling Control Transport Protocol
= transport protocol for e.g. signaling
modified from TCP

- SCTP/IP replaces MTP + (SCCP)
(SCCP =?, may survive.)