Signalling in telecommunication networks

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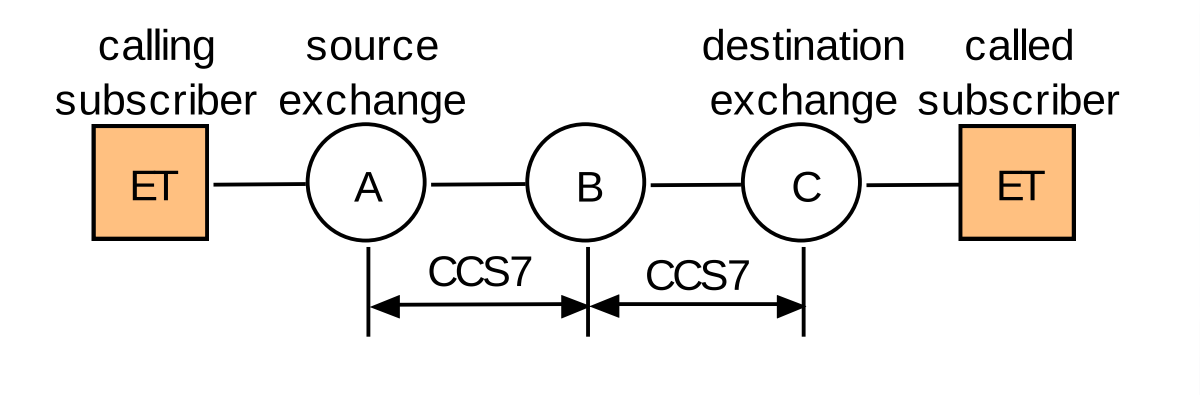
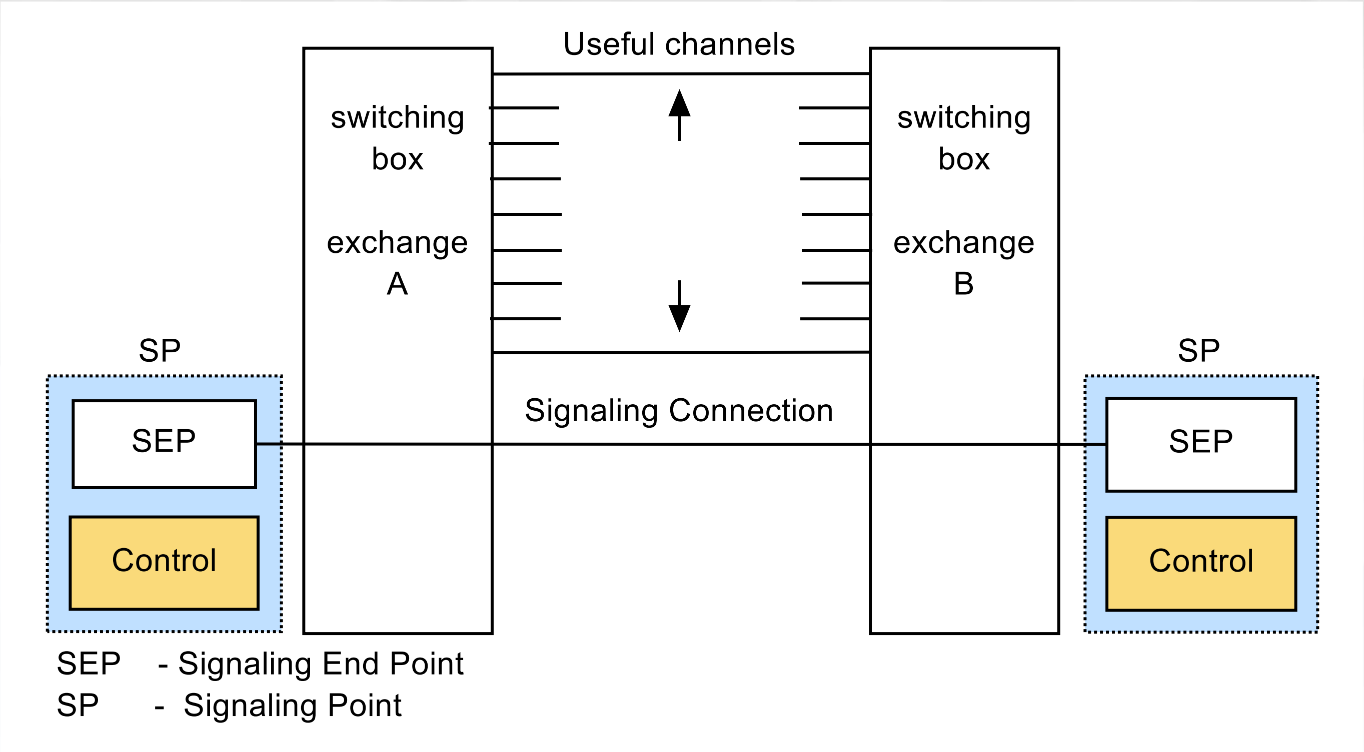
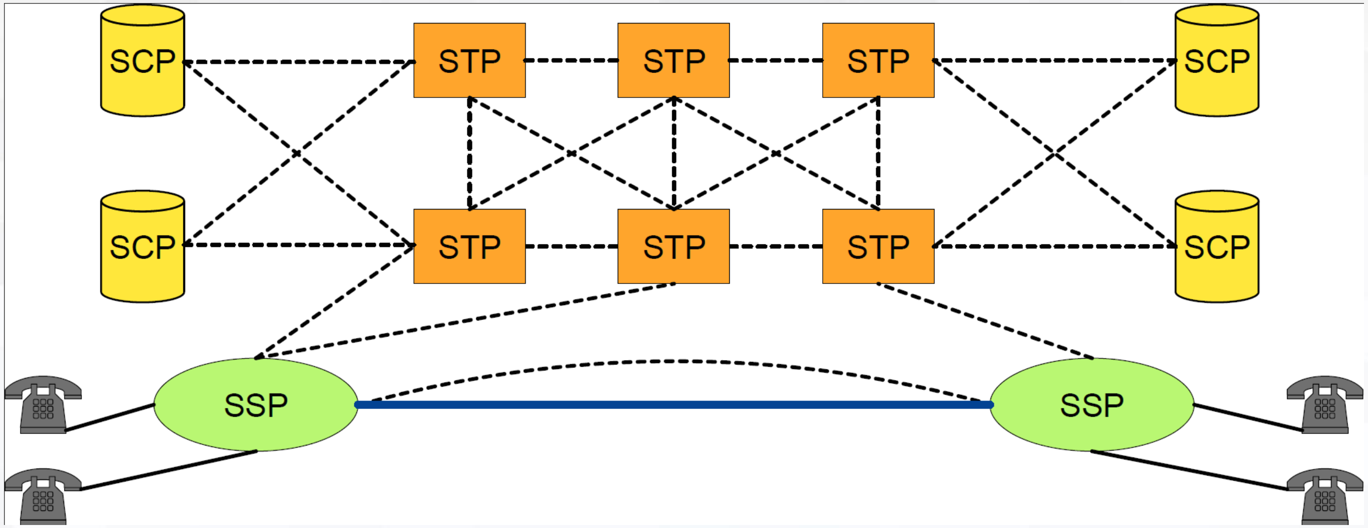
CCS7 (Common Channel Interoffice Signaling System No. 7)

- developed by ITU-T Q.700 - internationally standardized
- is used between the PSTN exchanges replacing in-band signaling
- big reliability
- for variable transmission media (metallic, optical, radio)
- provides call setup and terminate, network management, fault resolution and traffic management
- signaling information between switching systems (called signaling points) in the PSTN are carried on a special overlay network used exclusively for signaling
- the signaling points use routing information in the SS7 signals to transfer calls to their final destinations





- separate signaling connection is used for more useful channels
- **SP (Signaling Points)** are endpoints for signalling traffic
- **SEP (Signaling End Points)** – exchange signaling messages
- **Signalling Route** – bidirectional signalling channel (e.g. 1 timeslot in PCM)



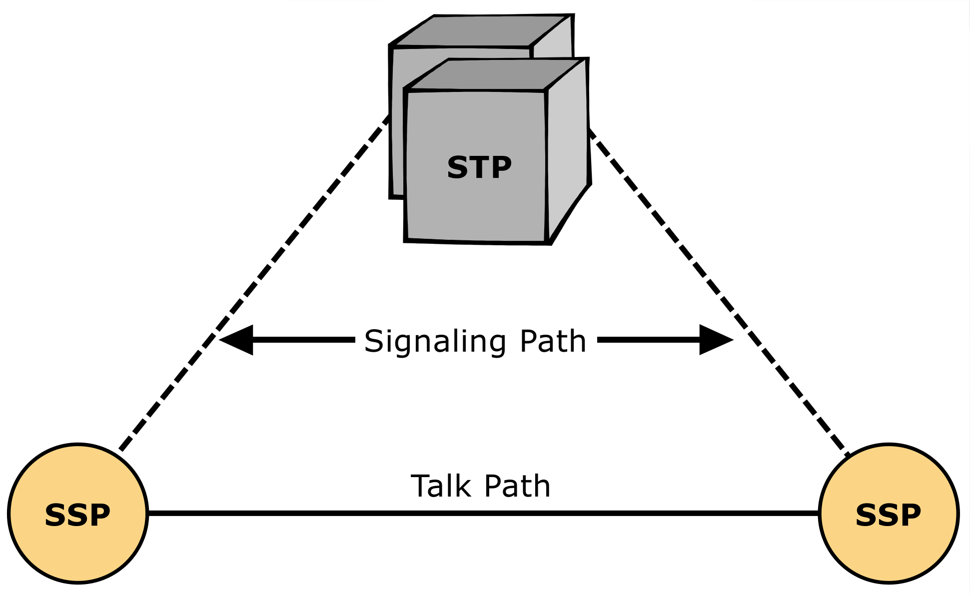
- **Signaling Transfer Points** (STPs) receive and route incoming signaling messages toward their destination based on destination address
- **Service Control Points** (SCPs) are databases that provide the necessary information for special call processing and routing
- **SSP (Signalling Switching Points)** - switches that originate or terminate calls
- SS7 network is packet based which transfer singnalling messages
- SS7 network is „overlayed“ to PSTN network

Modes of opperation

- **Associated**

- link is directly parallel with the voice path
- E1/T1
- channel number 32(24) is the associated out-of-band signaling channel for 31/23 talk channels

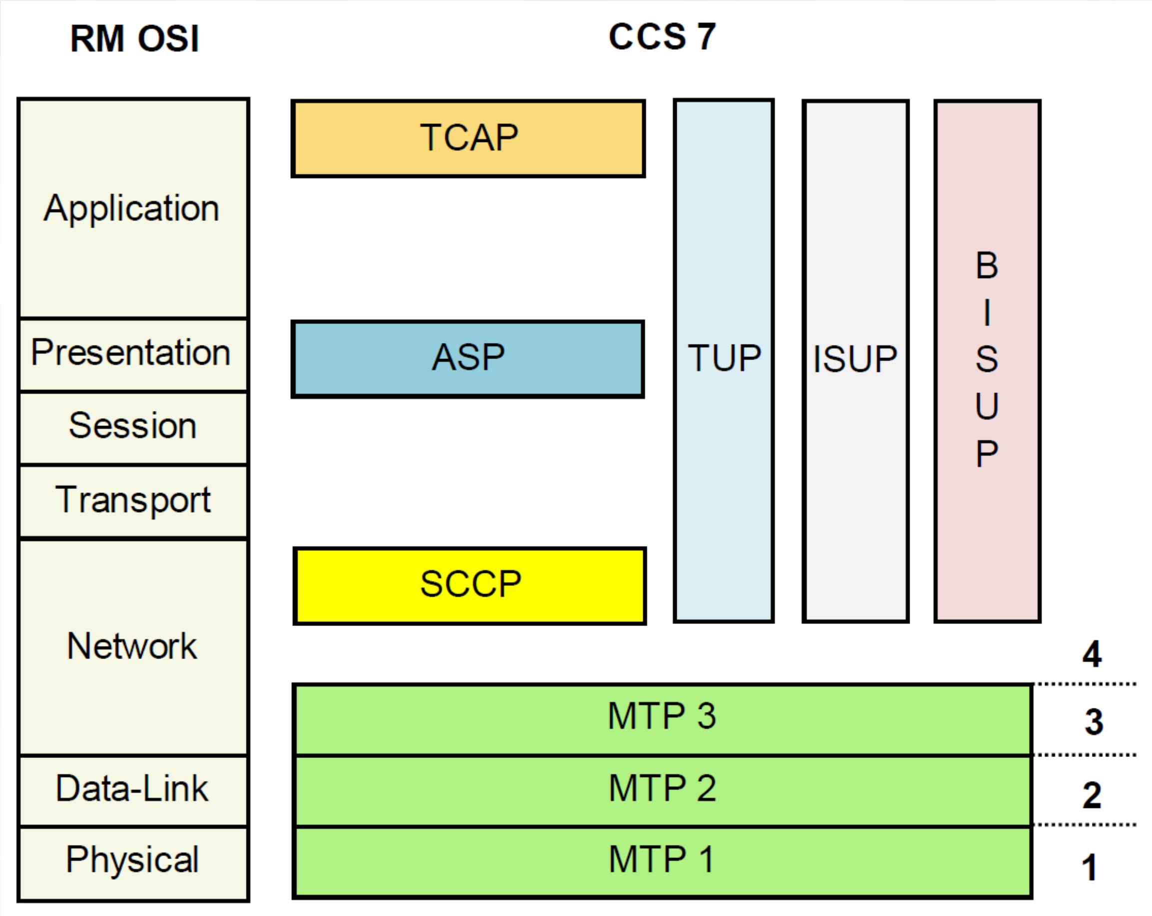
Quasi-associated signalling links



- Talk path connects directly both SSP
- Signalling path goes through one or more STP

CCS7 protocol stack

- CCS7 uses four-layer protocol stack
- the layers constitute a two part functionality
  - bottom three layers – transmission of the messages
  - upper portion – data process function
- signalling tasks are so divided to:
  - User Part (UP)
  - Message Transfer Part (MTP)


RM OSI
CCS 7
Application
Presentation
Session
Transport
Network
Data-Link
Physical
TCAP
ASP
SCCP
TUP
ISUP
B I S U P
MTP 3
MTP 2
MTP 1
4
3
2
1

- **Message Transfer Part Level 1** (MTP1)
    - defines the physical, electrical, and functional characteristics of the digital signaling link
- **Message Transfer Part Level 2** (MTP2)
    - at the data link layer
    - provides error detection, sequence checking, and initiates retransmission in case of erroneous reception of messages
    - MTP2 uses packets called **signal units** to transmit SS7 messages
    - three types of signal units:
        - **Fill-in Signal Unit** (FISU)
        - **Link Status Signal Unit** (LSSU)
        - **Message Signal Unit** (MSU)

- each Signal Unit contains:
  - **Flag** – sequence of bits
  - **FSN** (Forward Sequence Number) – counter of messages sent
  - **BSN** (Backward Sequence Number) – counter of successfully received messages
  - **BIB** (Backward Indicator Bit) – indications of erroneously received messages
  - **FIB** (Forward Indicator Bit) – indication of retransmission
  - **Length Indicator** – number of bits in information field

- **Fill-in Signal Unit** (FISU)
  - fill unit
  - no information field
  - sent when no other signal units are available

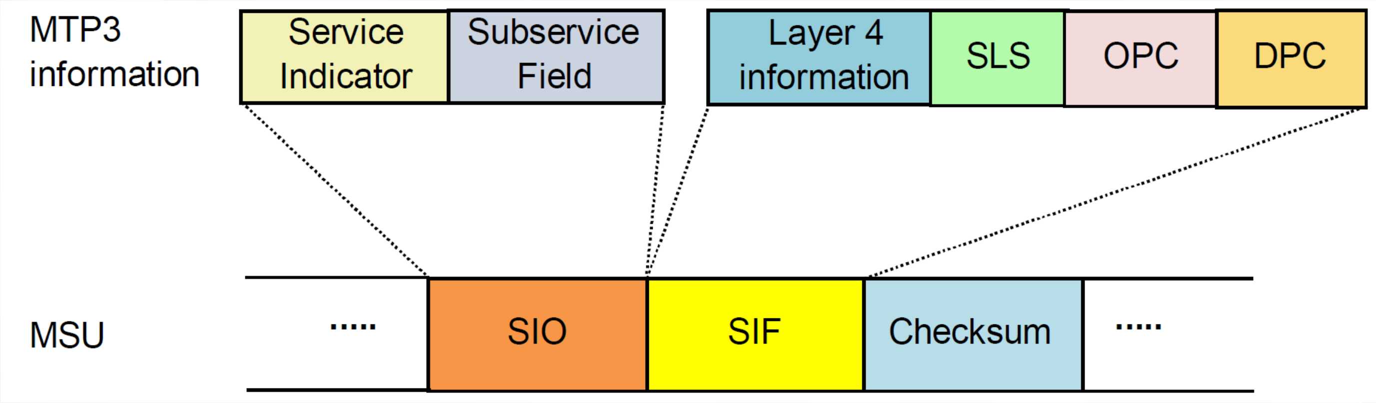
- **Link Status Signal Unit** (LSSU)
  - for monitoring of signalling connection,
  - i.e. when SP is busy, the receiver stops sending the MSU to the SP when the SIB (Status Indicator of Busy) is received

– **Message Signal Unit** (MSU)

Flag	BSN/BIB	FSN/BIB	Length Indicator	SIO	SIF	Checksum

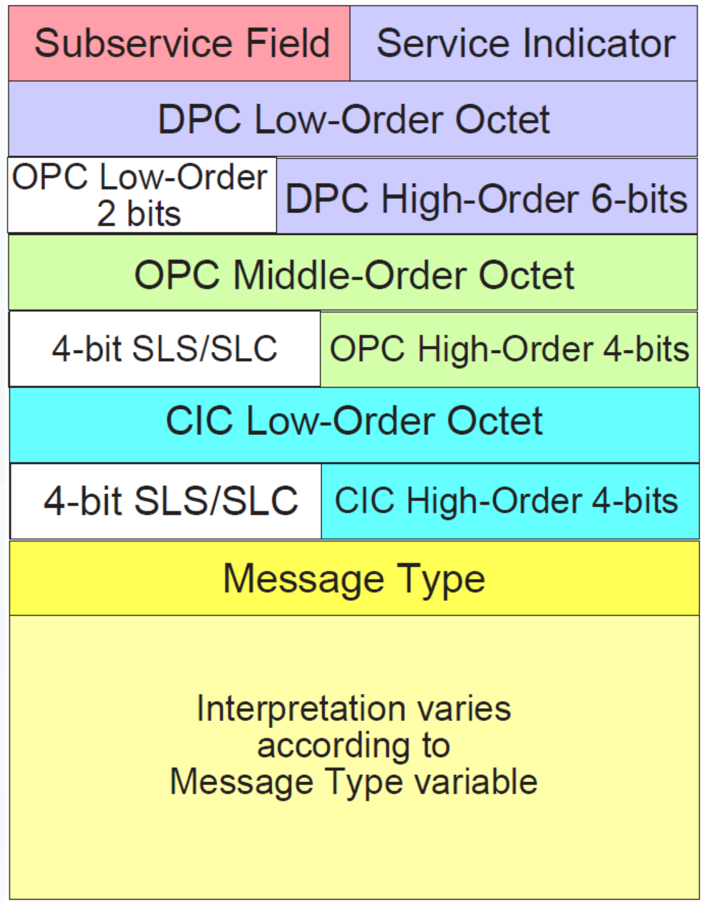
- for distribution of signalling messages
- is associated with call setup and termination
- provides MTP protocol fields, service indicator octet (SIO) and service information field (SIF)
- SIO identifies the type of protocol (ISUP, TCAP) and standard (ITU-TS, ANSI)
- SIF transfers control information and routing label

- **Message Transfer Part Level 3** (MTP3)
  - between MTP2 and the user parts
  - is the network layer in the CCS7 protocol stack
  - ensures reliable transfer of the signaling messages
  - the endpoint of SU is given by address
  - address is given by
    - **DPC** (Destination Point Code),
    - **OPC** (Origination Point Code),
    - **SLS** (Signaling Link Selection).

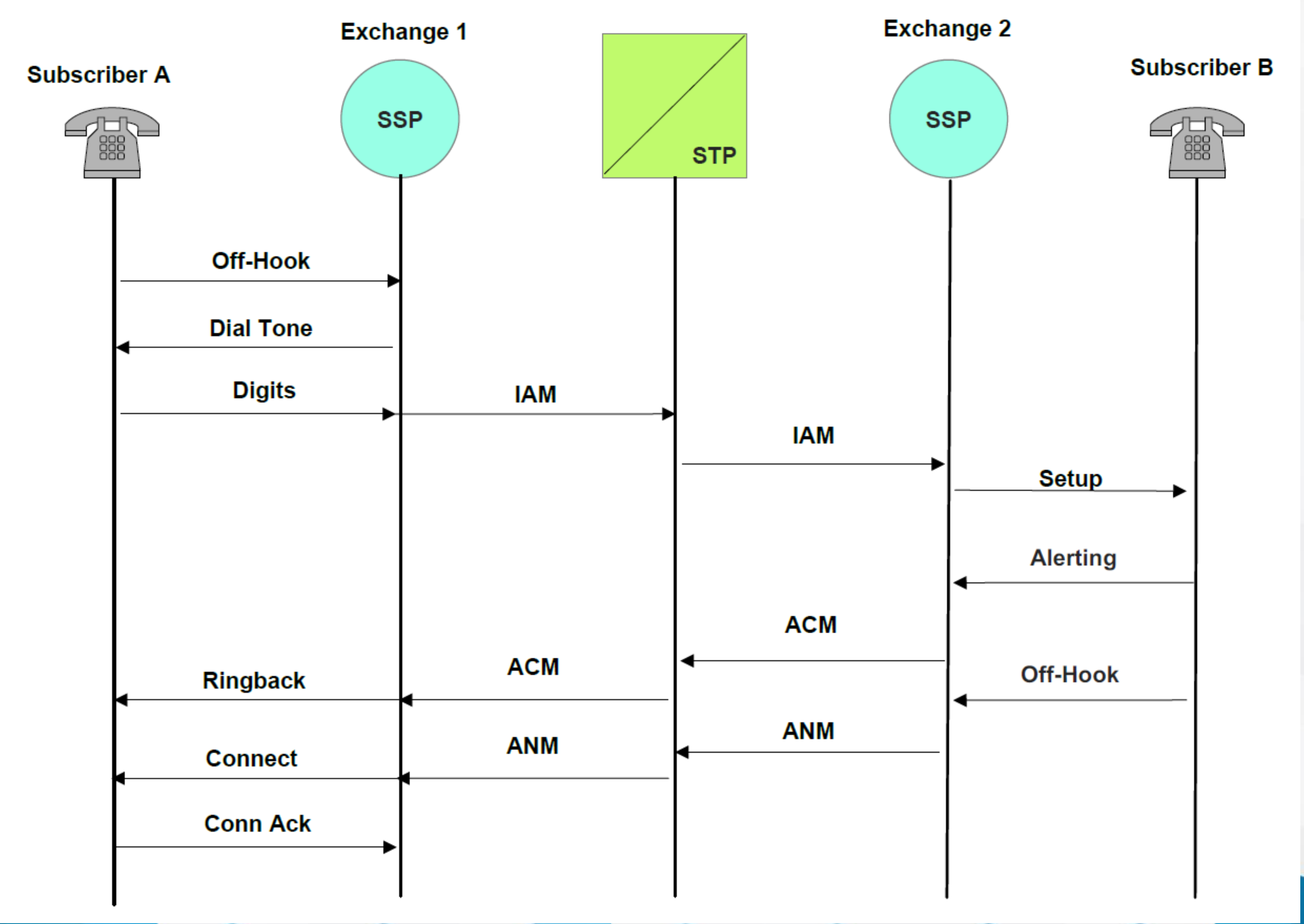


- **ISUP**
  - defines the protocol and procedures used to setup, manage and release trunk circuits that carry voice and data calls over the ISDN
  - used for both ISDN and non-ISDN calls
  - the basic service provided by the ISUP is the establishment and clearing of circuit-switched calls
  - ISUP defines signalling messages:
    - **IAM** (Initial Address Message) – initialization of speech connection
    - **ACM** (Address Complete Message) - message returned from the terminating switch when the subscriber is reached and the phone starts ringing
    - **ANM** (Answer Message)- Sent when the subscriber picks up the phone
    - **CPG** (Call Progress Message) - Contains additional information about the progress of a call
    - **REL** (Release Message) - Sent to clear the call when a subscriber goes on hook
    - **RLC** (Release Complete Message) - Acknowledgment of the release

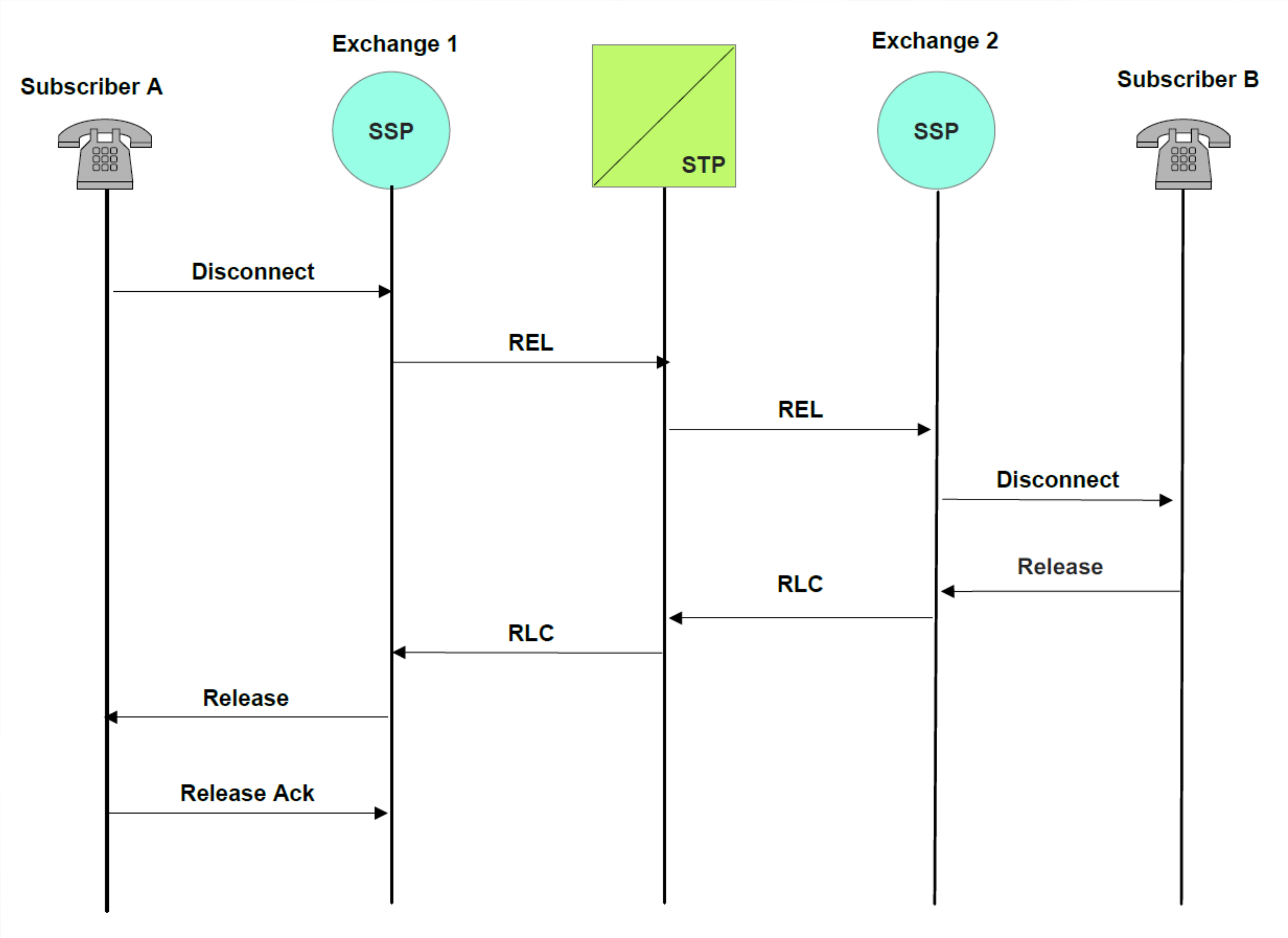
- structure of ISUP message



- call setup using ISUP



- call terminate using ISUP



- **Signaling Connection Control Part** (SCCP)
  - defined in Q.713
  - routing protocol in layer 4
  - provides connectionless and connection-oriented network services
  - SCCP provides connectionless and connection-oriented network services and **global titl**e translation (GTT) capabilities above MTP Level 3.
  - A **global title** is an address (e.g., a dialed 800 number or mobile subscriber identification number) that is translated by SCCP into a destination point code and subsystem number.

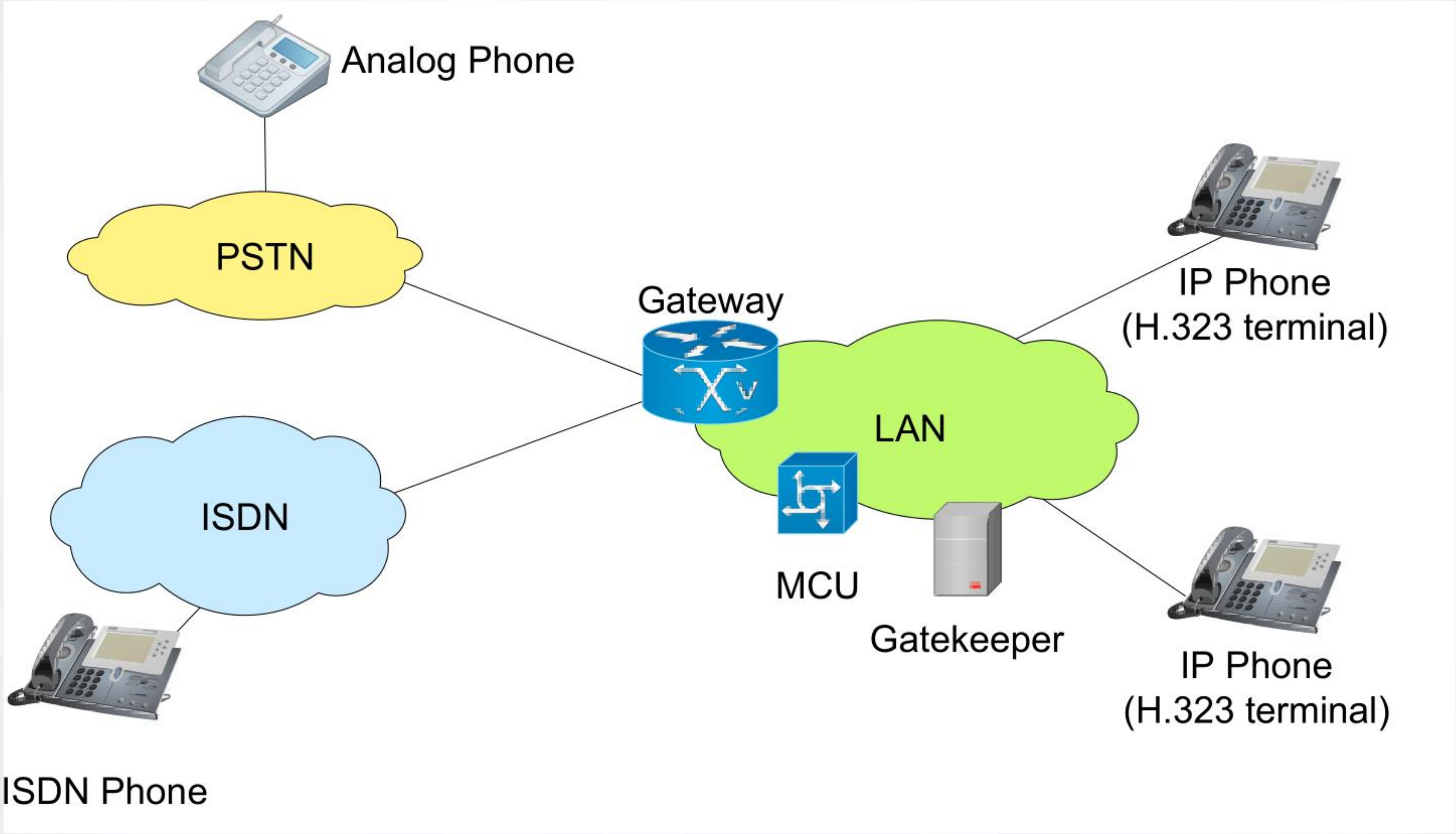
Error Correction in CCS7

- Two methods of error correction are available
  - **Basic Error Correction** (BEC) - sender retransmits the corrupt (or missing) MSU and all subsequent MSUs
  - **Preventive Cyclic Retransmission** (PCR) method - an alternative method for large propagation delays, such as satellite circuits. MSUs are stored by the transmitting terminal until a positive acknowledgment (ACK) is received. When no new MSUs are to be sent, unacknowledged MSUs are retransmitted cyclically until positively acknowledged.

H.323

- defines the protocols to provide audio-visual communication sessions on any packet network.
- H.323 defines the interworking of
  - call signaling,
  - call control,
  - and media stream protocols.
- An **administrative domain** is the collection of all zones that are under the control of a single person or organization, such as a service provider.
- **Gatekeeper** provides address translation and controls access to the network resources for H.323 terminals, GWs and MCUs
- **MCU** - enables conferencing between three or more endpoint
- **Gateway** - enable communication between H.323 networks and other networks, such as PSTN or ISDN networks

- H.323 architecture



SIP (Session Initiation Protocol)

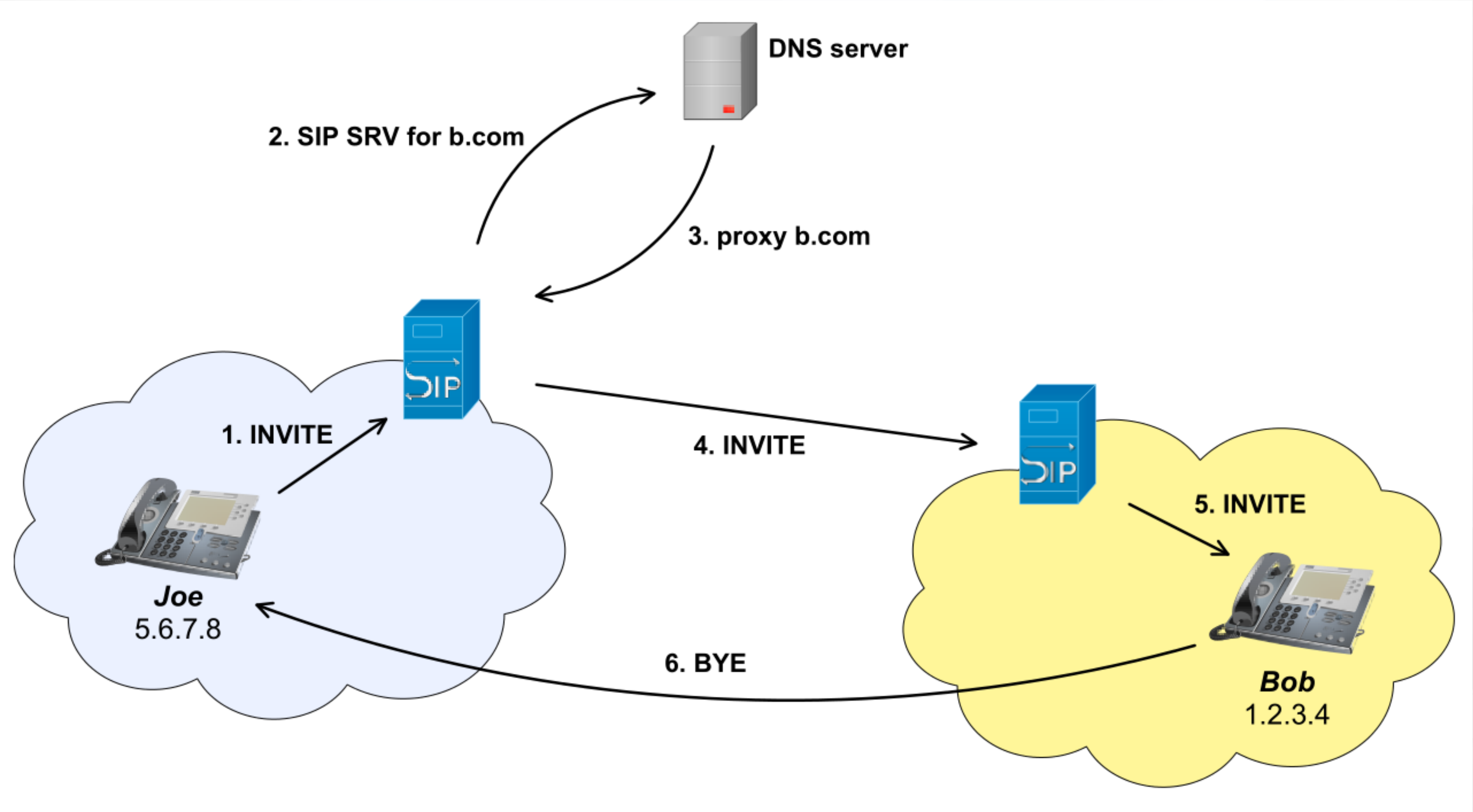
- For signalling and controlling multimedia communication sessions,
- the most common applications of SIP are in Internet telephony for voice and video calls,
- in conjunction with SIP, two other protocols are used
  - RTP (Real Time Protocol)
  - SDP (Session Description Protocol).
- RTP protocol is used to transmit multimedia in real-time, this protocol can transmit voice or video packets using IP.
- SDP is used to describe properties of the subscriber connection. This description is then used to negotiate the connection parameters of all devices involved in the concentration (codec negotiation of transport protocol).
- SIP is based on HTTP protocol also because that HTTP is undoubtedly the most successful and the most widely used protocol on the Internet.

- basic elements:
  - user agents,
  - proxies, registrars, and redirect servers.
- UAC (user agent client) - caller application that initiates and sends SIP requests.
- UAS (user agent server) - receives and responds to SIP requests on behalf of clients, accepts, redirects or refuses calls.
- SIP Terminal - supports real-time, 2-way communication with another SIP entity.
- Proxy – contacts one or more clients or next-hop servers and passes the call requests further. Contains UAC and UAS.

Connection progress using SIP

- A typical configuration is such that each unit (company) has its own SIP server, which is used by all UA administered within the unit.

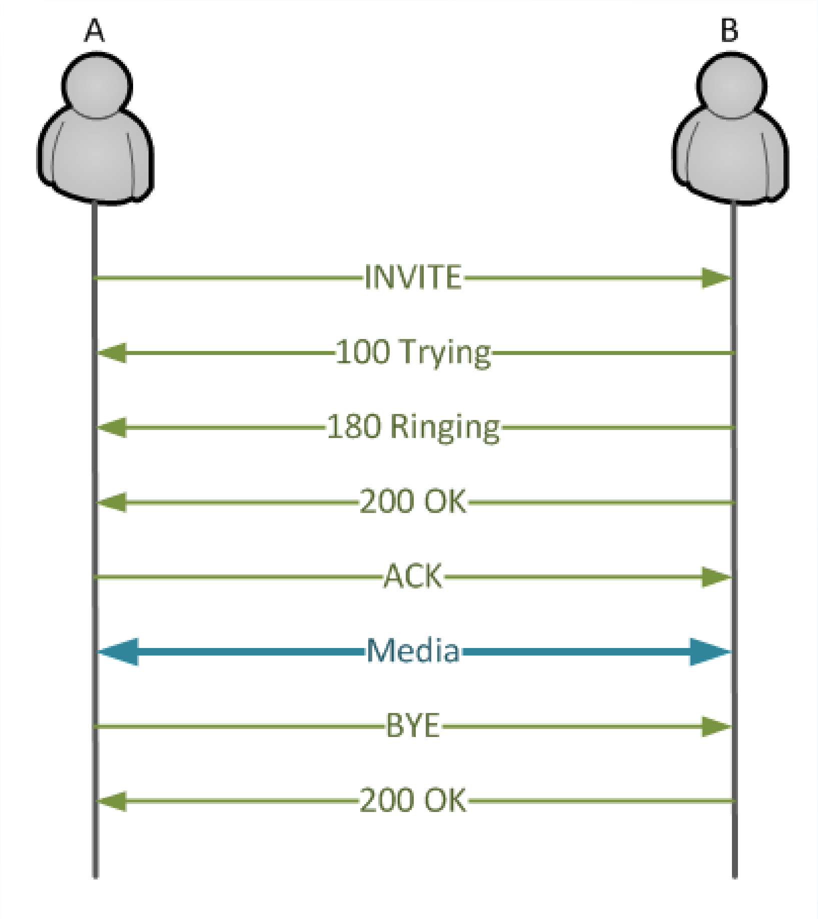
- User Joe Bob calls and uses the address **sip: bob@b.com**
- UA does not know where to send the request to establish a connection, but is configured so that all outbound traffic is sent to the SIP proxy server with the address of its company **proxy.a.com**.
- Proxy server detects that the user sip: bob@b.com is another company and thus by querying the DNS for matching SIP proxy server where to send the request.
- Correspondingly, the server is **proxy.b.com**.
- Request thus arrive at proxy.b.com. Proxy knows that Bob is currently in his office and reached for the phone on his desk, which has an IP address of **1.2.3.4**, so Proxy sends INVITE request.


DNS server
2. SIP SRV for b.com
3. proxy b.com
1. INVITE
4. INVITE
5. INVITE
Joe
5.6.7.8
6. BYE
Bob
1.2.3.4

- There are two different types of SIP messages: requests and responses
- requests:
  - **INVITE** — request to establish a call (a session),
  - **CANCEL** — stop an INVITE that is in progress (that is, the call has not been established yet),
  - **ACK** — to confirm that the endpoint has received a final response in a transaction,
  - **BYE** — to end an established call (compare with CANCEL that is used to stop the session before it has been established),
  - **REGISTER** — to register the SIP endpoint at the registrar server
  - **OPTIONS** — to ask the other party for the list of SIP methods it supports. The response may also contain the set of capabilities (i.e. audio/video codecs) of the responding party.

- responses:
  - **1xx** - Informational
  - **2xx** - Success
  - **3xx** - Redirection   R
  - **4xx** - Request Failure
  - **5xx** - Server Failure
  - **6xx** - Global Failure

SIP Dialog



ENUM

- tElephone NUmbers Mapping
- system of unifying the international telephone number system with the Internet addressing and identification name spaces.
- ENUM uses special DNS record types to translate a telephone number into a **Uniform Resource Identifier** (URI) or IP address that can be used in Internet communications
- through so-called **NAPTR** record a special form of SIP address is stored in the form of URI

  `sip: libor.michalek@vsb.cz`

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