VSB TECHNICAL UNIVERSITY OF OSTRAVA | FACULTY OF ELECTRICAL ENGINEERING AND COMPUTER SCIENCE | DEPARTMENT OF TELECOMMUNICATIONS

Telecommunication Networks

Signalling in telecommunication networks

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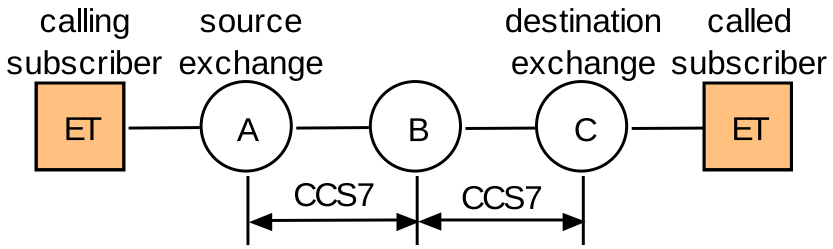
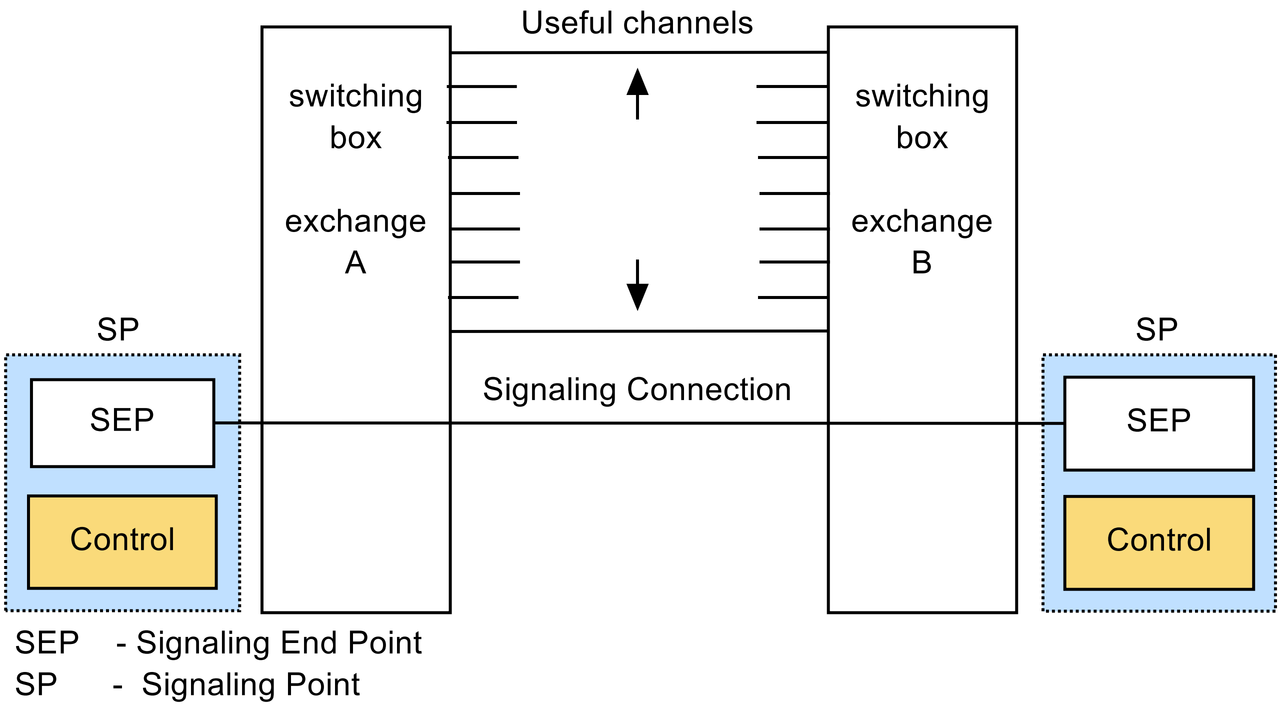
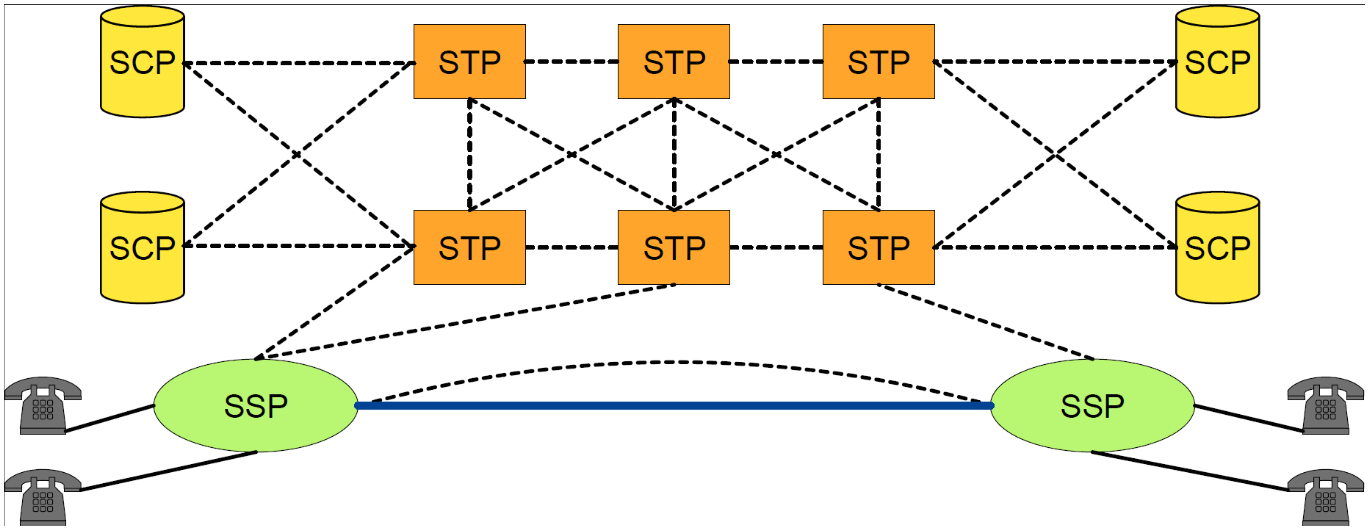
CCS7 (Common Channel Interoffice Signaling System No. 7)

- 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)



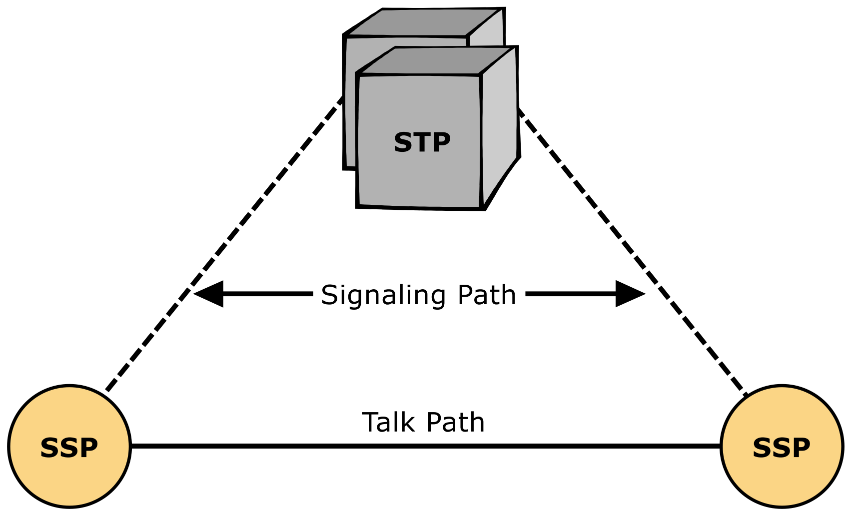
- **Signaling Transfer Points** (STPs) receive and route incoming signaling messages toward their destination based on destination address
- **Service Control Points** (SCPs) are databases that provide the necessary information for special call processing and routing
- **SSP (Signalling Switching Points)** - 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

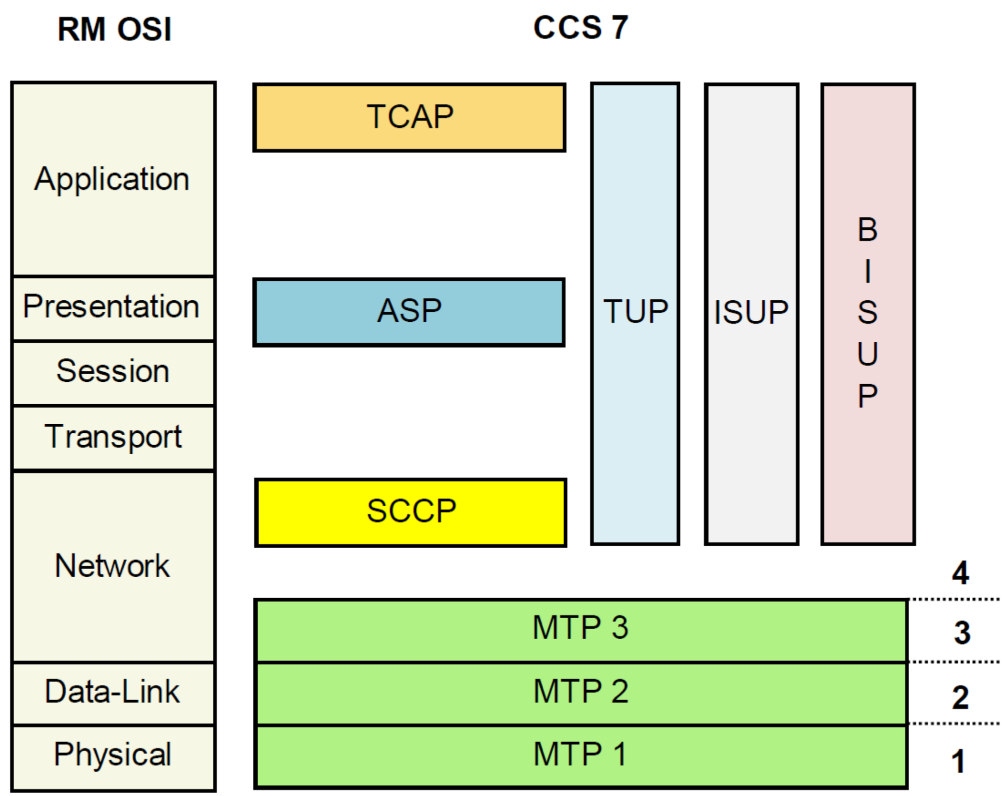
- Quasi-associated signalling links



- 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)



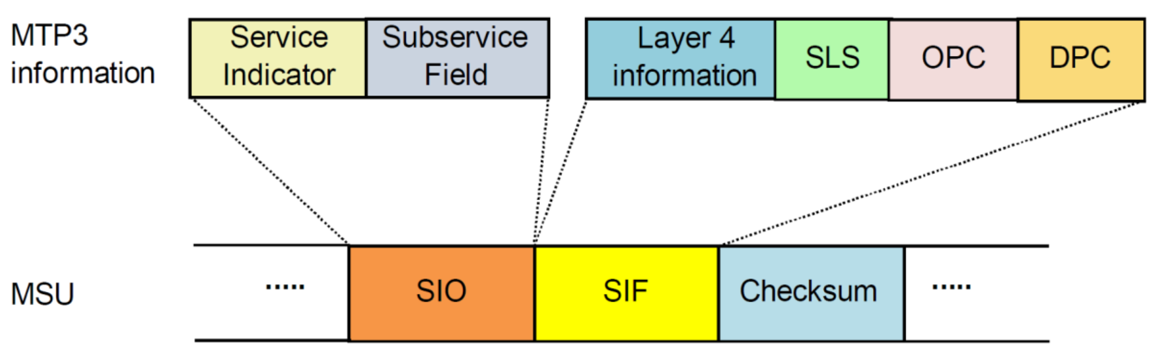
- **Message Transfer Part Level 1** (MTP1)
  - defines the physical, electrical, and functional characteristics of the digital signaling link
- **Message Transfer Part Level 2** (MTP2)
  - at the data link layer
  - provides error detection, sequence checking, and initiates retransmission in case of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

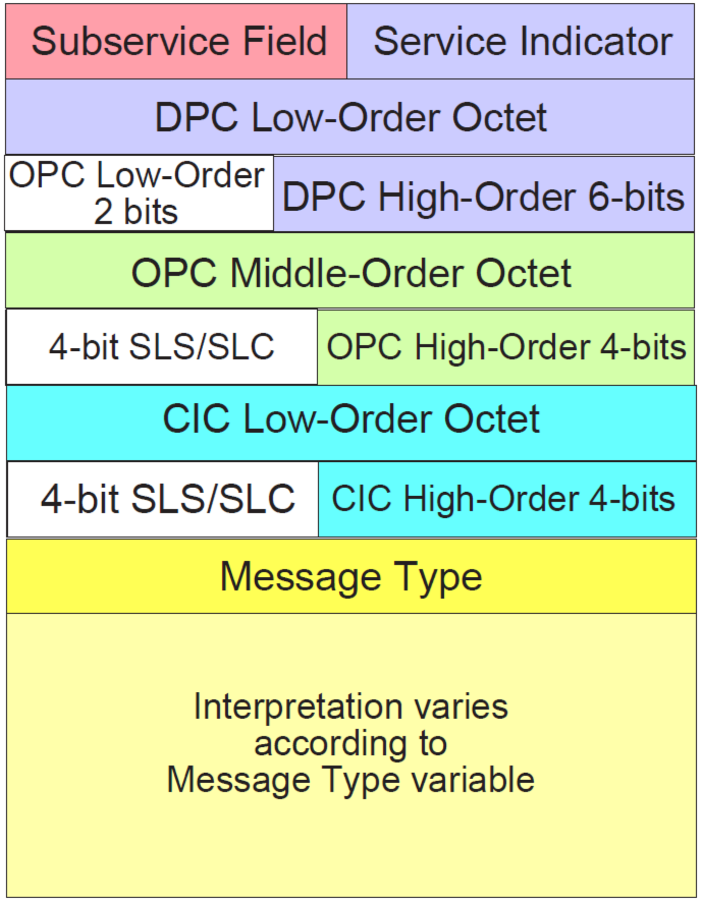
  - for distribution of signalling messages
  - is associated with call setup and termination
  - provides MTP protocol fields, service indicator octet (SIO) and service information field (SIF)
  - SIO identifies the type of protocol (ISUP, TCAP) and standard (ITU-TS, ANSI)
  - SIF transfers control information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

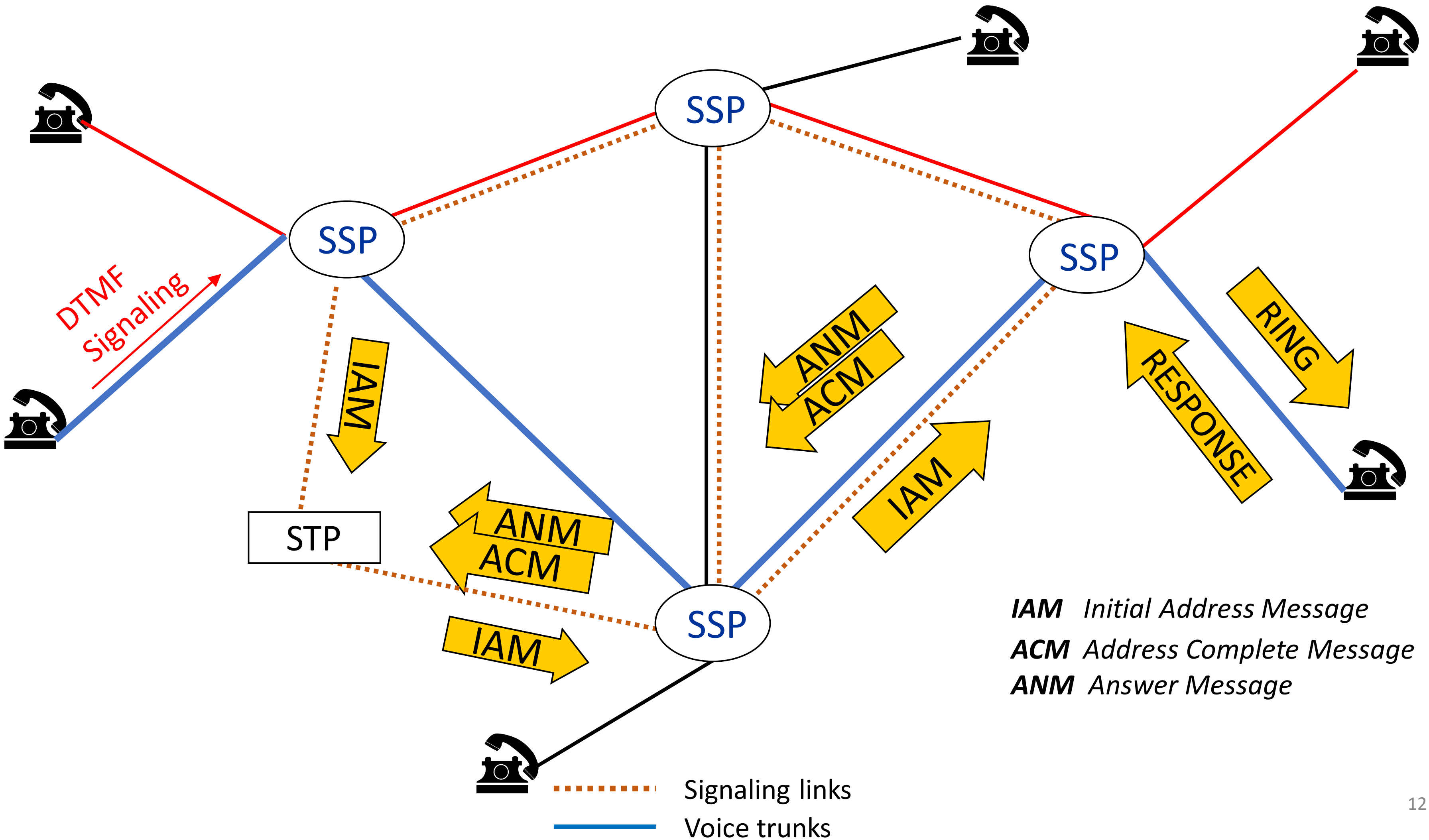


- **ISUP**
  - defines the protocol and procedures used to setup, manage and release trunk circuits that carry voice and data calls over the ISDN
  - used for both ISDN and non-ISDN calls
  - the basic service provided by the ISUP is the establishment and clearing of circuit-switched calls
  - ISUP defines 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

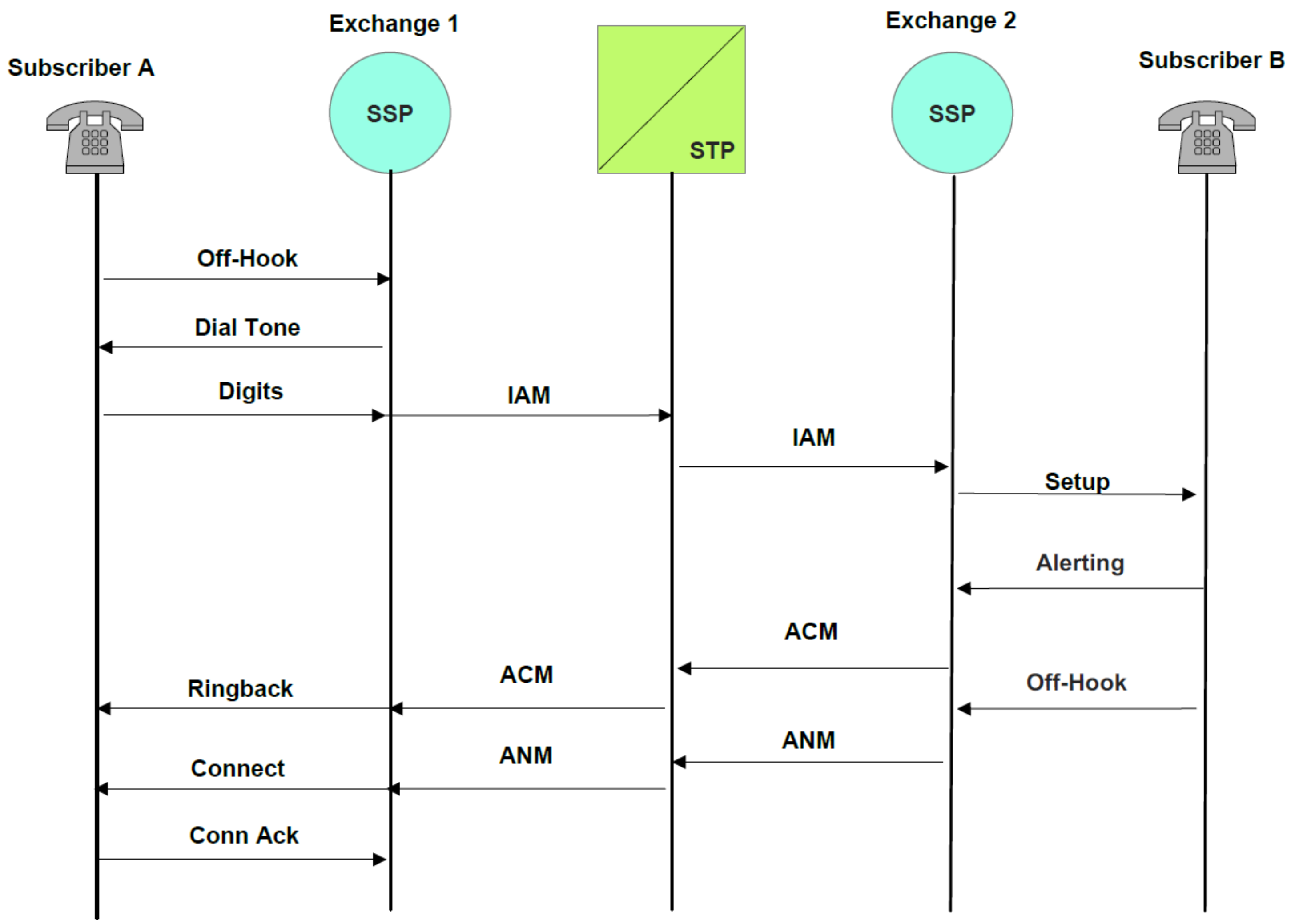


Basic call setup (using ISUP)

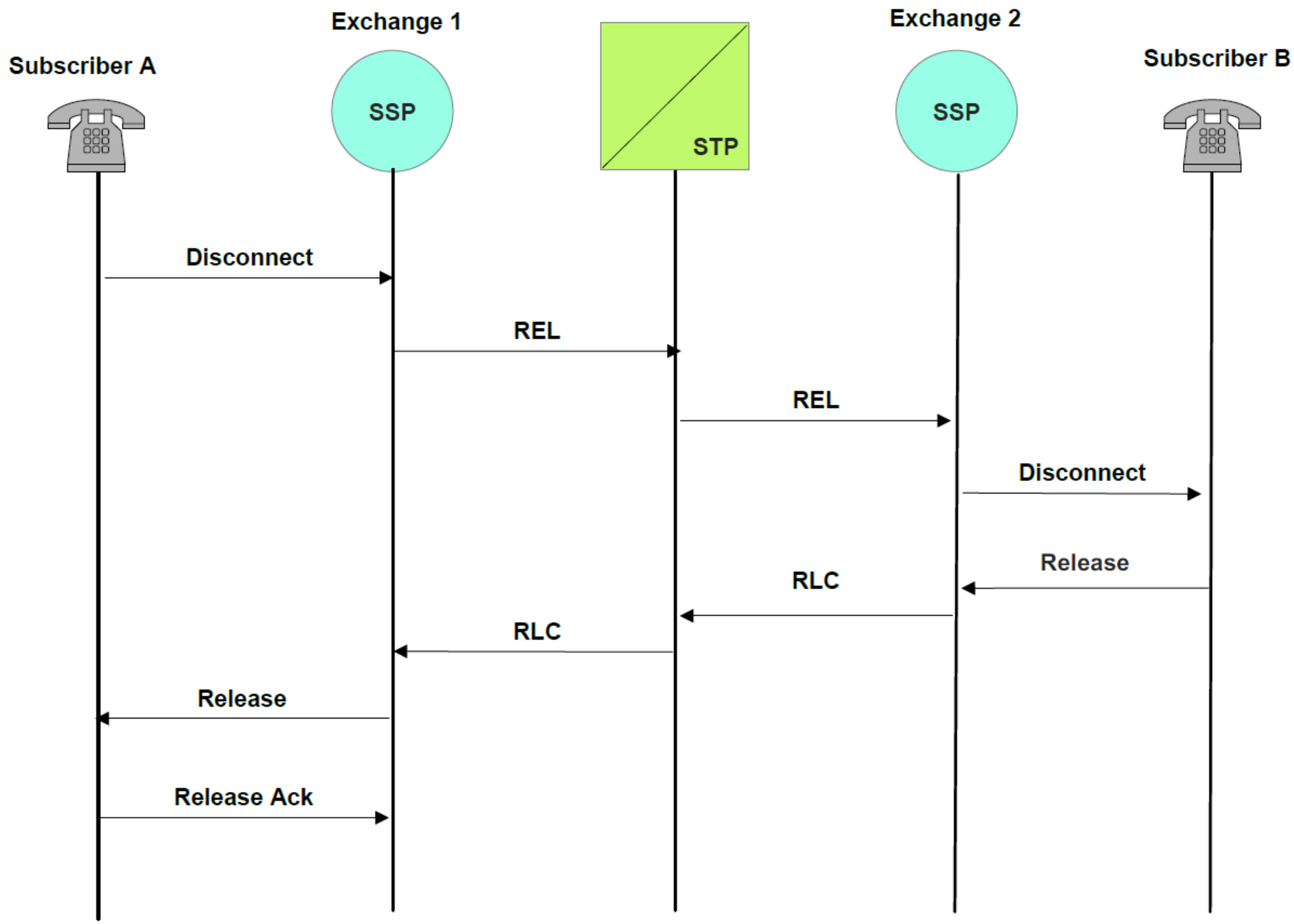


IAM *Initial Address Message*
ACM *Address Complete Message*
ANM *Answer Message*

Call Setup Using ISUP



Call Terminate Using ISUP



H.323 Protocol Suite

- H.323 standards specifies the elements, protocols, and procedures providing multimedia communication over packet-based networks.
- H.323 defines systems and functions for audiovisual services over packet switched networks which may not provide a guaranteed Quality of Service.
- References to other standards and ITU recommendations.
- Interoperability with other multimedia networks is the primary goal for developing H.323.

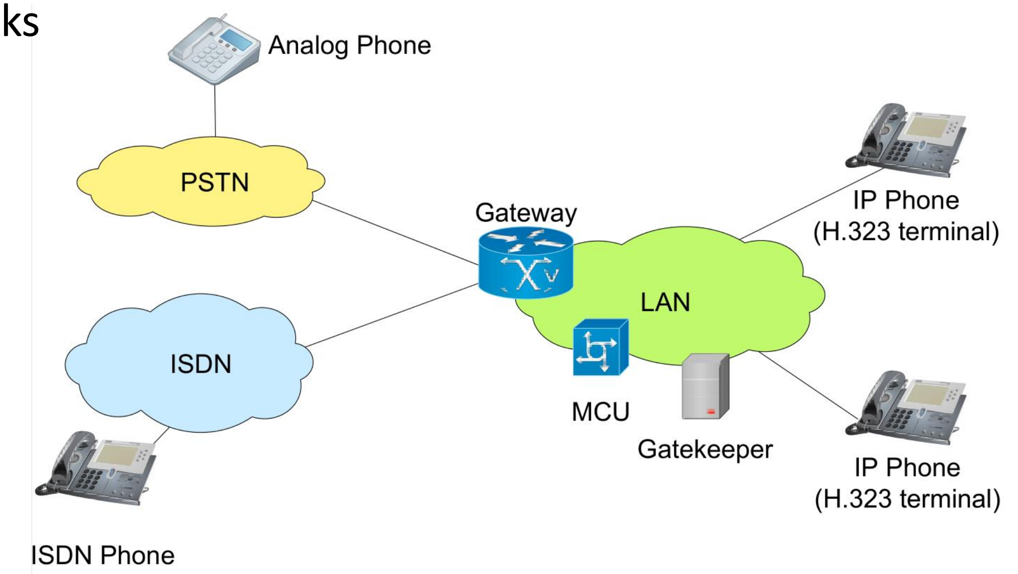
Network	Non-guaranteed Bandwidth packet-switched networks (e.g. IP)
Video	H.261, H.263
Audio	G.711, G.722, G.728, G.723, G.729
Call signaling and media packetisation	H.225
Call Control	H.245
Multipoint	H.323
Data	T.120

H.323 Architecture

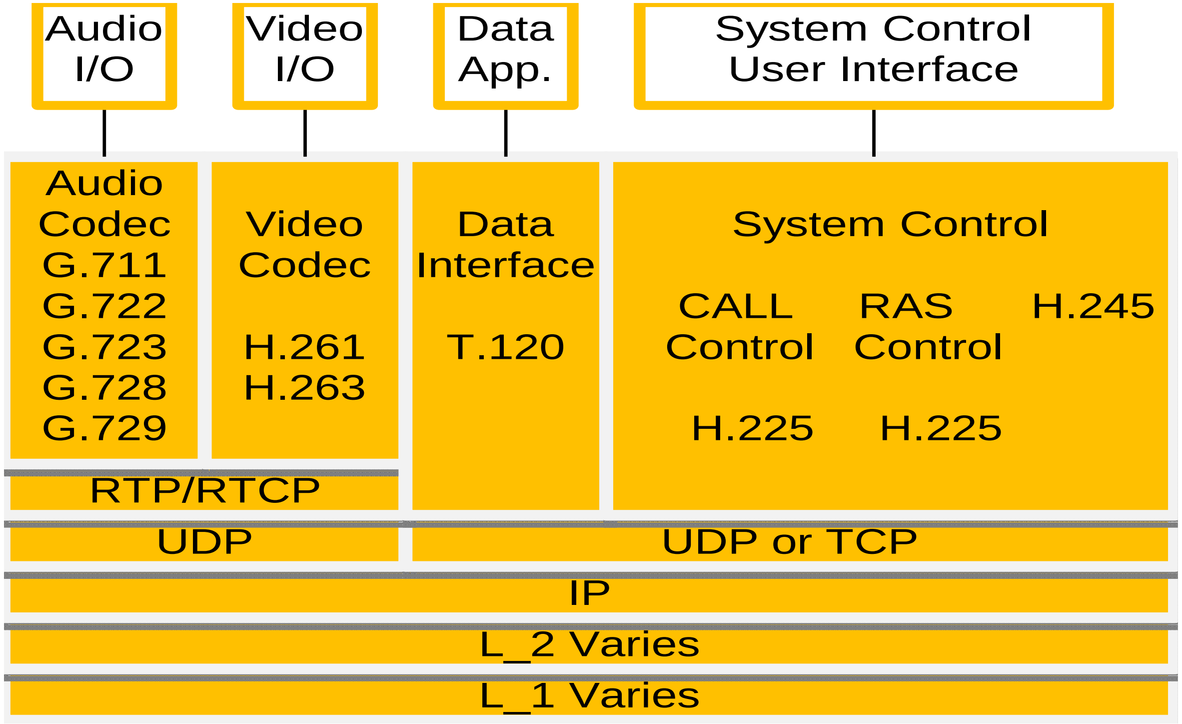
- The four main enlements are:
  - **Terminal -** including Video I/O equipment, Audio I/O equipment, User Data Applications, PC, and System Control User Interface etc.
  - **Gatekeeper -** admission control for the network, bandwidth control and management, address resolution

    978-555-4567 → 204.124.46.19

    E.164 Number Network address
  - **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,
- SIP is the core protocol for initiating, managing and terminating sessions in the Internet
- These sessions may be text, voice, video or a combination of these
- 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.

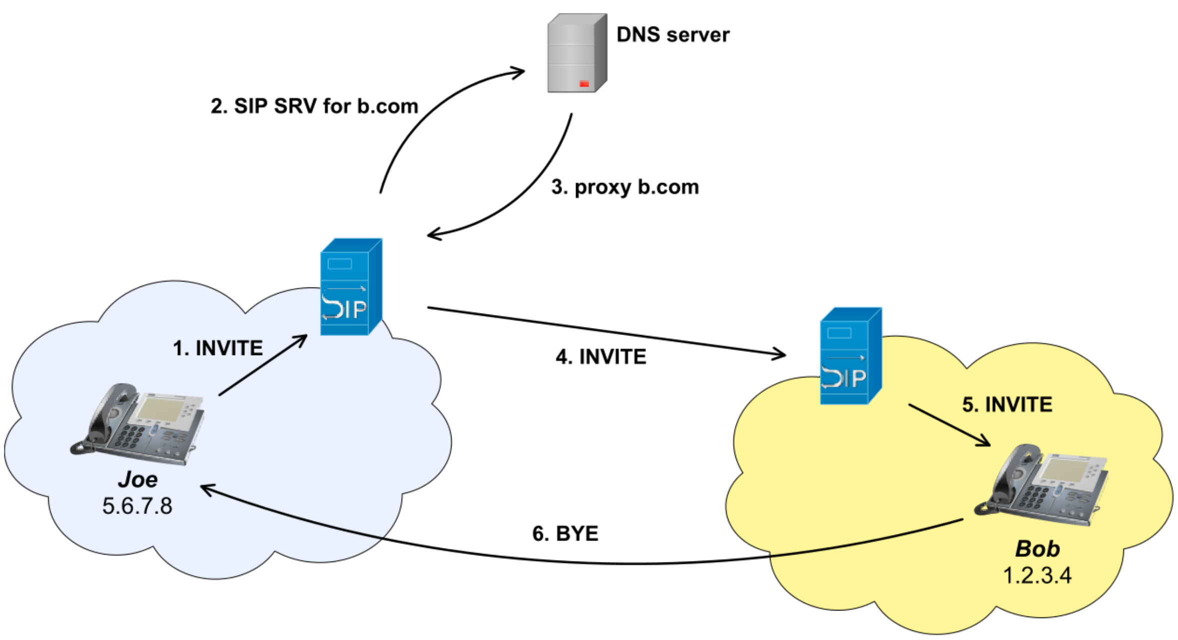
SIP Entities

- User Agents:
  - User Agent Client
  - User Agent Server
- Proxy Server
- Redirect server
- Registrar

Connection progress using SIP

- A 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.

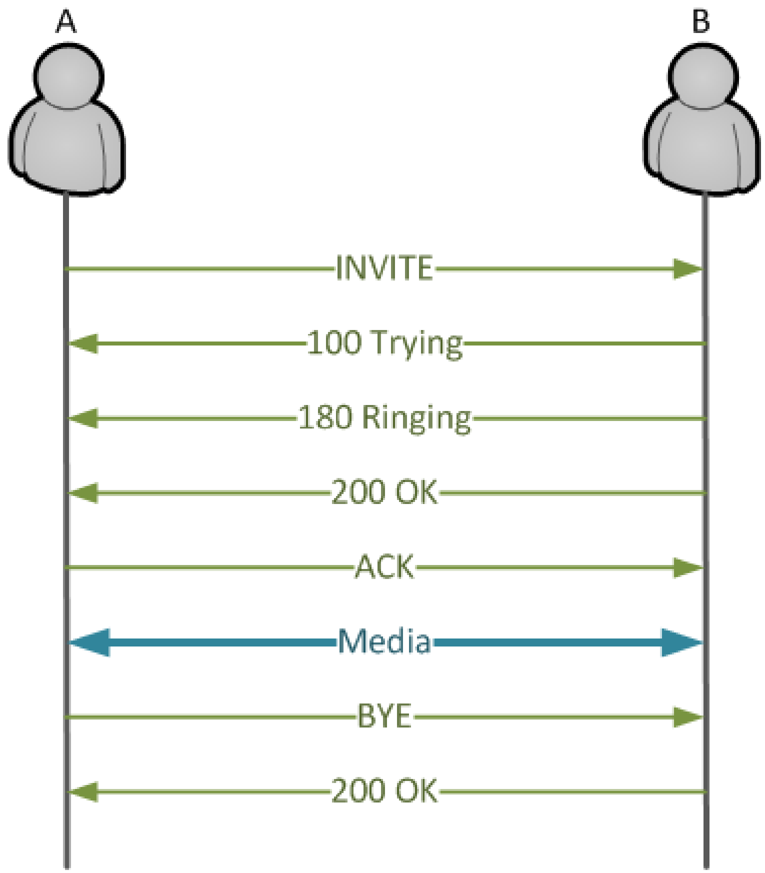
- Correspondingly, the server is **proxy.b.com**.
- Request thus arrive at proxy.b.com. Proxy knows that Bob is currently in his office and reached for the phone on his desk, which has an IP address of **1.2.3.4**, so Proxy sends INVITE request.



SIP 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
  - **4xx** - Request Failure
  - **5xx** - Server Failure
  - **6xx** - Global Failure

SIP Dialog



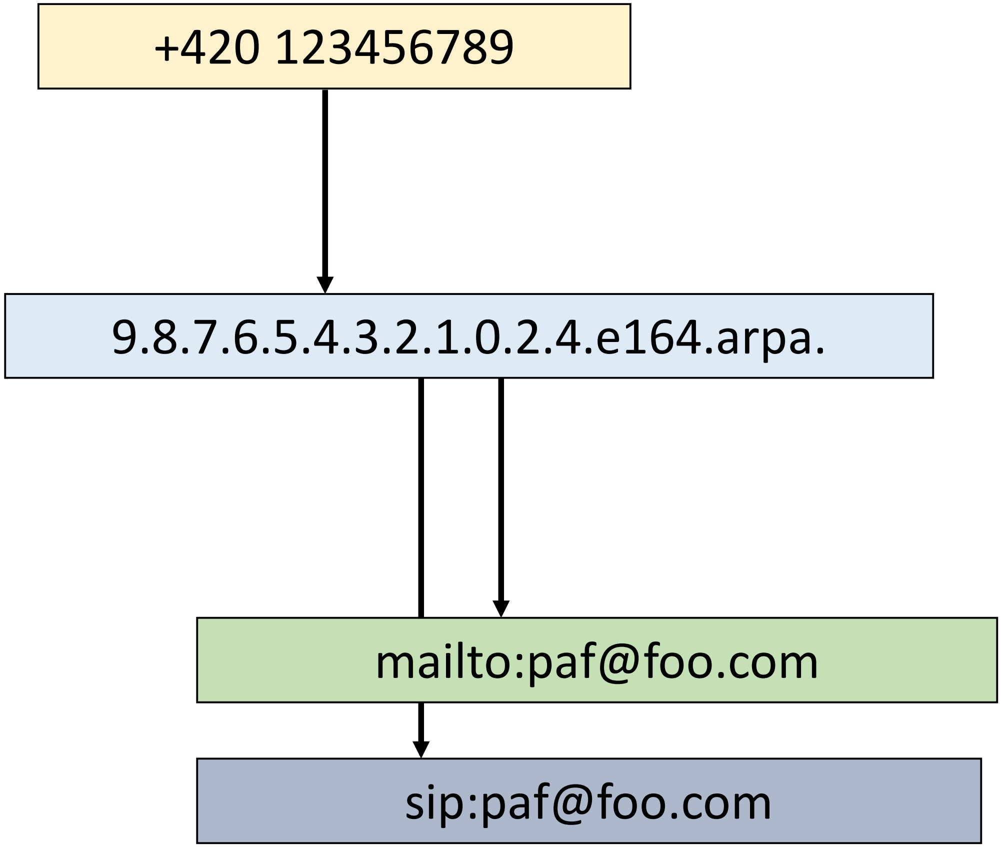
ENUM

- 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

ENUM

- take phone number
- turn into domain name
- ask the DNS
- return list of URI’s



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