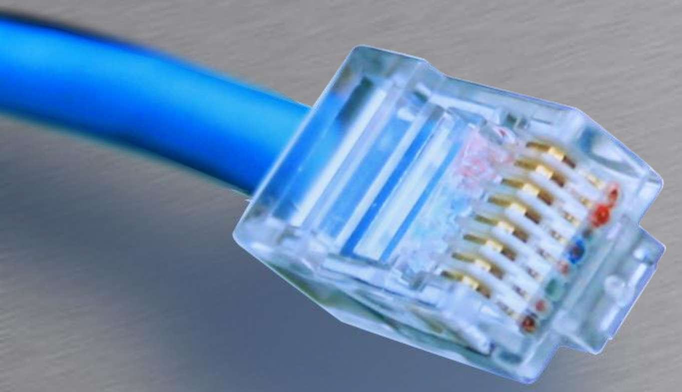
IP Telephony



HOUSE OF TECHNOLOGY

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IPT Protocols

Telephony Protocols and VoIP

Subjects

- VoIP signalling standards overview
- IETF SIP signalling including registration
- ITU H.323 protocol suite
- RTP/RTCP and ZRTP/SRTP overview
- The IP Phone soft/hard, DSP, PoE

REVIEW

Generic VoIP

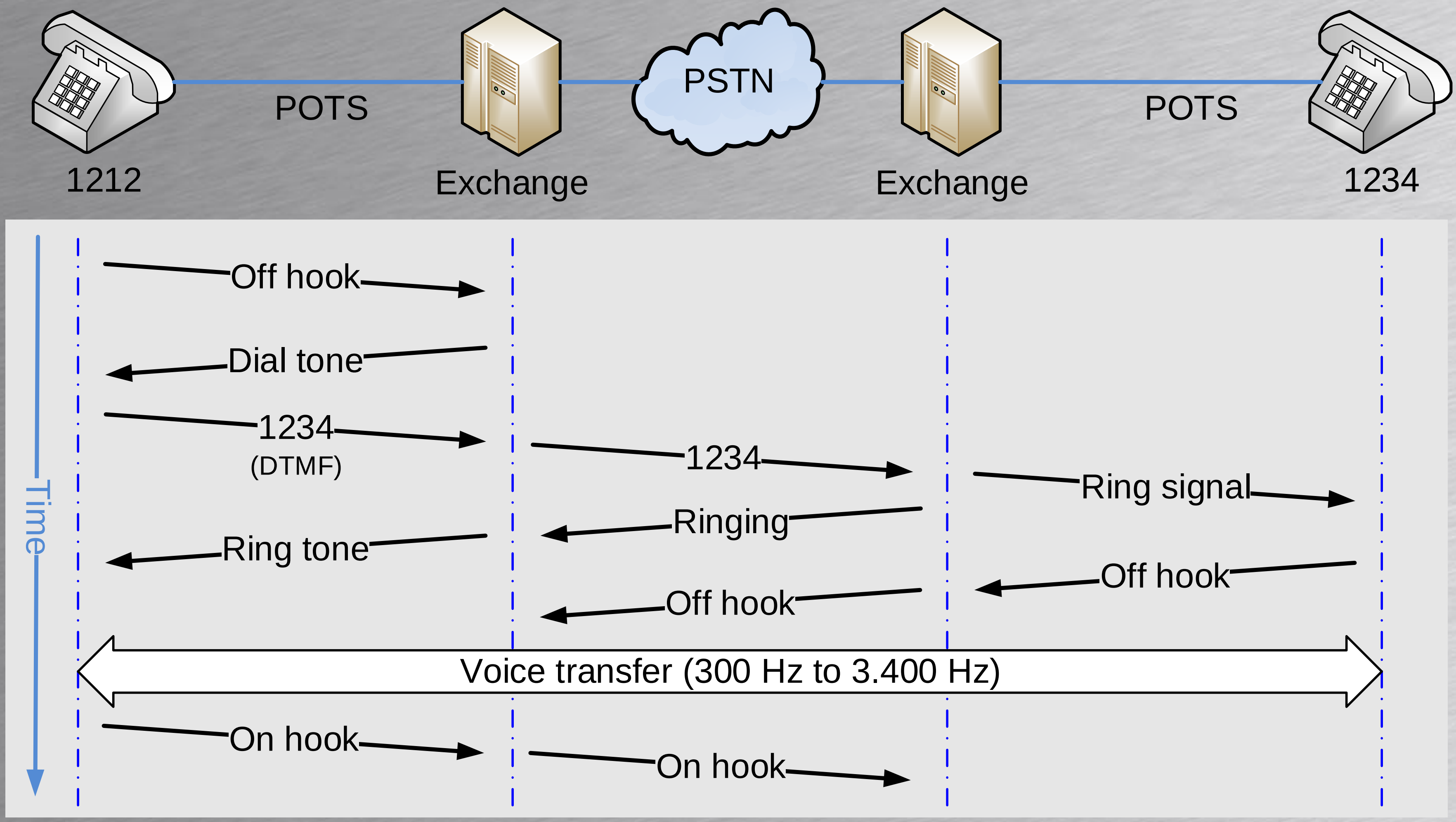
DTMF Dialing

- Telephone sends dual tone when keypad pressed
- Exchange recognizes dual tones and interprets them as digits
- All tones are within the 300 to 3.400 Hz band
  - This is called in-band signalling
  - Voice and signalling carried in the same band

Frequency	1209 Hz	1336 Hz	1477 Hz	1633 Hz
697 Hz	1	2	3	A
770 Hz	4	5	6	B
852 Hz	7	8	9	C
941 Hz	*	0	#	D

signalling and voice

POTS call progress



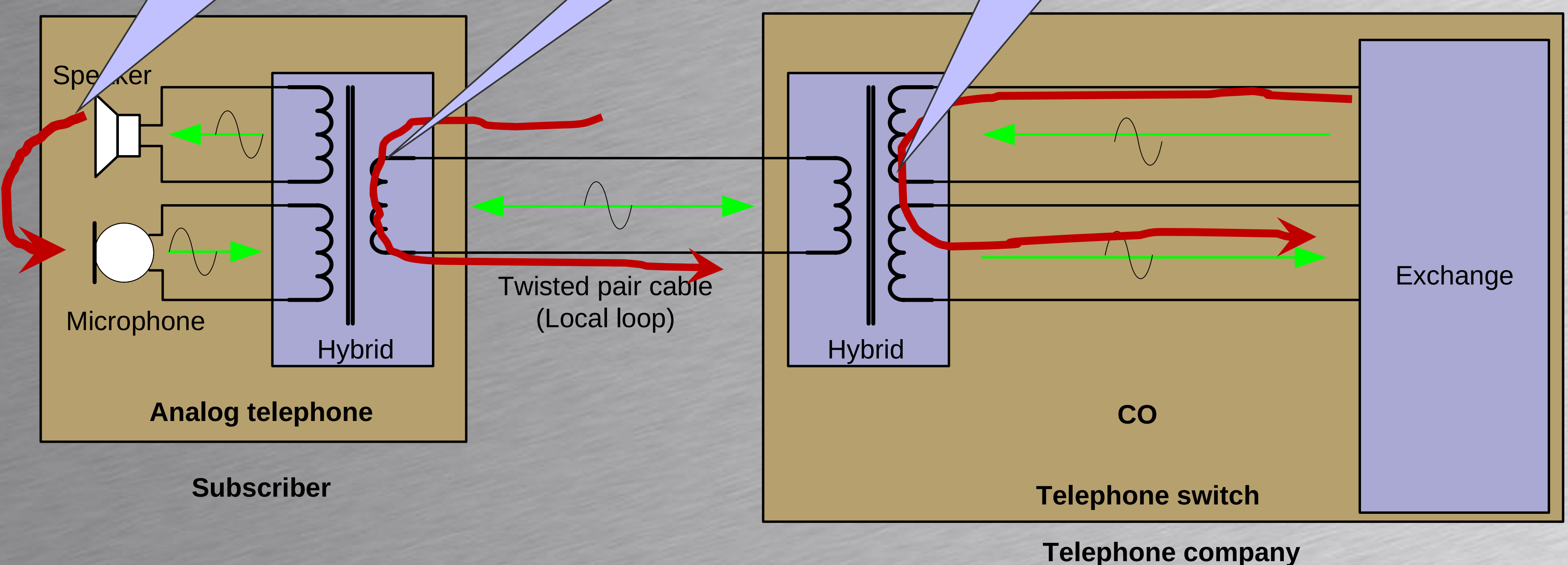
Echo

Mismatch impedance in hybrid echos energi back

Mismatch impedance in hybrid echos energi back as echo

Acustic feedback. Sound from Speaker sent back through the microphone to remote user as echo

- … ce in …
  - … ss … an 2…
  - A… ng if the dela… s greate… an 25 mS
  - …n be very ann…ying if grea… than 250 mS



Circuit vs. packet switched

- Circuit switched network (PSTN)
  - A logical connection is made between the two endpoints. (Phones)
  - Bandwidth guaranteed (64 Kbps for voice)
  - Delay constant
- Packet switched network (IP network)
  - A logical connection use TCP/UDP between the two endpoints. (IP Phones)
  - No bandwidth guarantee
  - Delay <u>not</u> constant

MOS
Mean Opinion Score

- MOS used to validate the quality of telephone voice quality

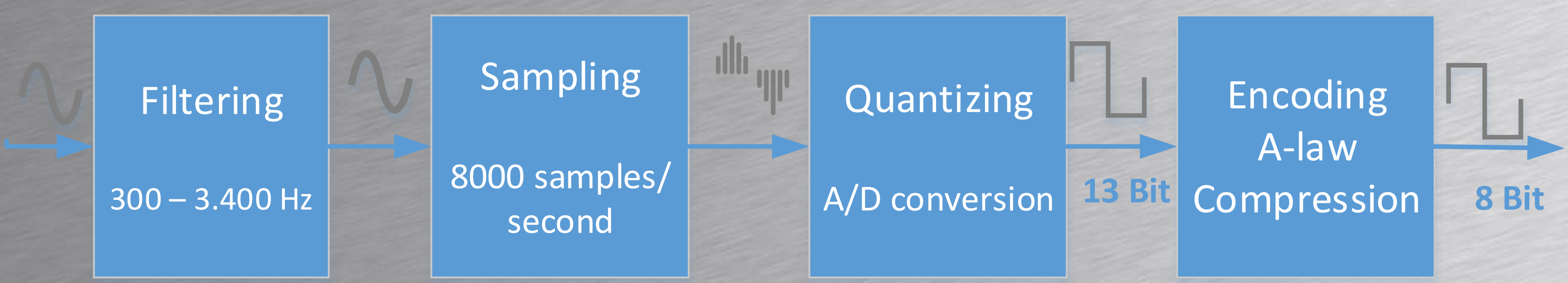
MOS	Quality	impairment
5	Excellent	Imperceptible
4	Good	Perceptible but not annoying
3	Fair	Slightly annoying
2	Poor	Annoying
1	Bad	Very annoying

Codec	From	Bandwidth	Data rate	MOS
G.711	1972	300-3.400 Hz	64 Kbps	4,1
G.722	1988	50-7.000 Hz	64 Kbps	~4,5
G.726	1990	300-3.400 Hz	32 Kbps	3,85
G.729	1996	300-3.400 Hz	8 Kbps	3,92
GSM EFR	1995	300-3.400 Hz	12,2 Kbps	3,8

Codec G.711 A-law

- From analog audio signal to coded bit stream
  - Advanced mathematics used in encoding to preserve voice quality

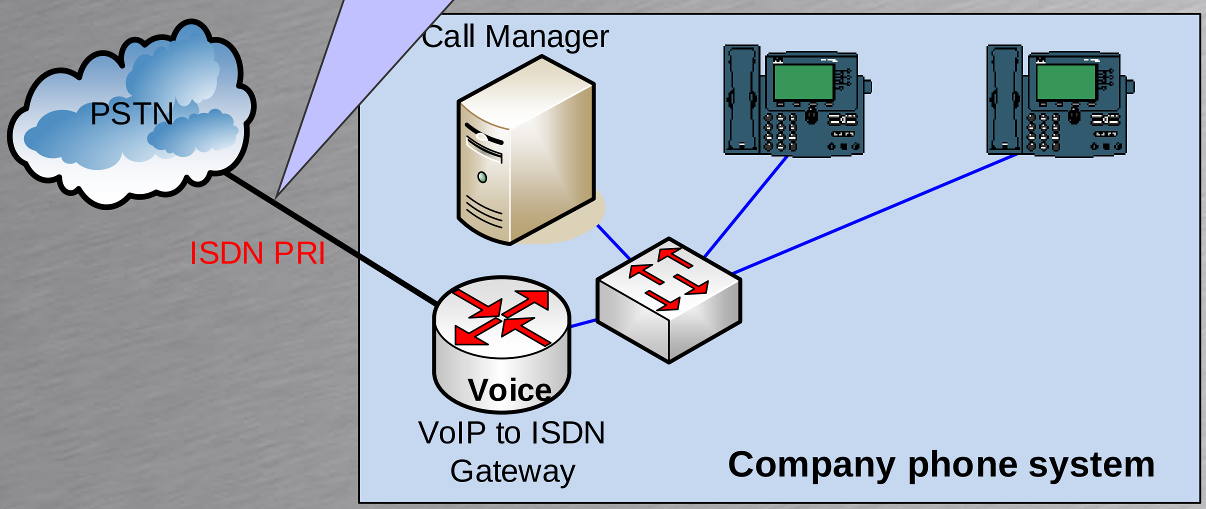
Stage	Signal	Comment
Filtering	Analog	Input from microphone. High bandwidth
Sampling	Analog filtered	8000 samples of input signal caught pr. Second
Quantizing	Analog samples	Analog to digital conversion. 13 bit precision
Encoding	Compression	8000 compressions second to 8 bit = 64.000 bps



ISDN PRI

- Mainly used to connect Private telephone exchanges or IP phone system to PSTN
  – Call ange)
- For exam al 0 to get a public line
- Custome alls are routed through ISDN PRI

ISDN PRI up to 30 outgoing/incoming simultaneous calls in one ISDN PRI



in/out of band signalling

- ISDN signalling
  - signalling carried in the D channel
  - Voice/data carried in B-channel
  - Signal and voice separated in two channels or bands
  - ISDN is an example of out-of-band signalling
- POTS
  - Voice carried in the 300 – 3.400 Hz band
  - signalling – DTMF – carried in the same band
  - POTS is an example of in-band-signalling

VoIP signalling standards

VoIP – Voice over IP

VoIP signalling standards

- A variety of IP based signalling standards to VoIP defined – and used
  - SIP
  - H.323
  - MGCP
  - ....
- SIP and H.323 included in this course
- The purpose of VoIP signalling
  - Establish, maintain and close voice calls
  - Eventually collect information for taxation

signalling standards

- Two kinds of information necessary to phone
  - Call control (signalling phone to/from exchange)
  - Voice transfer (Between participating phones)
- In traditional telephony
  - Call Control (For example POTS or ISDN)
  - Voice transfer (Circuit switched channel)
- In VoIP – Voice over IP
  - Call control (For example SIP or H.323)
  - Voice transfer (Packet switched)

SIP and H.323

- H.323
  - Defined by ITU in 1996
    - International Telecommunication Union
  - Uses many technologies from PSTN
  - A suite of different protocols incl. voice/video
- SIP – Session Initiation Protocol
  - Defined by IETF in 1996
    - Internet Engineering Task Force
  - Uses many technologies from the Internet
  - Is a signalling protocol for voice/video

SIP SIGNALLING

Session initiation protocol

SIP

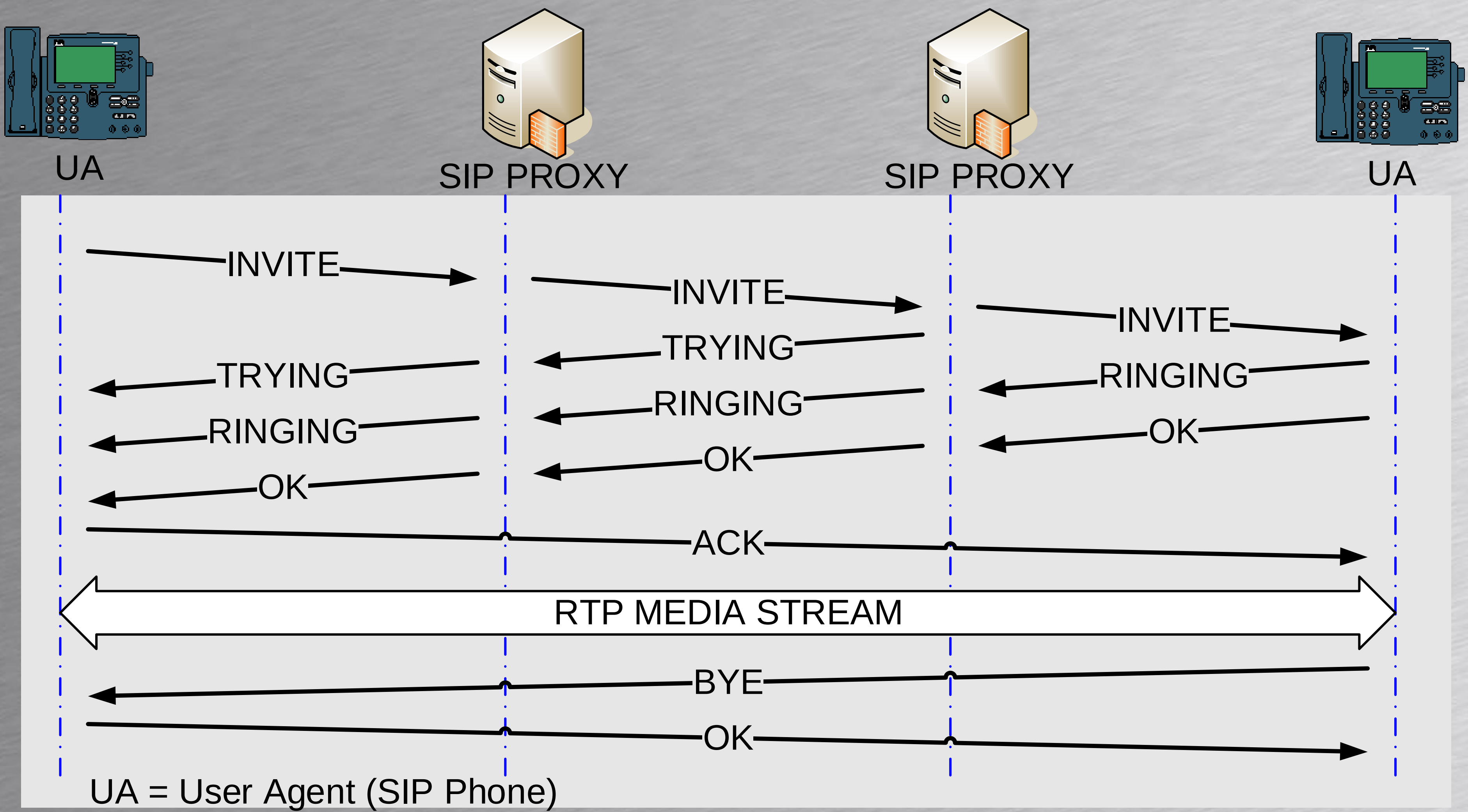
Session Initiation Protocol

- Defined by IETF in 1996
  - Internet Engineering Task Force
- Uses many technologies from the Internet
- Is a signalling protocol for voice/video
- SIP are widely used as VoIP signalling

SIP Design

- Expandable
  - Easy integration of new facilities
- (Not) a substitute for PSTN
  - SIP are used by many service providers
- Diversity of end-devices
  - Intelligent end-devices
- Multimedia
  - Voice, video….

SIP call flow example



SIP

- Is a application layer protocol
- Is a text based signalling protocol
  - Reminding of the HTTP and SMTP protocols
- Are used to establish, maintain and terminate multimedia sessions
- Support unicast and multicast sessions
- Location independent
  - Suitable for mobile users

SIP components

- There are two SIP components
- User Agents (UA) – SIP End-points (IP Phones) consisting of
  - User Agent Client (UAC): Initiates sessions
  - User Agent Server (UAS): Respond sessions
- SIP servers
  - Proxy server – Like a SIP IP-PBX
  - Registrar server – User Agent register
  - Redirect server – Redirect sessions

SIP addressing

- SIP addresses are identified by SIP URI’s
  - Uniform Resource Identifier
- Example

  sip:4103@mercantec.dk;transport=UDP
  - sip – The URI service
  - 4103 – The URI user part (extension/phone)
  - Mercantec.dk – The URI host part
  - ;transport=UDP – A URI parameter

SIP addressing

- Different formats of URI’s
  - FQDN’s: sip:heth@mercantec.dk
  - E.164: sip:30539361@mercantec.dk
  - E.164: tel:30539361
  - Mixed: sip:heth@194.123.12.23
- E.164 numbers in DNS

SIP messaging

- Messages contain header describing the communication in details
- Uses a text based syntax and header like HTTP
- Transport protocols TCP, UDP or SCTP
  - SCTP: Stream Control Transport Protocol
- Basically there are two kinds of messages
  - Requests: Sent by Clients
  - Response: Sent by servers

SIP messaging

- Request message header

Method	Request URI	SIP version

  - Method: INVITE, ACK, BYE ….
  - Request URI: The receiver of the request
  - SIP version: Used SIP version (2.0 in use)
- Response message header

SIP version	Status code	Reason phrase

  - SIP version: Used SIP version (2.0 in use)
  - Status code: An integer describing the answer
  - Reason Phrase: A text description of status code

SIP requests methods

Command	Function
INVITE	Used to establish a media session between user agents. Starting a call
ACK	Confirms reliable message exchanges
BYE	Terminates a established session between agents
CANCEL	Terminates a pending request
OPTIONS	Requests information about the capabilities of a caller, without setting up a call
REGISTER	Used by a user agent to register to registrar. The user agent informs the registrar of its IP address and URI.

NOTE: There are defined more SIP request methods than indicated in the above table. The shown methods are from RFC 3261

SIP response codes

With examples

Respons class	Status code	Reason phrase
Info	**1xx**	**Provisional responses**
	100	Trying
	180	Ringning
Succes	**2xx**	**Succesful responses**
	200	OK
Redirect	**3xx**	**Redirection responses**
	302	Moved temporarily
Client error	**4xx**	**Client failure responses**
	401	Unauthorized
Server error	**5xx**	**Server failure responses**
	503	Service unavailable
Global failure	**6xx**	**Global failure responses**
	603	Decline

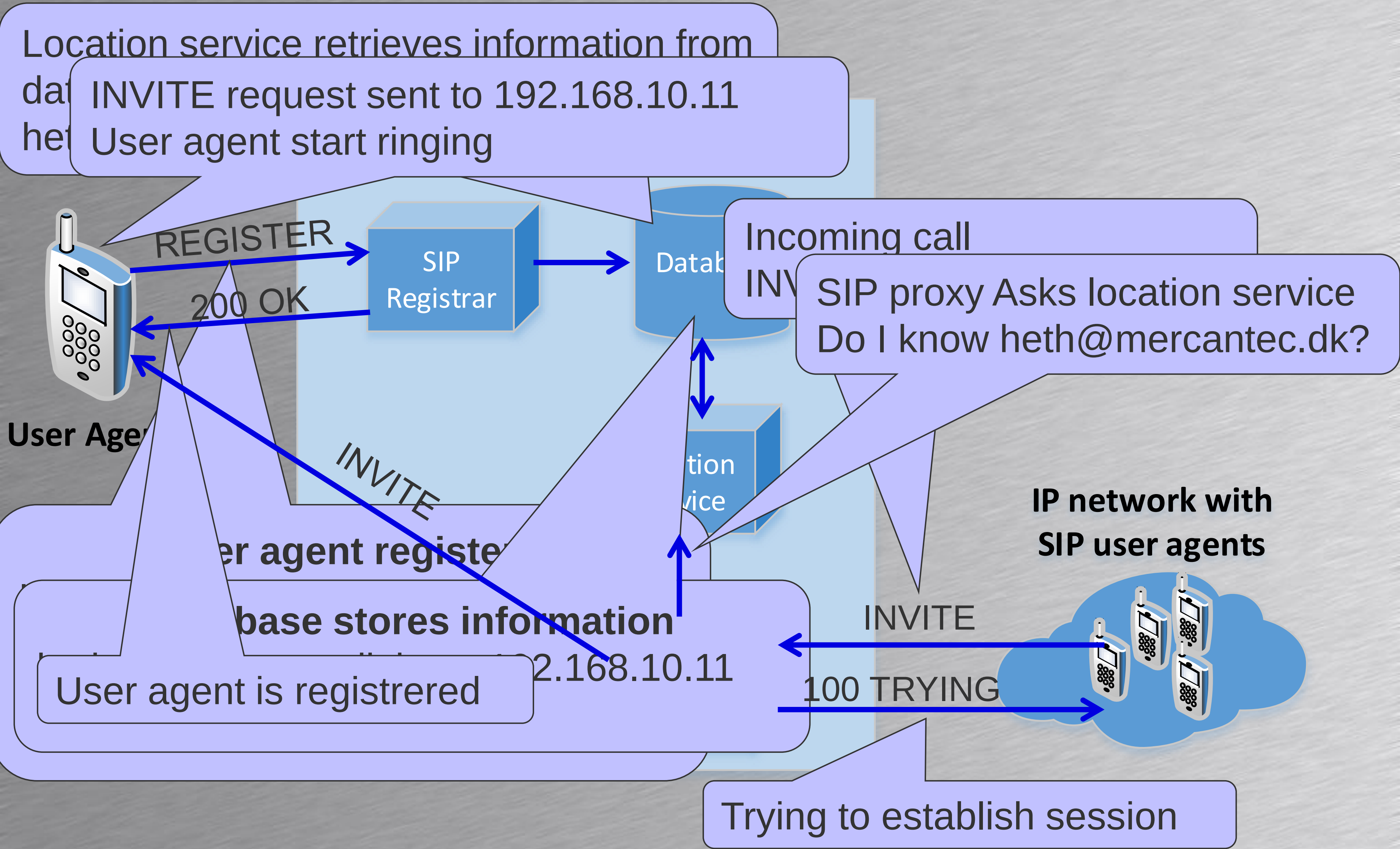
SIP User Agent – UA

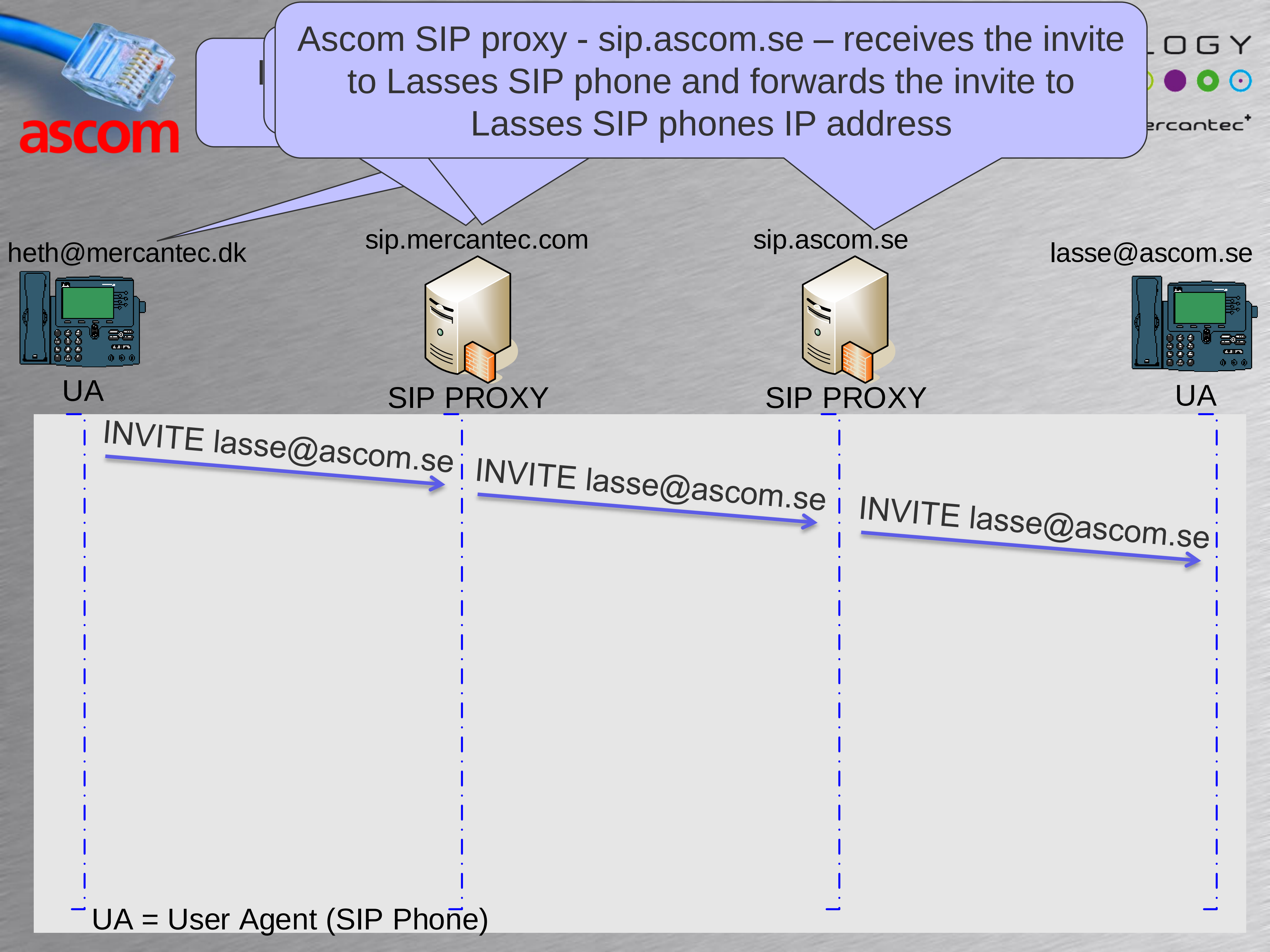
- A User Agent is a software component acting on behalf of a user
- The SIP phone soft or hard is a SIP UA
- User Agent Client – UAC
  - Initiates SIP sessions when calling
  - Send SIP requests
- User Agent Server – UAS
  - Accepts or rejects SIP session requests
  - Receives SIP requests and return SIP response

SIP servers

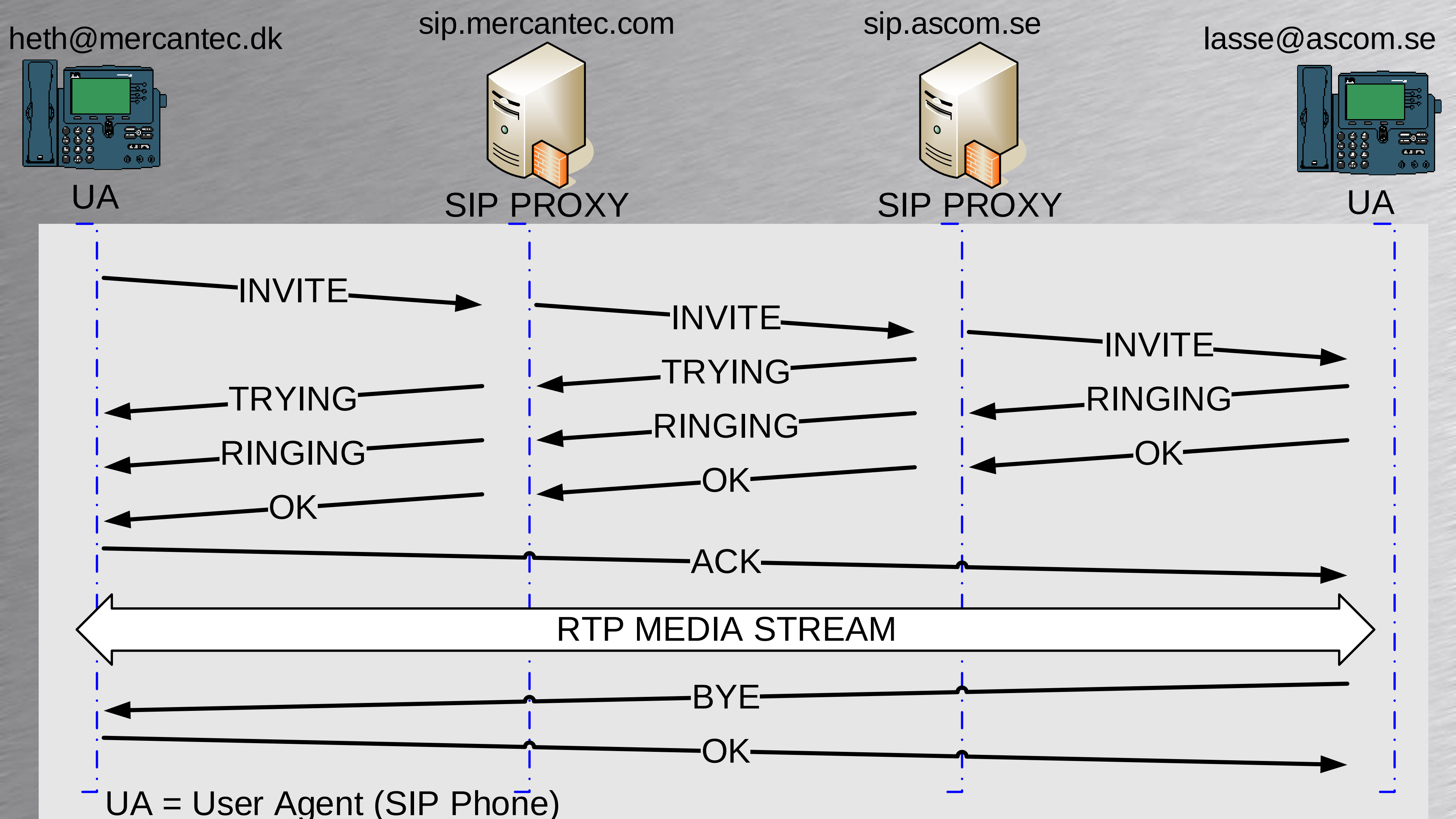
- Proxy server
  - Receive and sends SIP request and responses on behalf of the User Agents. (UA)
- Registrar server
  - User Agents register to the registrar server
  - Keeps track on User Agents URI and IP addresses and acts as a location service
- Redirection Server
  - Returns “contact this address” responces
- The servers often live on the same “box”

SIP servers

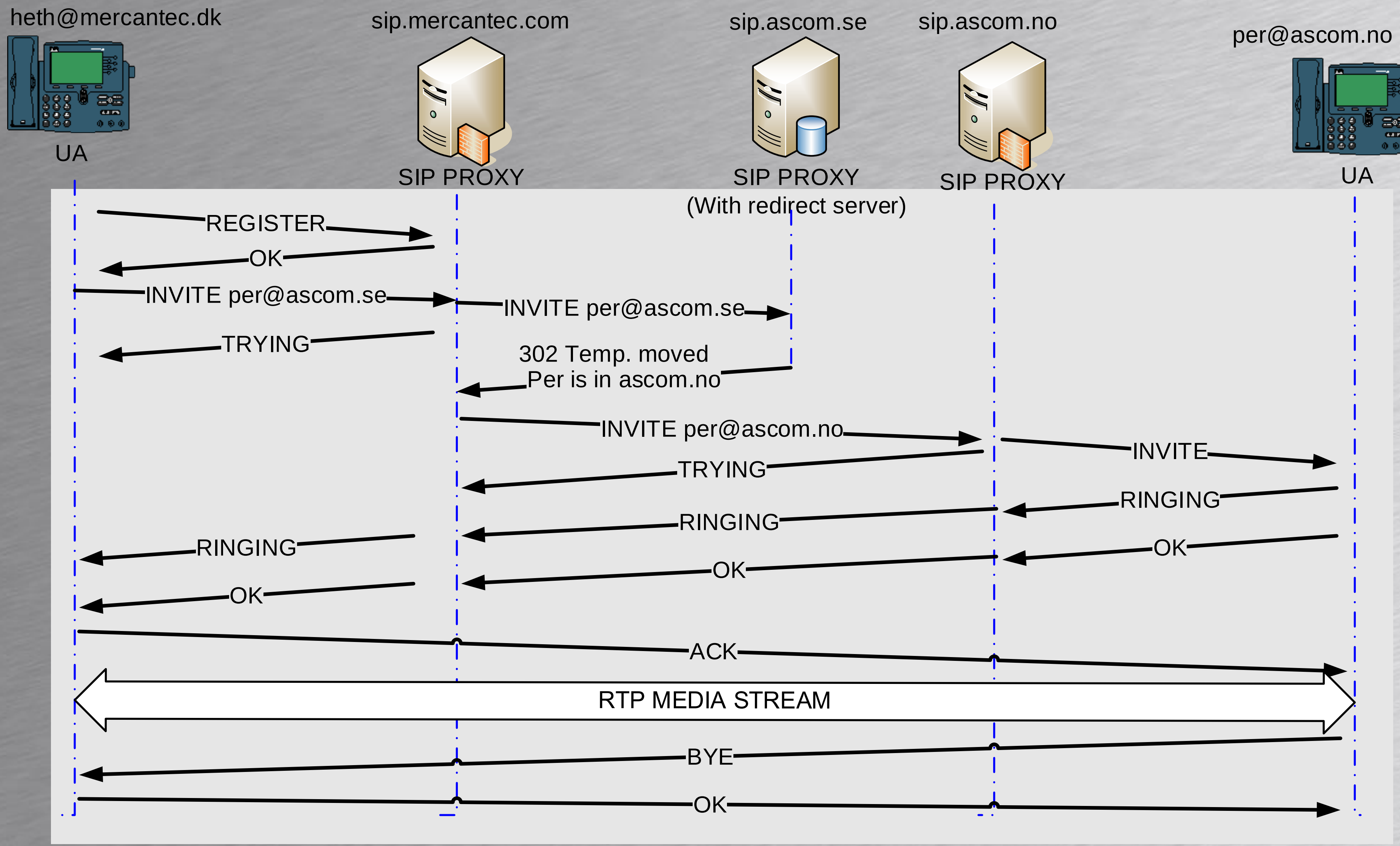

Location service retrieves information from
dat
het
INVITE request sent to 192.168.10.11
User agent start ringing
REGISTER
200 OK
SIP
Registrar
Datab
Incoming call
INV
SIP proxy Asks location service
Do I know heth@mercantec.dk?
User Age
INVITE
tion
vice
er agent registe
base stores information
2.168.10.11
User agent is registrered
IP network with
SIP user agents
INVITE
100 TRYING
Trying to establish session


Ascom SIP proxy - sip.ascom.se – receives the invite to Lasses SIP phone and forwards the invite to Lasses SIP phones IP address
ascom
heth@mercantec.dk
sip.mercantec.com
sip.ascom.se
lasse@ascom.se
UA
SIP PROXY
SIP PROXY
UA
INVITE lasse@ascom.se
INVITE lasse@ascom.se
INVITE lasse@ascom.se
UA = User Agent (SIP Phone)

SIP Proxy Server



SIP Redirect



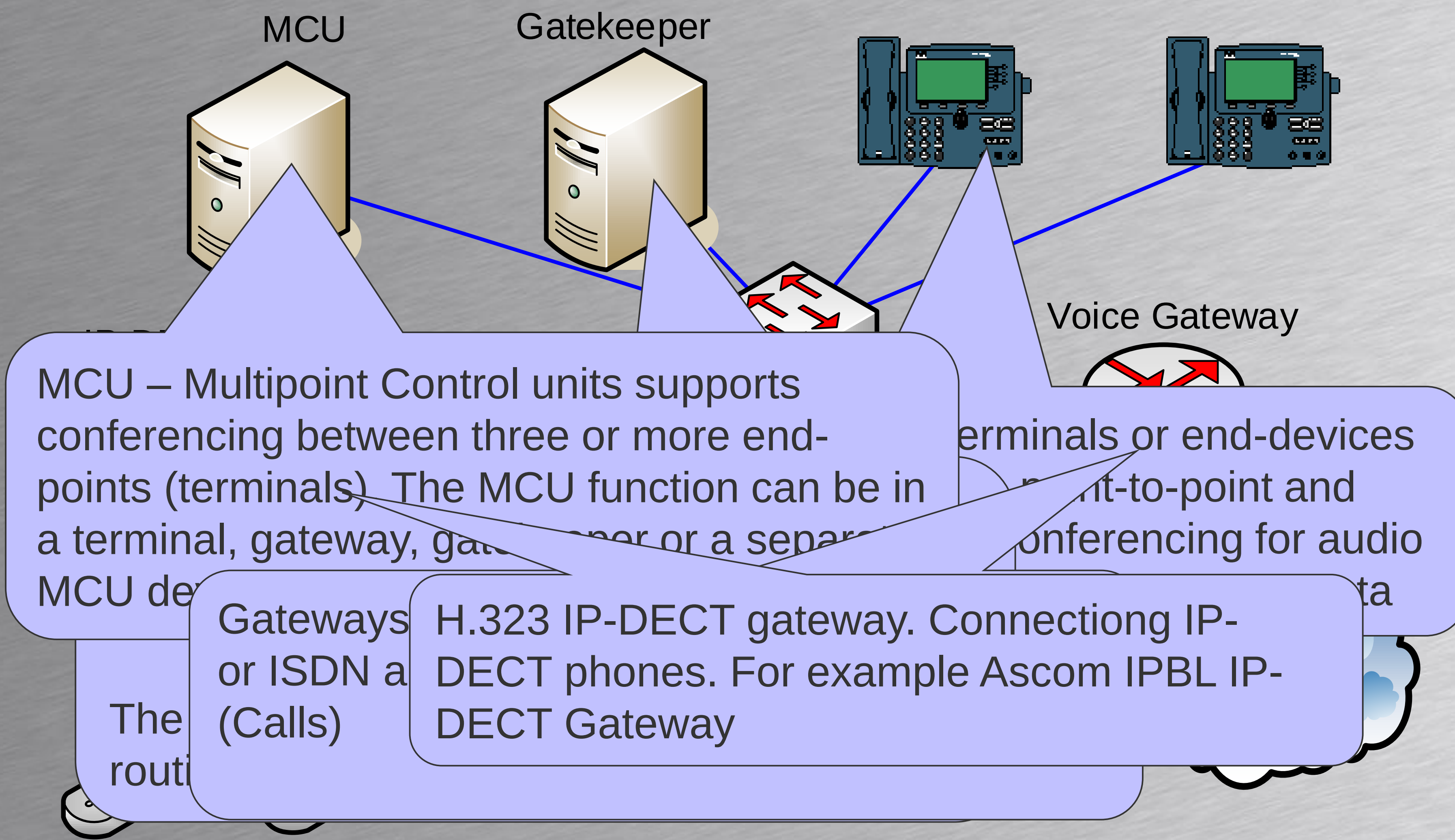
Task

- Connect FreePBX
- Flash phones
- Create phones and create extensions

- Phone Boot?

H.323 SIGNALLING

H.323 terms



DECT – Digital Enhanced Cordless Telecommunications
PSTN – Public Switched Telephone Network

H.323 functions and devices

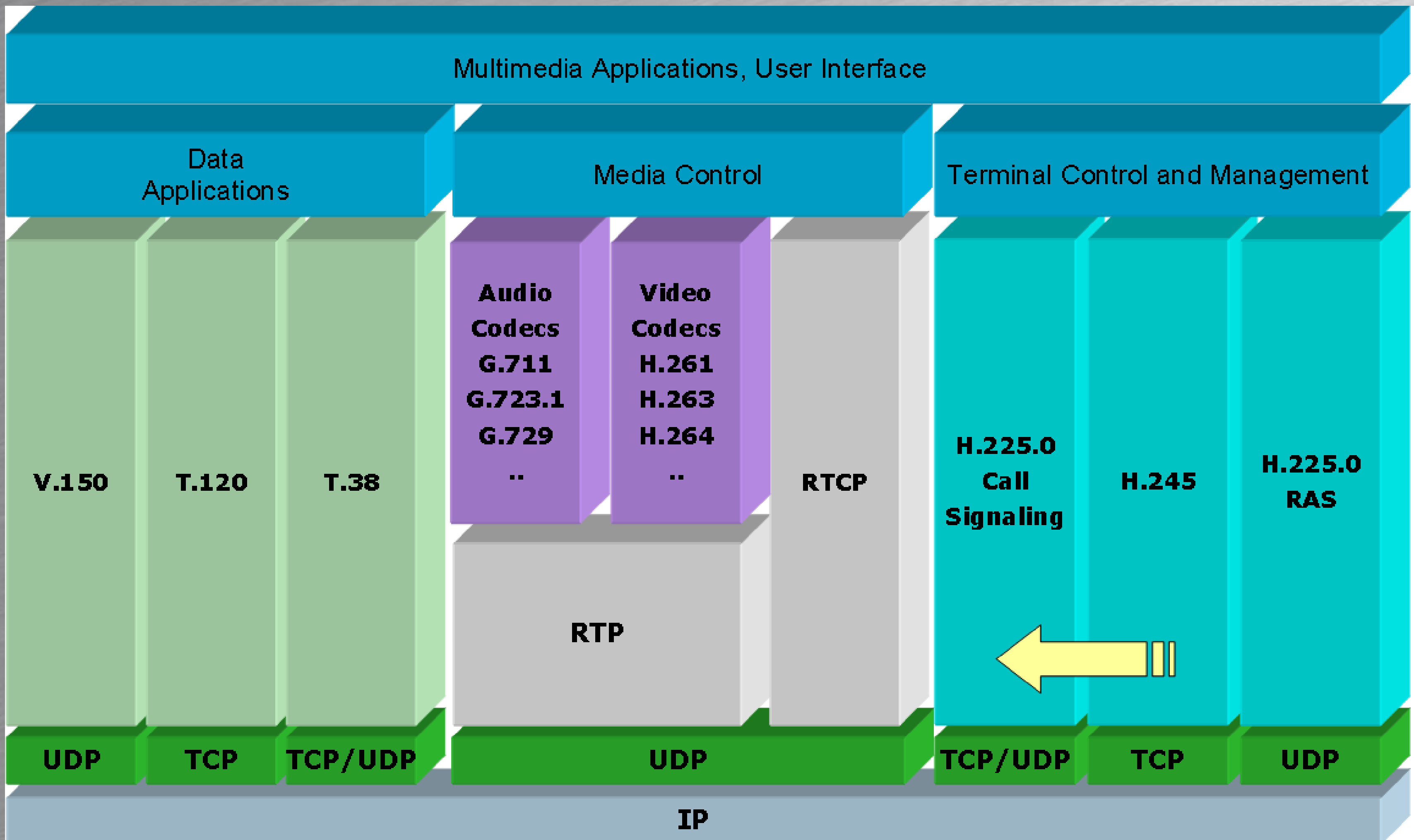
Function	Description
Terminal	Also called end-device. In daily speech called H.323 IP phone
Gatekeeper	Main function is to provide call routing. (Establish call sessions) In daily speech referred to as the gatekeeper or H.323 IP PBX
MCU	Multipoint Control Unit – used with conferencing
Gateway	Connects two dissimilar networks. For example connects a H.323 network with the PSTN or IP-DECT

H.323

- H.323 is a suite of protocols
  - Each with its own defined purpose

Protocol	Description
H.225.0	RAS – Registration, Admission and status
H.225.0	Call signalling (Based on ISDN’s Q.931 signalling)
H.245	Control protocol for multimedia communication (Non telephone)
RTP	Real Time Transport Protocol – Carries voice/video between end-points
H.235	Security and encryption standard for H.323 devices
H.239	Used for dual-streams for example video conferencing
H.450	Supplementary services (Call transfer, hold, waiting call …..)
H.460	Extensions to H.323 videoconferencing standard (NAT traversal …..)

H.323 overview



H.225 RAS signalling

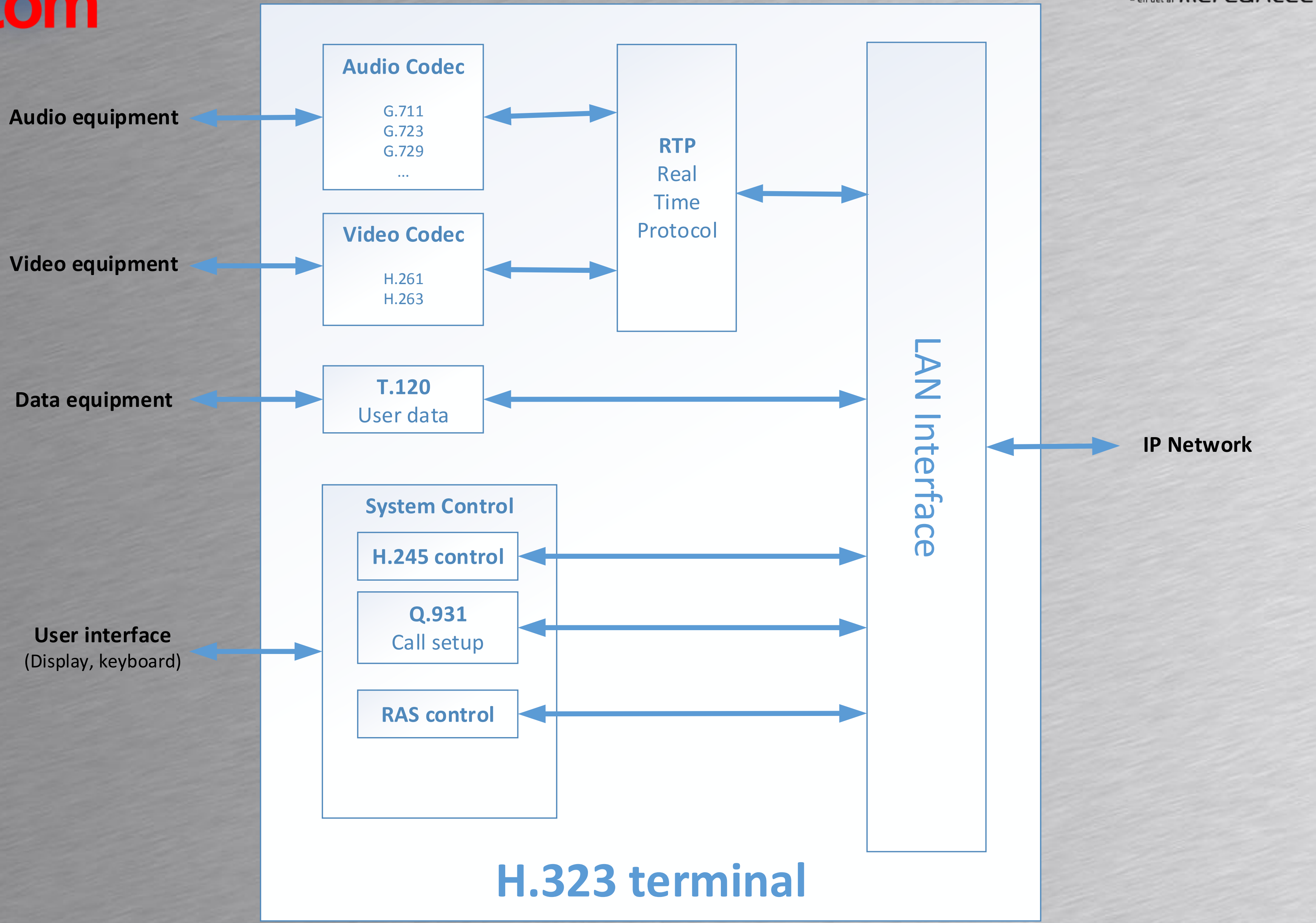
- RAS – Registration Admission and Status
  - Pre-call control in H.323 networks
- Registration:
  - Enables end-points and gateways to register to a gatekeeper
- Admission:
  - Admission messages between end-points and gatekeepers provide the basis for call control
- Status
  - Gatekeeper can obtain status information from end-points. (Check if on-line)

Gatekeeper discovery

H.225 RAS signalling

- End-points must register to gatekeepers
  - The gatekeeper must know all end-points
- End-points need to know IP address of the gatekeeper or gatekeepers
- Manual or automatic gatekeeper discovery
  - Manual: end-points have manually entered IP address/hostname for gatekeeper
  - Automatic: End-points use IP multicast to discover gatekeepers

H.323 terminal or end-point

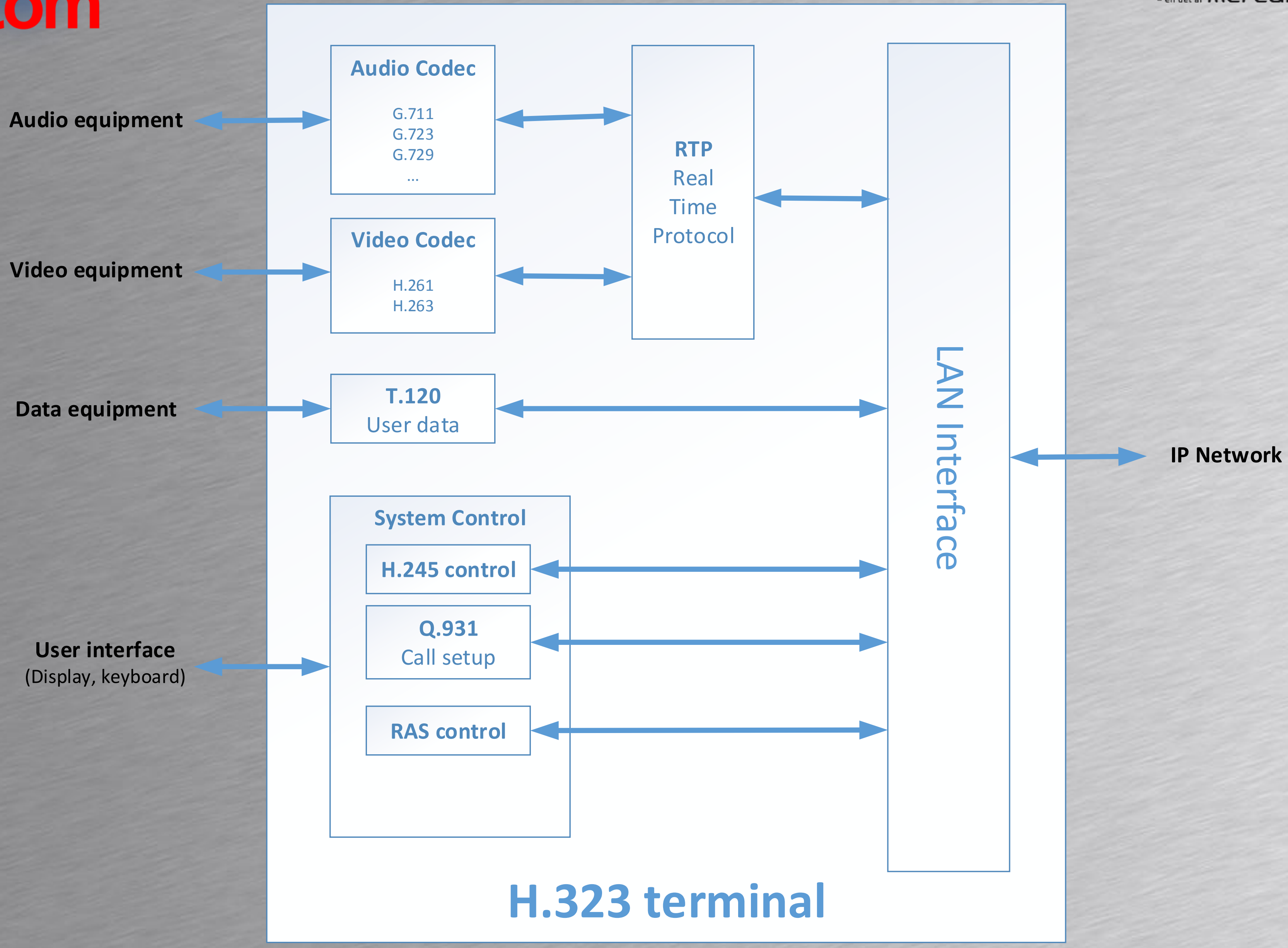


H.323 port numbers

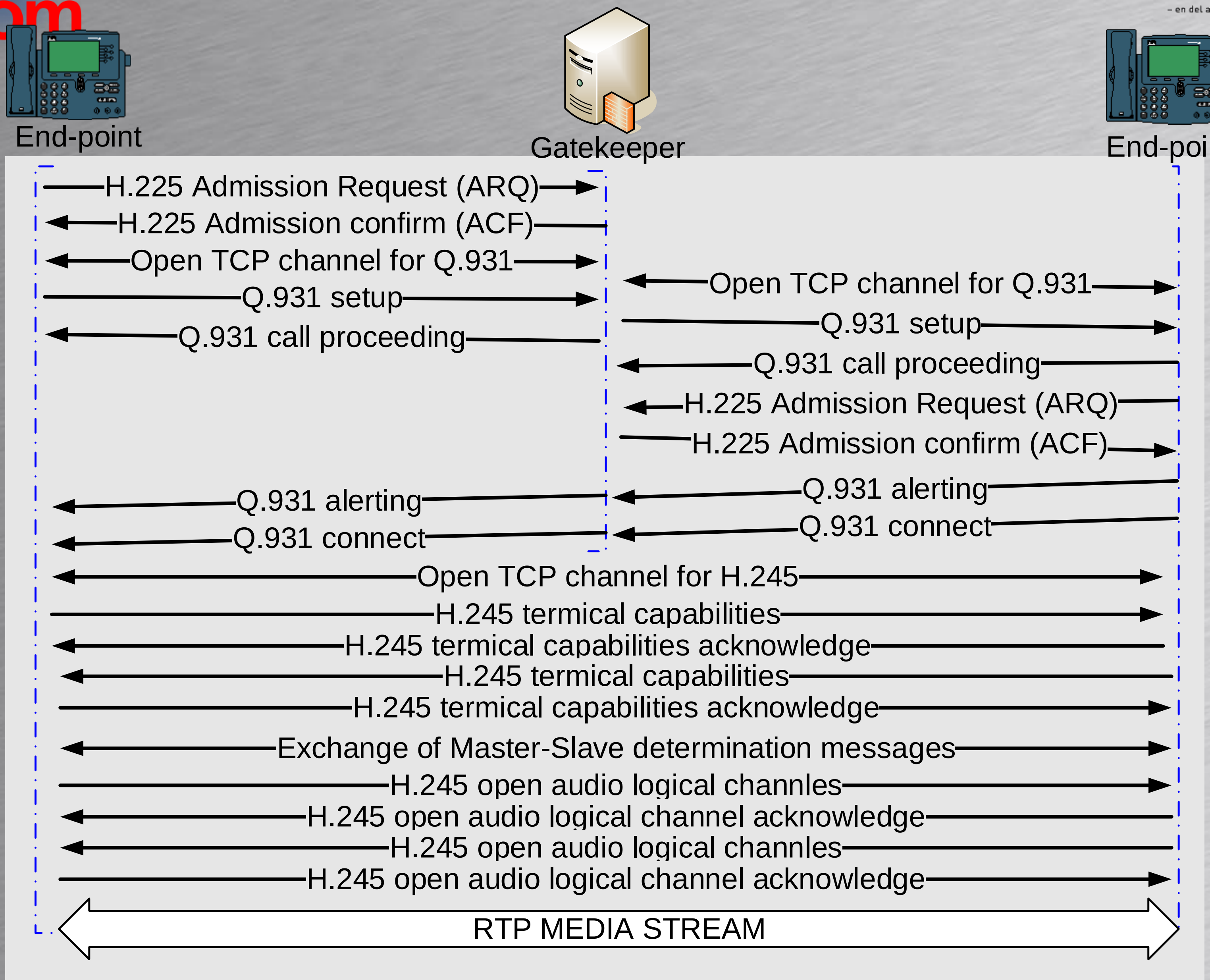
- Protocols and port numbers used

Protocol	Port number	Description
H.225.0	UDP/1718	RAS Multicast gatekeeper discovery port (224.0.1.41)
H.225.0	UDP/1719	RAS Gatekeeper Unicast registration and status port
H.225.0	TCP/1720	Call signalling
H.245	TCP Dynamic 1024 – 65535	Control protocol for multimedia communication
RTP	UDP Dynamic 1024 – 65535	Real-time Transport Protocol

H.323 terminal



H.323 call flow

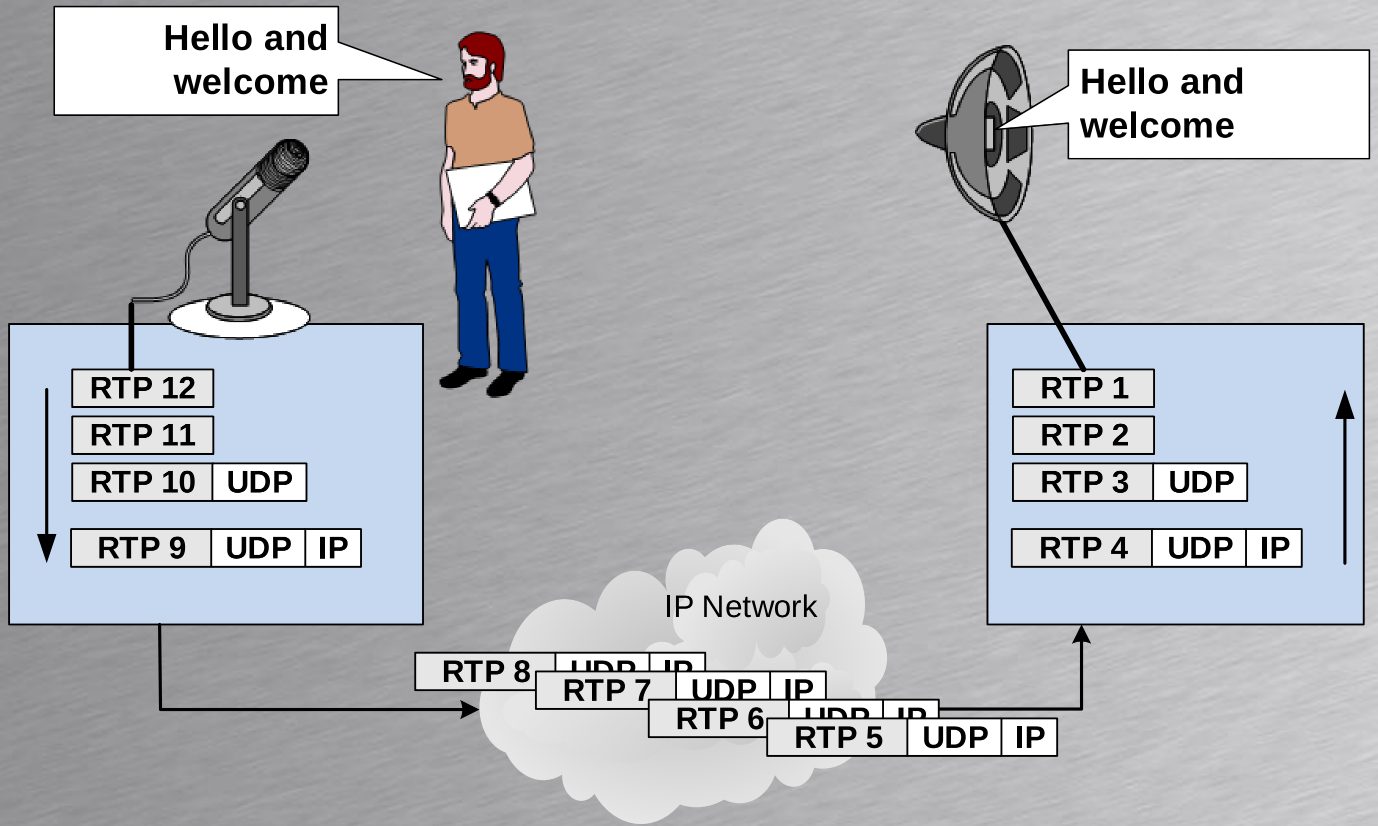


RTP FAMILY

Real-time Transport Protocol

Voice quality

- Low constant delay is important for a good voice quality with low delay

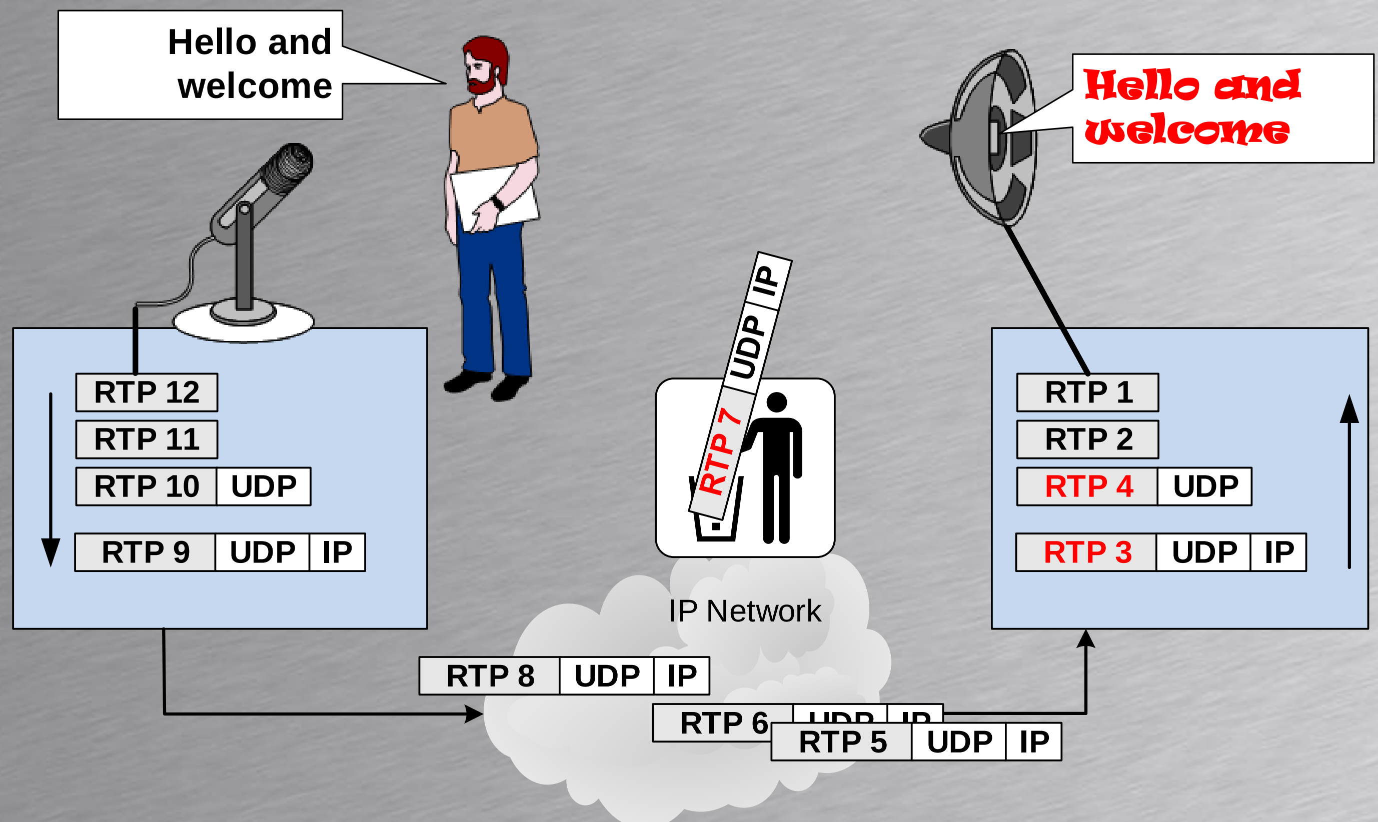


Packet switched networks

– No guaranty of constant delay in packet !

– No guaranty of orderly delivery of packets !

– No guaranty packets are delivered !



RTP

- Designed for end-to-end real-time transfer of data
- High lights
  - Primary standard for audio/video transport
  - Jitter Compensation
    - Variable delay between packets is called Jitter
  - Detection of out of sequence packets
  - Possibility for multicasting
  - Transport via UDP or TCP
  - Secure version available SRTP with encryption

RTP

- Real-time multimedia streaming
  - Can tolerate small amounts of lost packets
    - Rule of thumb in VoIP is up to 1% packet loss
    - Receiver uses an error concealment algorithm
  - Receiver uses small jitter-buffer
    - Delays playback a little allowing a little jitter
    - 20 mS often used in VoIP
  - Requires an optional signalling protocol
    - Such as SIP or H.323 to establish a session

RTP

- Most implementations of RTP uses UDP
  – If data is not present in the “magic” moment the data is useless
- Reliability and timely delivery is the responsibility of the IP network
  – QoS necessary for good voice quality

Ethernet / frame header	IP header 20 bytes	UDP header 8 bytes	RTP header 12 bytes	Payload (Voice,video …) 20 -160 bytes

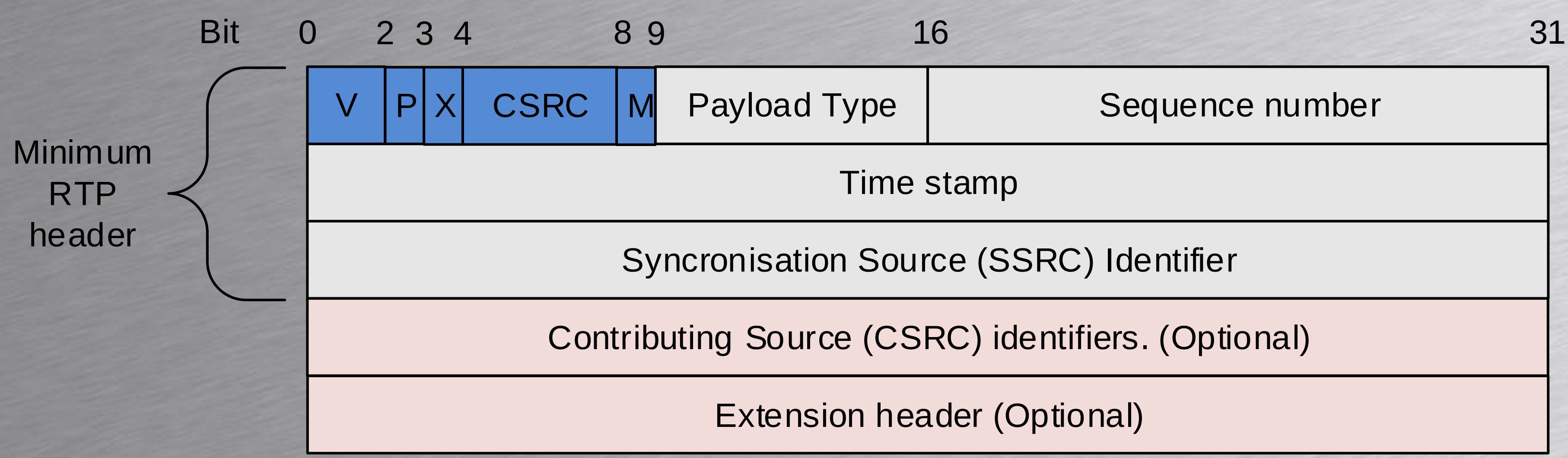
RTP header

- The RTP header consist minimum of 12 bytes
  - – 32 bits or 4 bytes in each line

Bit	0	2	3	4	8	9	16	31
Minimum RTP header	V	P	X	CSRC	M	Payload Type	Sequence number	
Minimum RTP header	Time stamp							
Minimum RTP header	Synkronisation Source (SSRC) Identifier							
	Contributing Source (CSRC) identifiers. (Optional)							
	Extensions (Optional)							

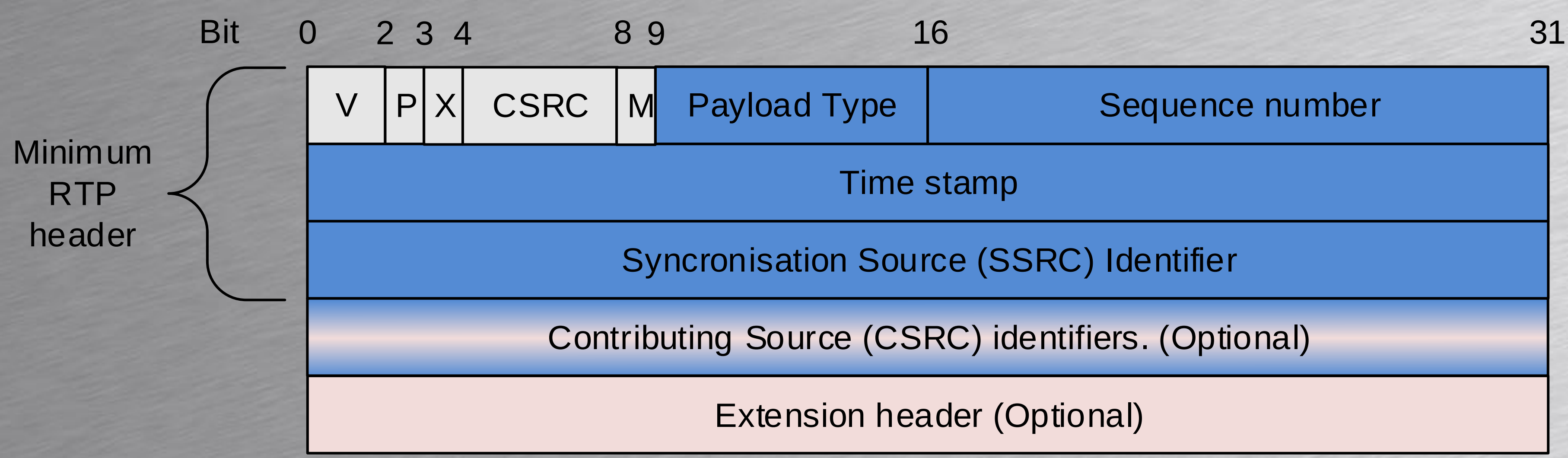
RTP header

Field	Bits	
V	2	Version. Current standard 2
P	1	Padding – Indicating extra fill bytes to fill up a block to a certain size
X	1	Optional eXtension header present
CSRC	4	Optional CSRC present count (1 to 15)
M	1	Marker – Indicating this packet has special meaning. For example can be used to synchronize clock rates between peers



RTP header

Field	Bits	
Payload type	7	Which kind of data carried – Codec
Sequence nr.	16	Incremented in each packet. Used by the receiver to detect packet loss and restore packet order
Time stamp	32	Used by the receiver to start playback at appropriate time
SSRC	32	Uniquely identifies the source of a particular stream
CSRC	´n X 32	Optional field identifying if more contributing sources (For example a voice conference)



RTP media stream example

- VoIP conversation profile
  - G.711 a-Law codec used
  - RTP time interval 20 mS = 0,02 second
- a-Law bit rate = 64.000 bps = 8.000 Bps
- RTP payload data pr. Packet
  - 8000 bytes/second * 0,02 second = 160 bytes
- Packet size: 20 + 8 + 12 + 160 = 200 Bytes
  - Without OSI layer 2 overhead (ethernet frame)

IP header 20 bytes	**UDP** header 8 bytes	**RTP** header 12 bytes	Payload a-Law 160 bytes

RTP media stream example

- 0,02 second between packets
  - 1 / 0,02 = 50 packets pr. Second
- Each packets is 200 bytes
  - 50 packets * 200 byte * 8 = 80.000 bps
- VoIP streams can be bidirectional
  - Network load = 2 * 80.000 bps = 160.000 bps

IP header	**UDP** header	**RTP** header	Payload a-Law
20 bytes	8 bytes	12 bytes	160 bytes

Silence suppression

- Voice conversations are often half duplex
  - One listen while the other talks
- No reason for sending RTP packets with background noise only
  - Silence suppression stops RTP transmission
    - Silence suppression is an option
  - Receiving phone makes comfort noise
    - Speaker has a feeling of the connection
- Saves bandwidth – important in trunks

RTP family

- RTP family of protocols

Protocol	Description
RTP	End-to-end transfer of real-time stream data (Unencrypted)
RTCP	Provides statistics and control information for an RTP media stream
SRTP	End-to-end transfer of real-time stream data (Encrypted)
ZRTP	Cryptographic key-exchange protocol between VoIP end-points

Protocol	Name	Date standardized
RTP	Real-time Transport Protocol	1998
RTCP	Real-time Transport Control Protocol	1998
SRTP	Secure Real-time Transport Protocol	2004
ZRTP	Zimmerman Real-time Transport Protocol	2006

RTCP

Real-time Transport Control Protocol

- Related to RTP
- Provides statistics and control information for an RTP media stream
- Sender report – SR
  - The RTP sender periodically sends SR reports including the sent and received RTP packets
- Receiver report – RR
  - Send by receive-only participants. Those who do not transmit RTP packets.

SRTP

Secure Real-time Transport Protocol

- A version of RTP providing security
- Encryption
  - Only the intended receiver can read the data
- Authentication
  - Identity check of other party
- Integrity
  - Data cant be changed in transit
- Anti replay
  - Data cant be recorded and retransmitted

SRTP

Secure Real-time Transport Protocol

- Uses standard security features
  - Defaults to AES for encryption
  - HMAC-SHA1 for authentication and integrity
- The security standards are out of scope in this course module
- A new version of RTCP providing security
  - SRTCP – Secure Real-time Control Protocol

ZRTP

Zimmermann Real-time Transport Protocol

- Cryptographic key-agreement protocol
- Can be used without a PKI or certificates
  - Public Key Infrastructure
- Uses RTP as transport channel
- signalling protocol unaware of ZRTP
- Negotiate keys between end-points
  - Uses Diffie-Hellman key exchange

THE IP PHONE

Hard and soft phones

IP phones

- Soft, hard, wired and wireless
- Standardized signalling protocols
  - For example H.323 and SIP
- Vendors add on functionality
  - Not 100% comparability between vendors
- Configuration of IP phones
  - Manually
  - Automatically – Vendor specific configuration

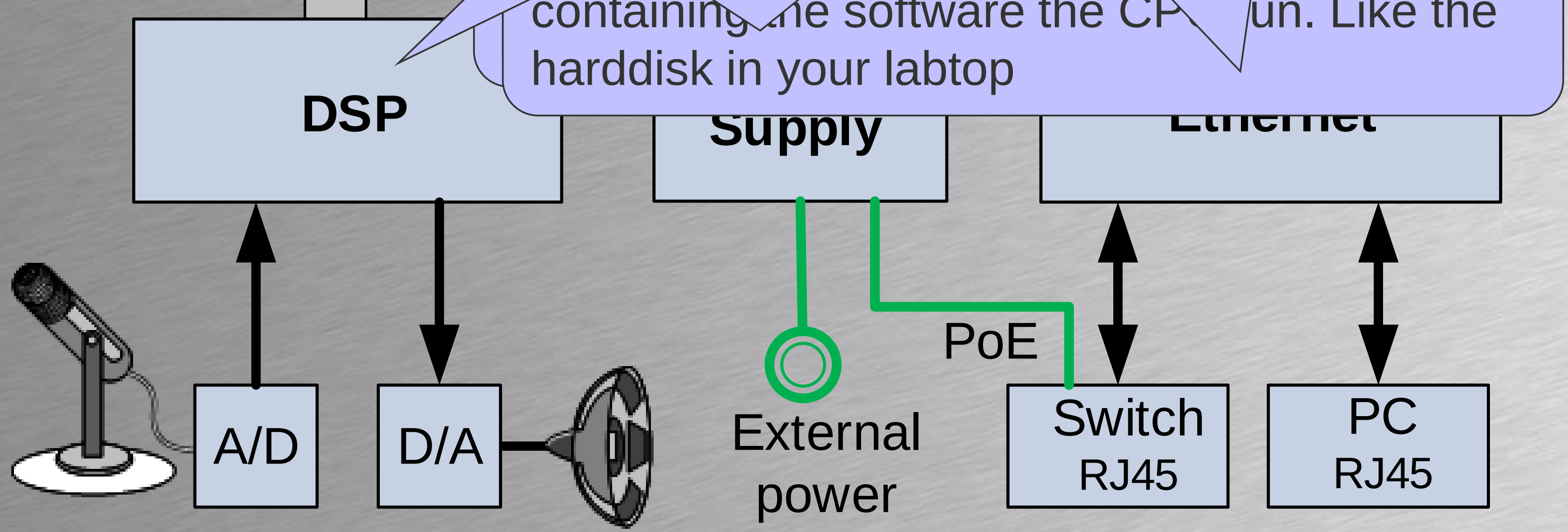
Hard phone parts

Power supply: A IP phone is really a computer disguised to look like a phone and need power.

To power options ofte ethernet – PoE – or external ... ugh an power adapter.

DSP: Digital Signal Processor – A dedicated processor used to process voice and contain the Codec and echo cancelling

... torage memory ... he phone containing ... e software the CP ... un. Like the harddisk in your labtop

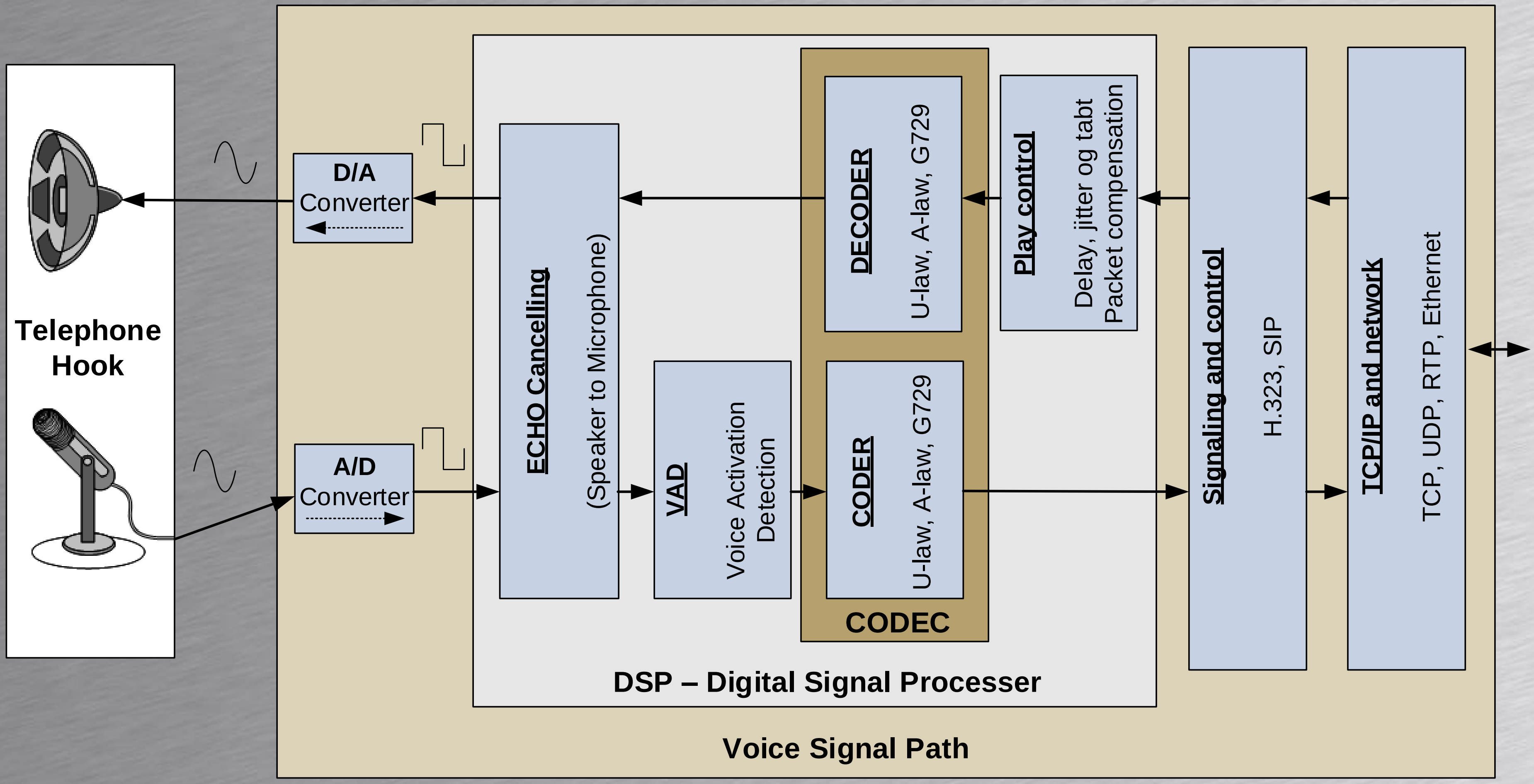


DSP

Digital Signal Processor

- A DSP is a microprocessor designed to process analog signal such as voice and video
- The analog signal has to be converted to digital by a A/D converter. (Analog to Digital converter)
- The DSP offload the CPU for working with the signal processing.
- A normal CPU can be used for signal processing
  - CPU's are not optimized for signal processing
  - All running programs share the same CPU. (Jitter)
- CPU's in PC's use Intels MMX instruction set
  - Special instructions for multimedia processing
- A dedicated DSP guaranties processing

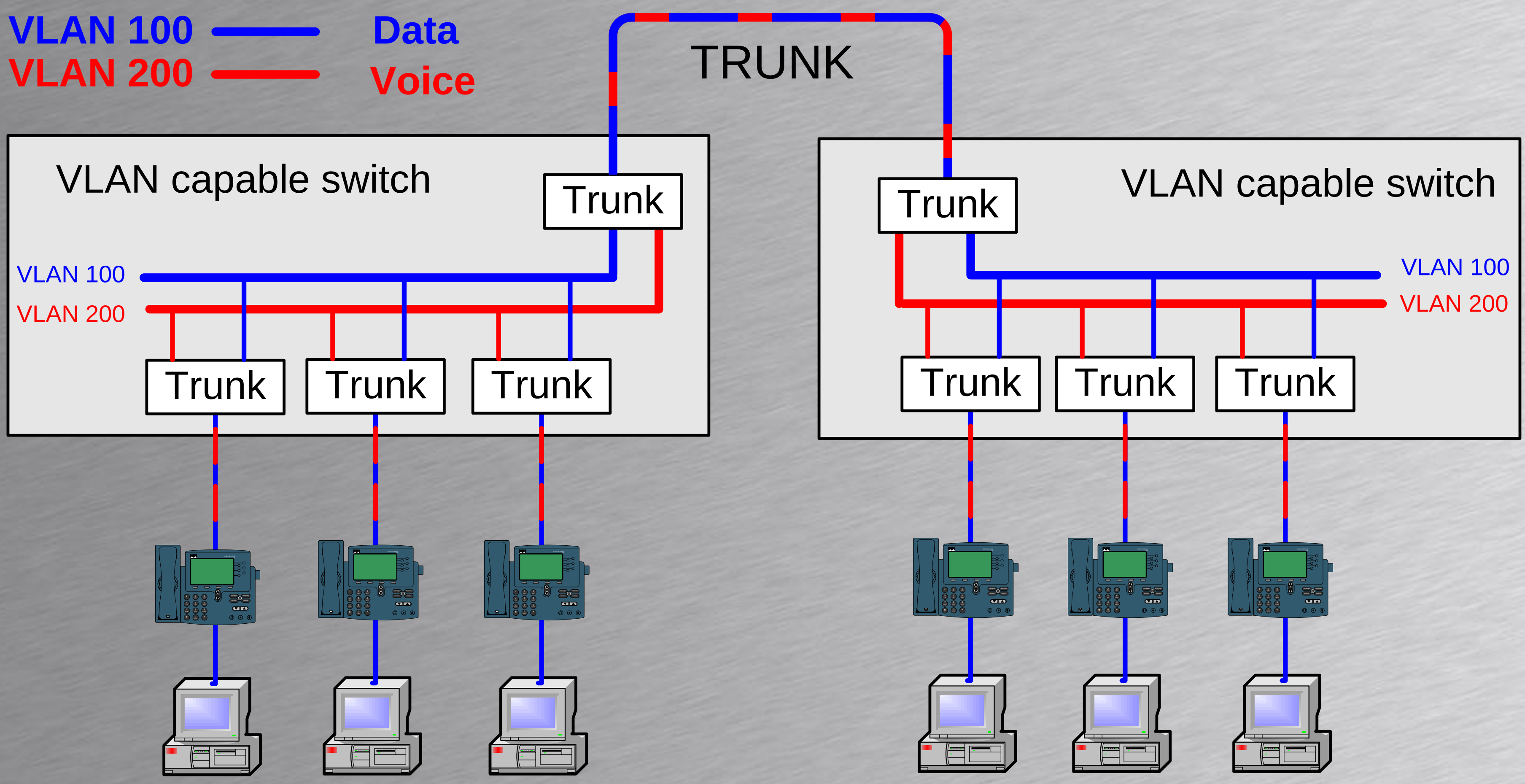
DSP – Voice signal Path



Voice VLAN

– Data traffic carried in VLAN 100

– Voice traffic carried in VLAN 200



PoE

Power over Ethernet

- Most IP phones are powered by 48 Vdc
- Many IP phones can get power from
  - External power supply connected to mains
  - From switches capable of delivering power
    - PoE or Power over Ethernet
    - Picture below is a partial printout from a PoE capable switch

mars.tekkom.dk - PuTTY

```
Campus1#show power inline
Available:280.0(w)  Used:44.1(w)  Remaining:235.9(w)

Interface Admin  Oper       Power   Device              Class Max
                            (Watts)
--------- ------ ---------- ------- ------------------- ----- ----
Fa0/1     auto   on         6.3     IP Pnone 7940       2     15.4
Fa0/2     auto   on         6.3     IP Pnone 7940       2     15.4
Fa0/3     auto   off        0.0                         -     15.4
```

PoE
Power over Ethernet

- Two standards
  - 802.3af supplying device with up to 12,95 Watt
  - 802.3at supplying device with up to 25,50 Watt
- Can be used with 10,100 and 1000 Mbps ethernet
- Some vendors invented their own version of PoE before IEEE standards
  - IEEE 802.3af from 2003
  - IEEE 802.3at from 2009 also called PoE plus

ascom

PoE
Power over Ethernet

HOUSE OF TECHNOLOGY

– en del af mercantec+

Pin RJ45	T568A RJ45	Pair	10/100 Spare Mode B	10/100 mixed Mode A	1000 Mbps Mode B	1000 Mbps Mode A
1		3	Rx+	RX+ DC+	TxRx A+	TxRx A+ DC+
2		3	Rx-	RX- DC+	TxRx A-	TxRx A- DC+
3		2	Tx+	TX+ DC-	TxRx B+	TxRx B+ DC-
4		1	DC+	Unused	TxRx C+ DC+	TxRx C+
5		1	DC+	Unused	TxRx C- DC+	TxRx C-
6		2	Tx-	TX- DC-	TxRx B-	TxRx B- DC-
7		4	DC-	Unused	TxRx D+ DC-	TxRx D+
8		4	DC-	Unused	TxRx D- DC-	TxRx D-

Abbr.	Explanation
Rx+, Rx-	Receive plus and minus signal (Differential on pair)
Tx+, Tx-	Transmit plus and minus signal (Differential on pair)
DC+, DC-	PoE DC delivered common mode in pair. Plus or minus in pair.

WIRESHARK

Packet analyzer

Wireshark

- Capture SIP conversations
- Create a visual document explaining the packet and the flow in a conversation.