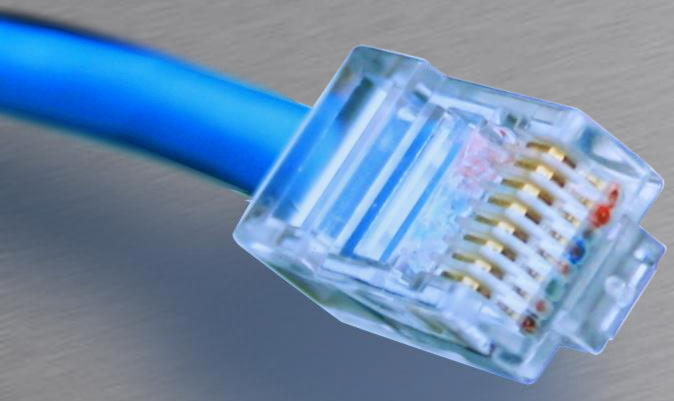
IP Telephony



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IPT Intro

Telephony principles and VoIP

Subjects

- Basic Telephony PSTN/POTS/ISDN.
- ITU E.164 numbering plan.
- Circuit switched vs. packet switched telephony.
- Basic signaling in-band/out-of-band (DTMF/Q.931).
- Codec operation (G.726, G.711, G.722, G723, G.729).
- Wireshark

BASIC TELEPHONY

- PSTN
- POTS
- ISDN

PSTN

Public switched telephone network

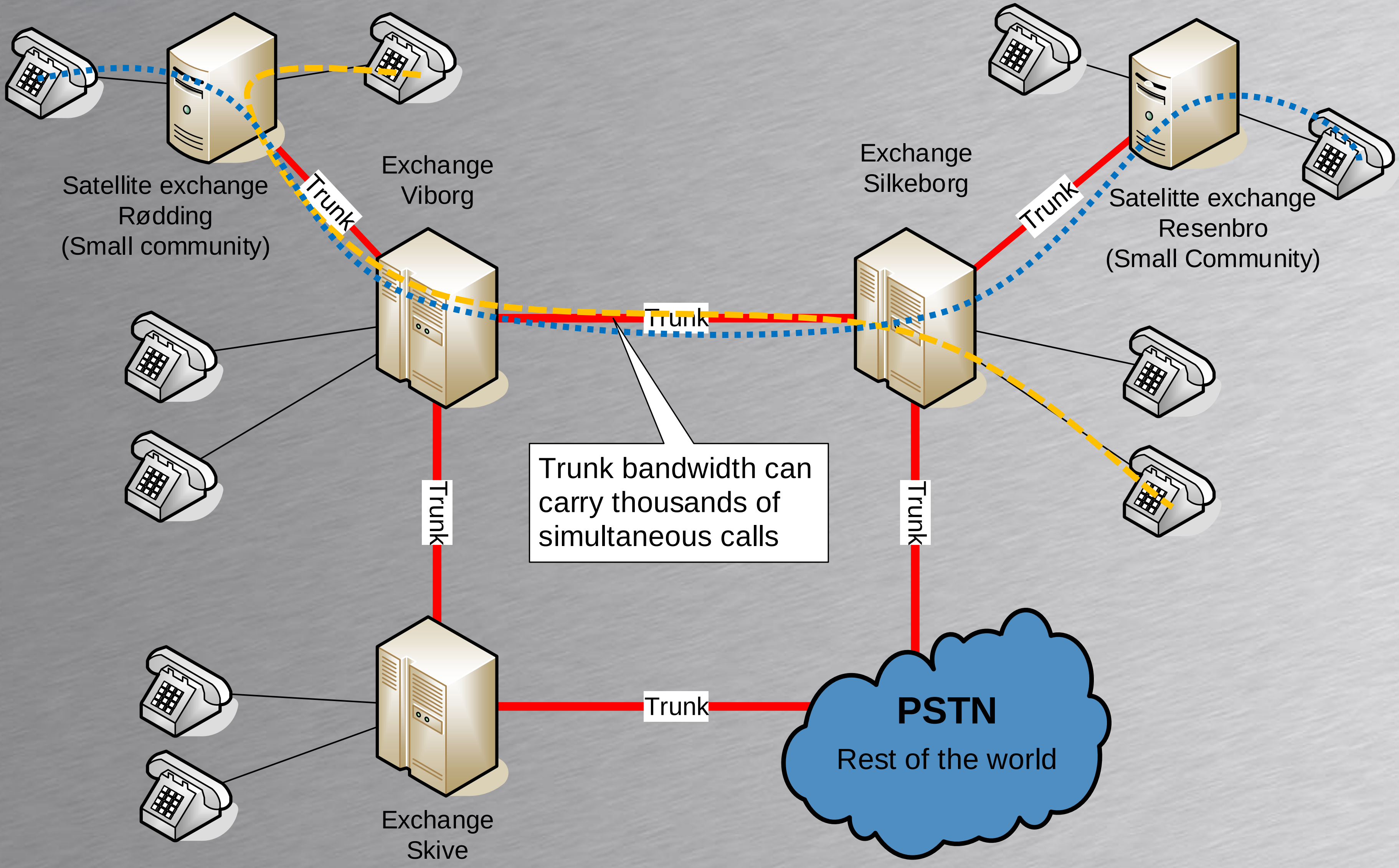
- PSTN or public switched telephone network
  - Worlds public switched telephone network
  - Connecting phones worldwide
  - Share common international standards (ITU)
    - International Telecommunication Union
  - ITU Telecom (ITU-T) is a subdivision of ITU
  - Share common numbering plan (ITU-T E.164)
    - For example: +45 48198283

PSTN

Public switched telephone network

- PSTN consists of
  - Telephone lines for fixed phones
  - Cellular networks for mobile phones
  - Fiber optic cables
  - Microwave transmission links
  - Communication satellites
  - Undersea telephone cables
  - Telephone switches (Exchanges)

PSTN


Satellite exchange
Rødding
(Small community)
Exchange
Viborg
Exchange
Silkeborg
Satellitte exchange
Resenbro
(Small Community)
Trunk
Trunk
Trunk
Trunk
Trunk
Trunk
Trunk bandwidth can carry thousands of simultaneous calls
Exchange
Skive
PSTN
Rest of the world

POTS

Plain old telephone service

- A part of PSTN
- Connects subscriber to PSTN
- Available since late 19th century
  - Graham Bell introduced his telephone in Europe in 1877
- Not much change since ☺

POTS

Plain old telephone service

- Offer bidirectional analog voice
  - Full duplex communication
- Voice band 300 Hz to 3.400 Hz
  - Human voice in limited frequency range
  - HIFI systems offer 20 Hz to 20.000 Hz
- Call progress
  - Dial tone and ringing signal
- Subscriber dialing
  - Able to dial other subscriber

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POTS

Plain old telephone service

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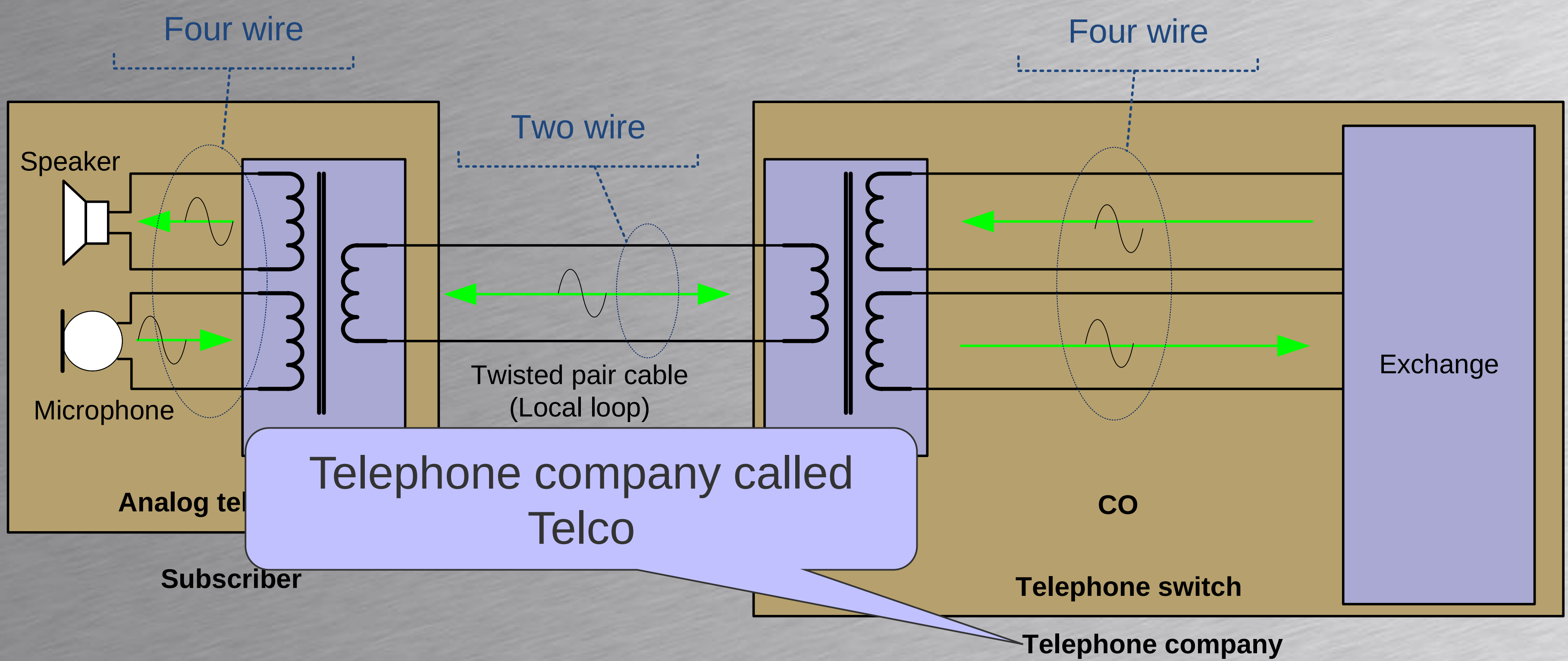
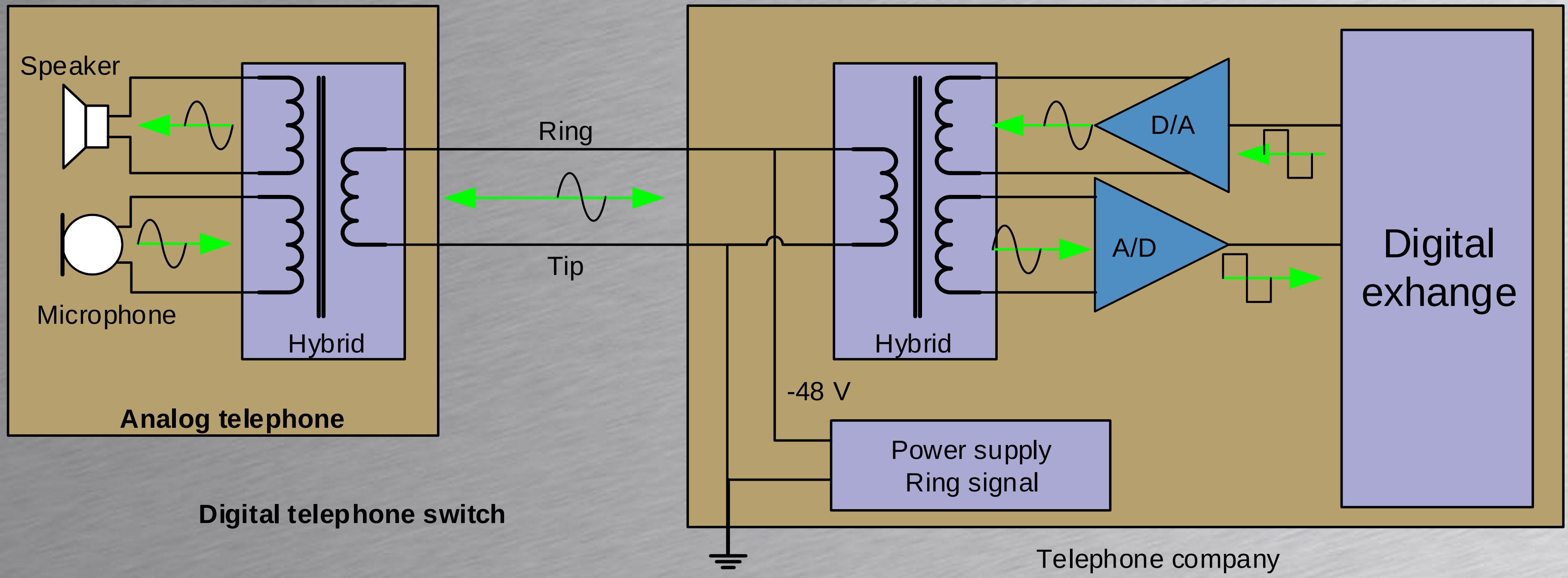
- Two wire twisted pair cable between subscriber and CO (Central Office) or telephone company.
  - Also called local loop or the last mile.
  - Bidirectional communication on two wires



POTS

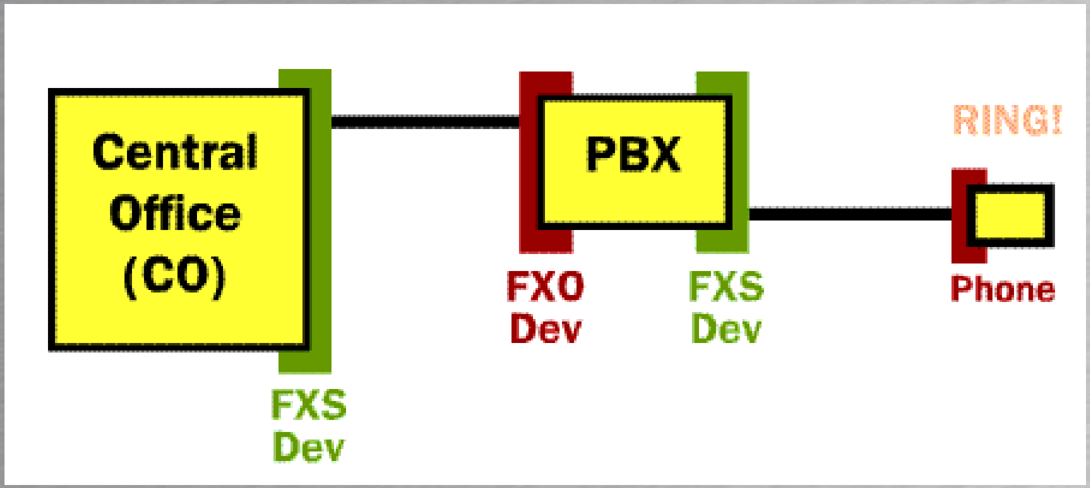
Plain old telephone service

- Power supply to telephone
  - -48 volt DC from exchange to telephone (Tip ground)
  - Often referred to as battery power
- Ring signal from exchange to telephone
  - 90 volt AC at 20 Hz



FXS and FXO

- FXS or foreign exchange service
  - Telephone interface supplying battery power, dial tone and ringing signal
- FXO or foreign exchange office
  - Telephone interface generating off/on-hook

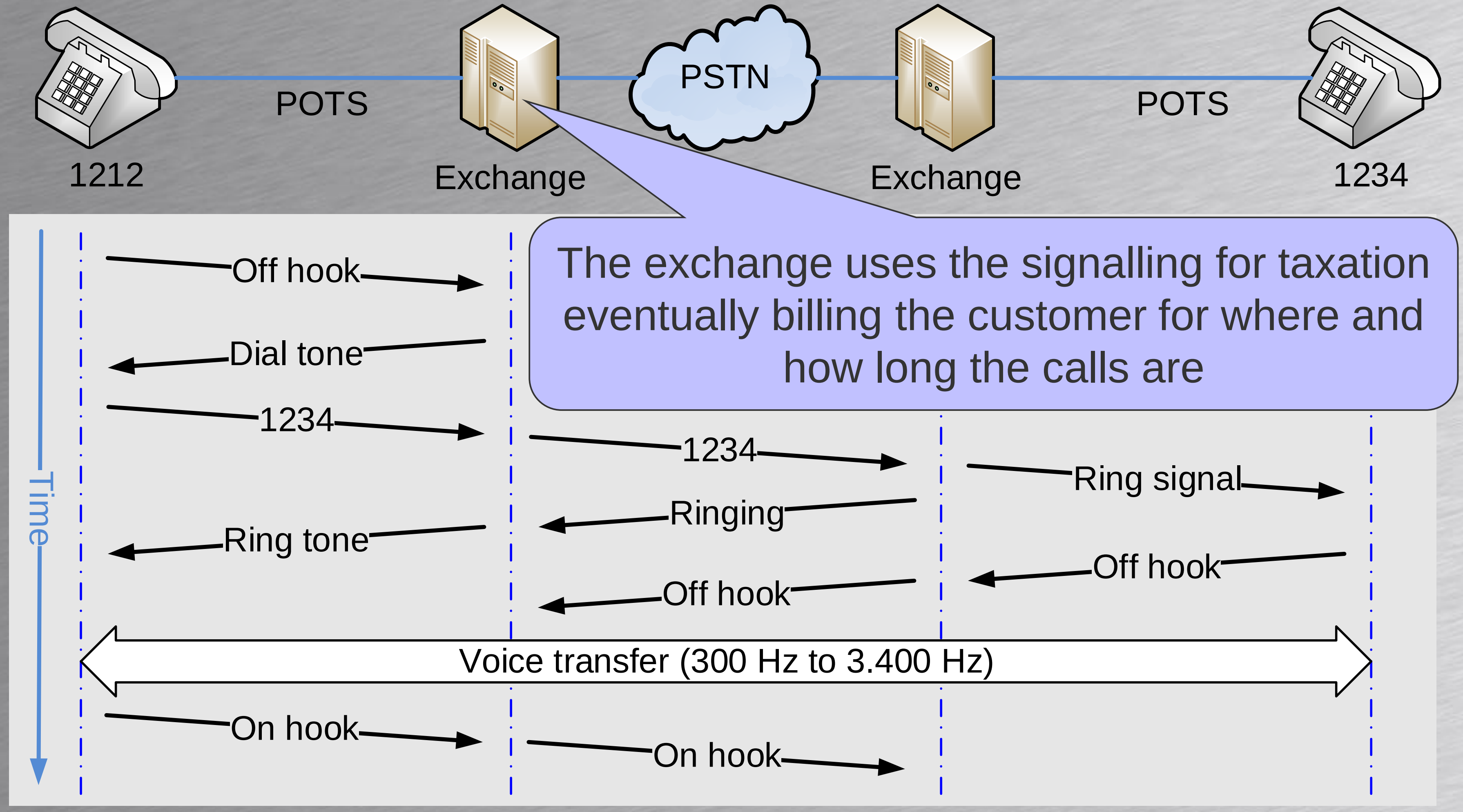


Signaling and voice

- Two kinds of data transmitted on POTS
  - Analog voice between 300 Hz and 3.400 Hz
  - Signaling – When dialing – for example:
    - Off hook signal from phone to exchange
    - Dial tone from exchange to phone
    - Transfer of dialed number from phone to exchange
    - Ring tone from exchange to phone
    - On hook signal from phone to exchange

Signaling and voice

POTS call progress



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POTS signaling

- Transfer of dialed number between phone and exchange
- Two ways of transferring number
  - Pulse (Old way – can still be used on POTS)
  - DTMF

Pulse signaling

DTMF signaling

Pulse dialing

- Different standards
  - Sweden:
    - 1 pulse for 0
    - 2 pulses for 1
    - …
    - 10 pulses for 9
  - World (except Oslo and Australia)
    - 1 pulse for 1
    - 2 pulses for 2
    - …
    - 9 pulses for 9
    - 10 pulses for 0

Pulse dialing

- Different standards
  - Oslo and Australia:
    - 1 pulse for 9
    - 2 pulses for 8
    - …
    - 9 pulses for 1
    - 10 pulses for 0

  10 – digit = number of pulses

Oslo and Australia

- Pulses are generated through the making and breaking of the telephone connection

DTMF Dialing

- Dual Tone Multiple Frequencies
  - Two tones sent simultaneously when button pressed

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

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 signaling
  - Voice and signaling 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

Other tones

- Tones from exchange to telephone reports
  - status of line
  - Equipment
  - status off calls

Examples of events	Low frequency	High frequency
Dial tone (Most of Europe)	425 Hz	None
Dial tone (UK and US)	350 Hz	440 Hz
Busy signal (Most of Europe)	425 Hz	None
Busy signal (UK)	400 Hz	None
Busy signal (US)	480 Hz	620 Hz

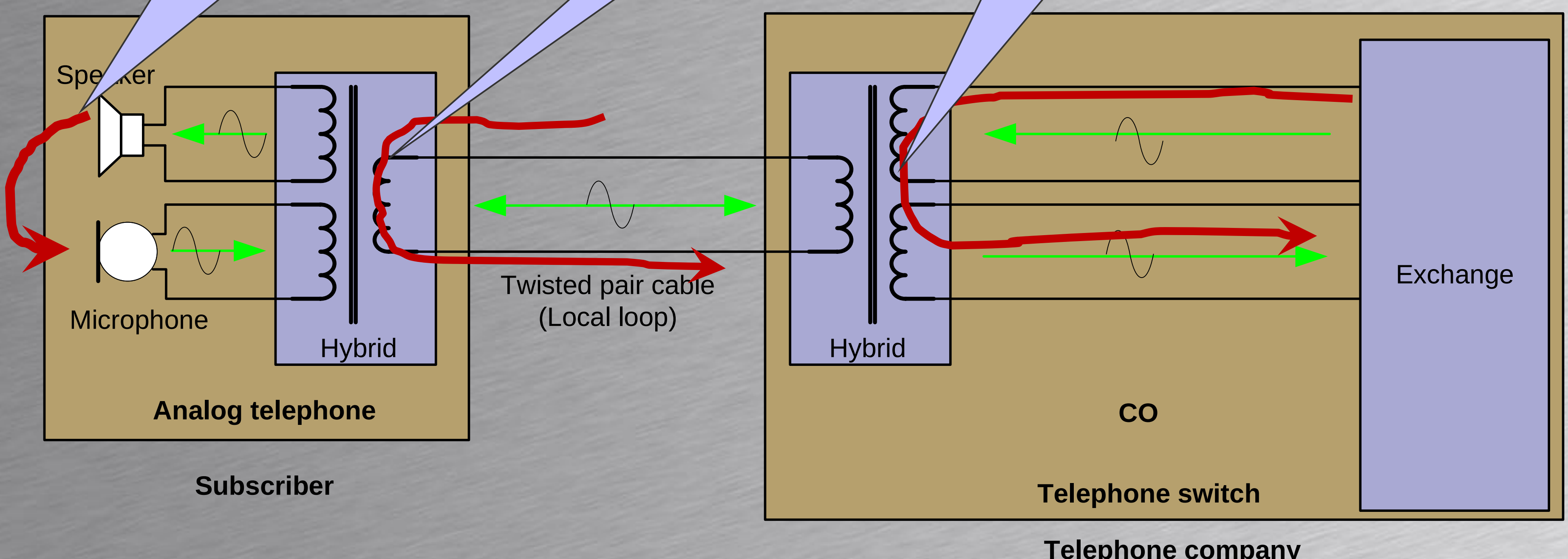
Echo

Mismatch impedance in hybrid echos energi back

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

Mismatch impedance in hybrid echos energi back as echo

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



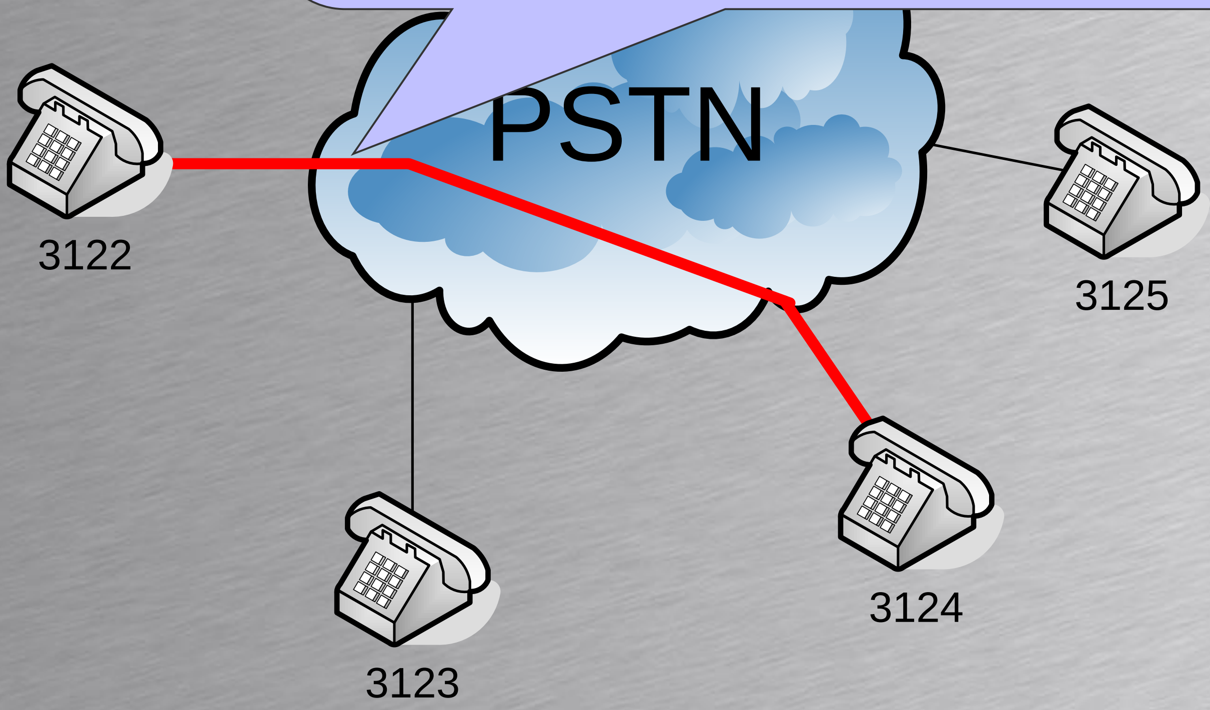
Circuit vs. packet switched

- 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

Circuit switched

PSTN equipment builds connection between phones guaranteeing the necessary bandwidth and giving a constant delay.

Very good and stable voice quality



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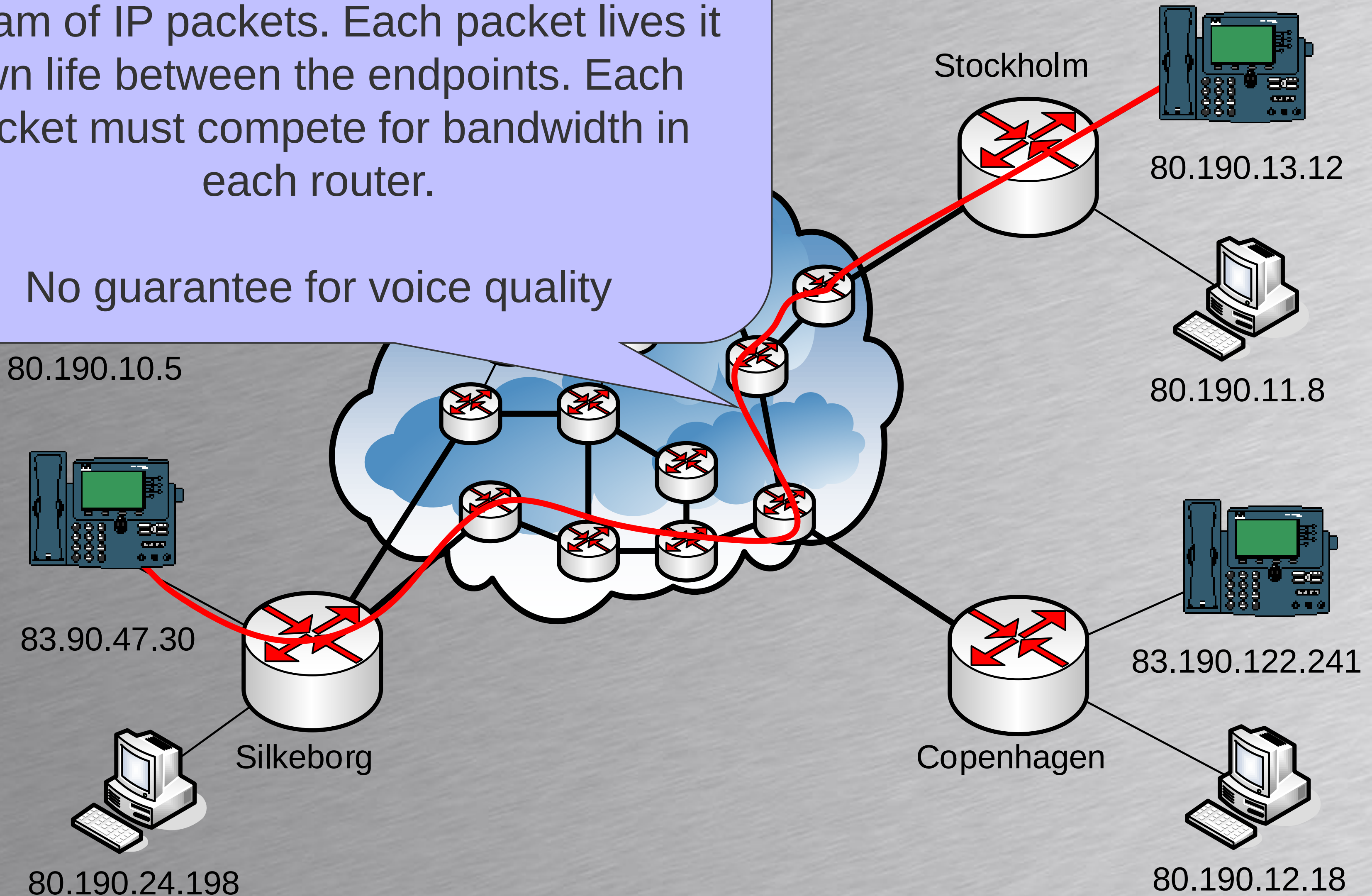
Packet switched

The logical connection consists of a stream of IP packets. Each packet lives it own life between the endpoints. Each packet must compete for bandwidth in each router.

No guarantee for voice quality

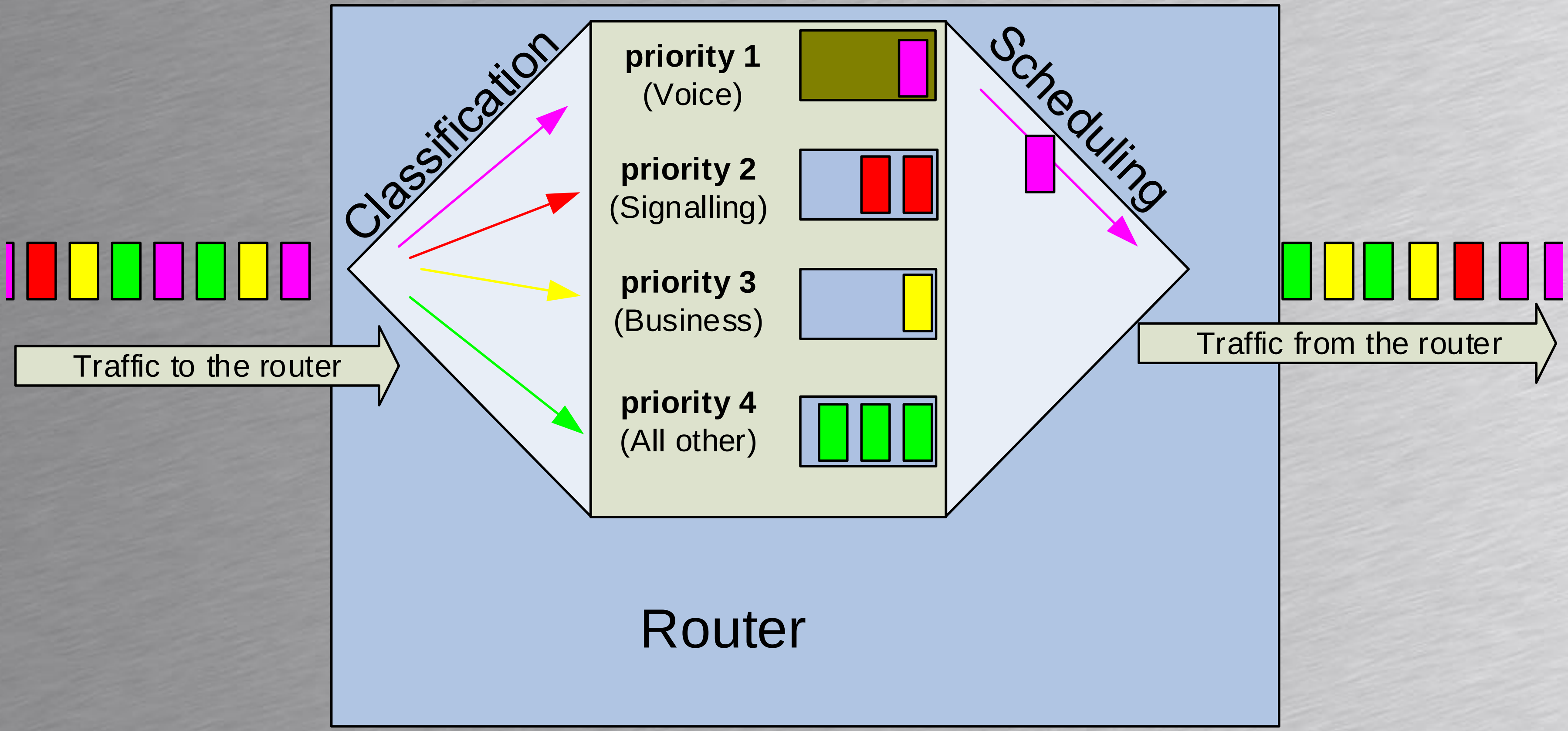


QoS: Quality of Service

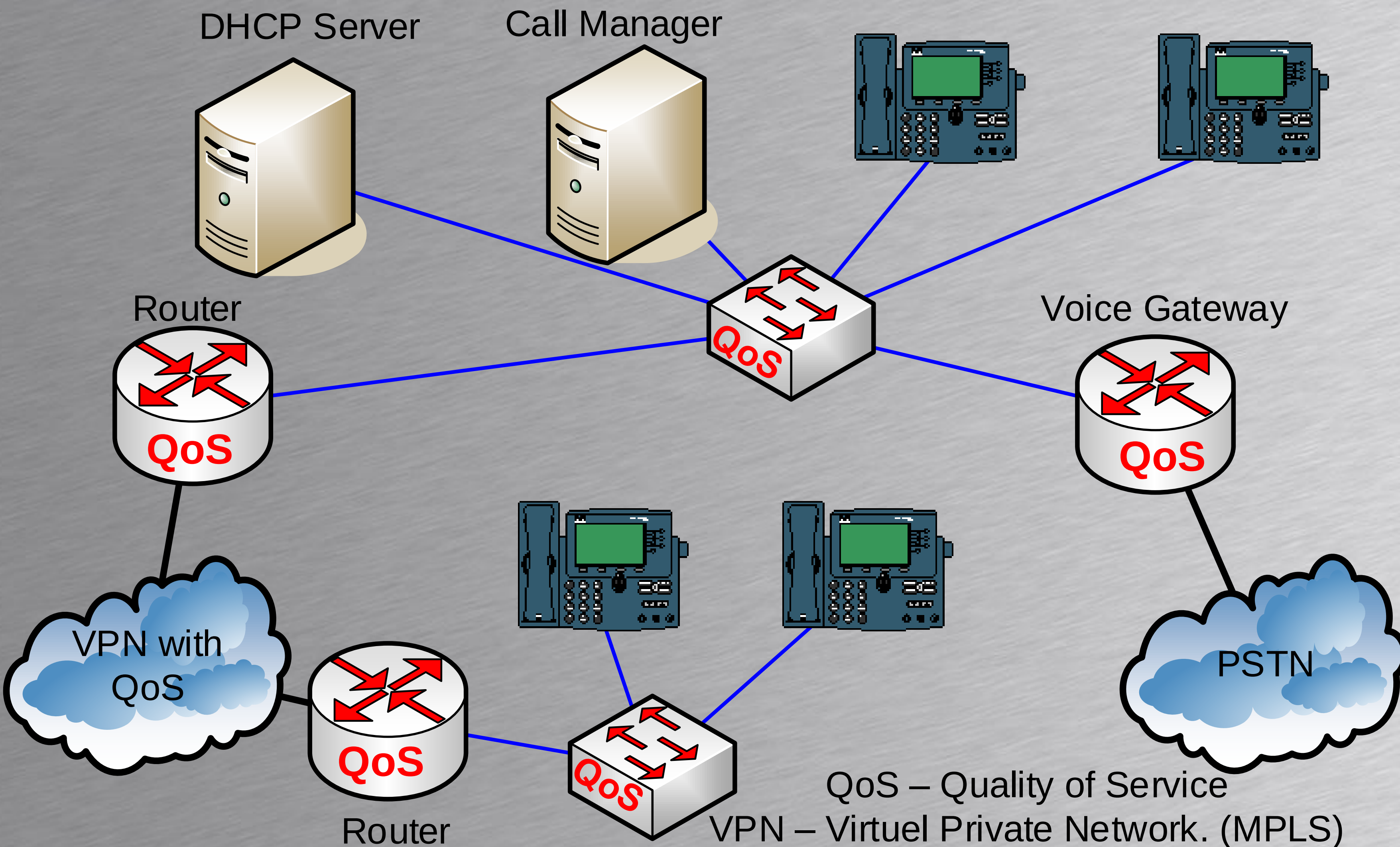
- Used in packet switched networks
- Some packets are prioritized (Voice packets)
- Routers and switches are configured to use QoS and which packets to prioritize
- Widely used by companies using VoIP
- No QoS support on the Internet
  - Variable voice quality (Example Skype)

QoS principle

- The high priority packets overtake the lower prioritized packets in the queues



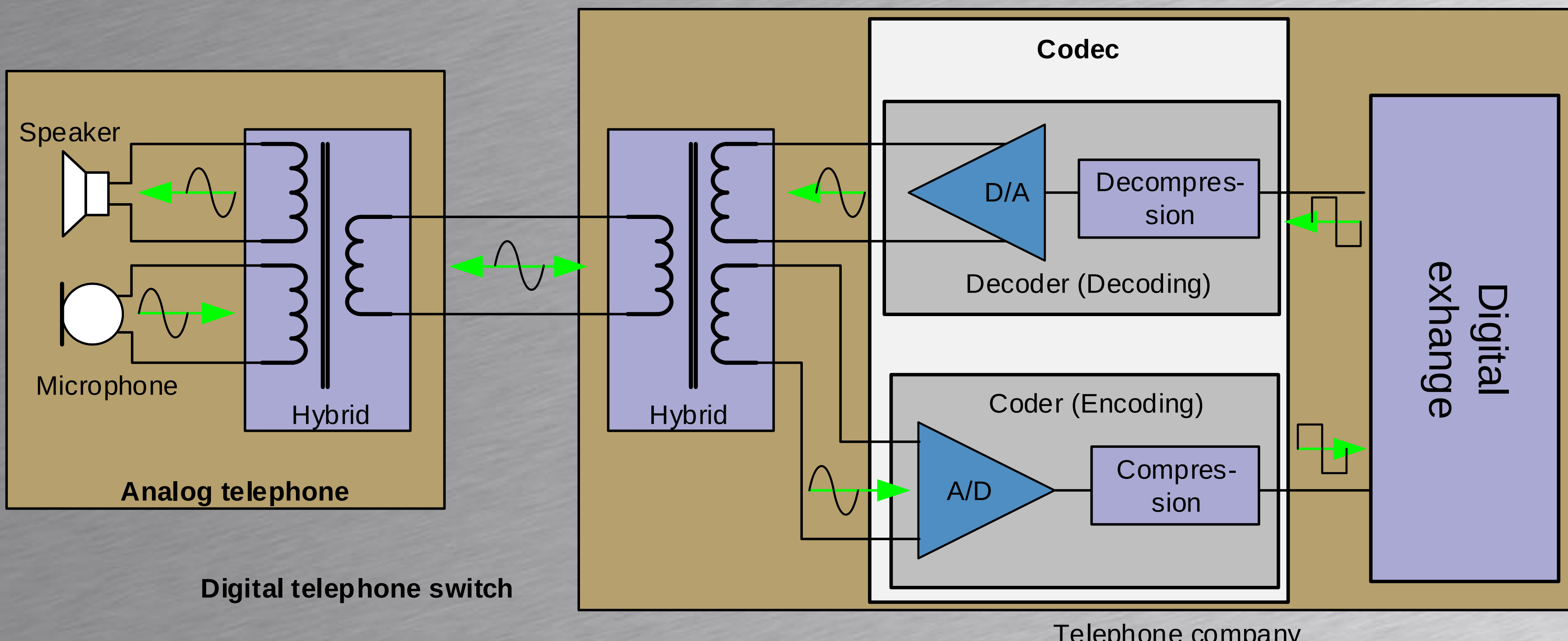
QoS configured



QoS – Quality of Service
VPN – Virtuel Private Network. (MPLS)
PSTN – Public Switched Telephone Network

Codec
coder-**dec**oder

- A coder can encode a signal to a compressed data representation
- Decoding is the reverse of encoding



Codecs

- Different kinds of audio codecs
  - Many codecs defined and used
  - G.standards developed by ITU-T

Codec	Use	Bandwidth	Data rate	Comment
G.711	Telephony	300-3.400 Hz	64 Kbps	Used on PSTN
G.722	Telephony	50-7.000 Hz	48, 56 or 64 Kbps	
G.723	Telephony	300-3.400 Hz	24 or 40 Kbps	Superceded by G.726
G.726	Telephony	300-3.400 Hz	16, 24 or 32 Kbps	32 Kbps used the most
G.729	Telephony	300-3.400 Hz	8 Kbps	License required
Audio CD	Audio	20-20.000 Hz	1,411 Mbps	HiFi stereo (2 channels)
MP3	Audio	20-20.000Hz	128 to 320 Kbps	Many data rates avail.

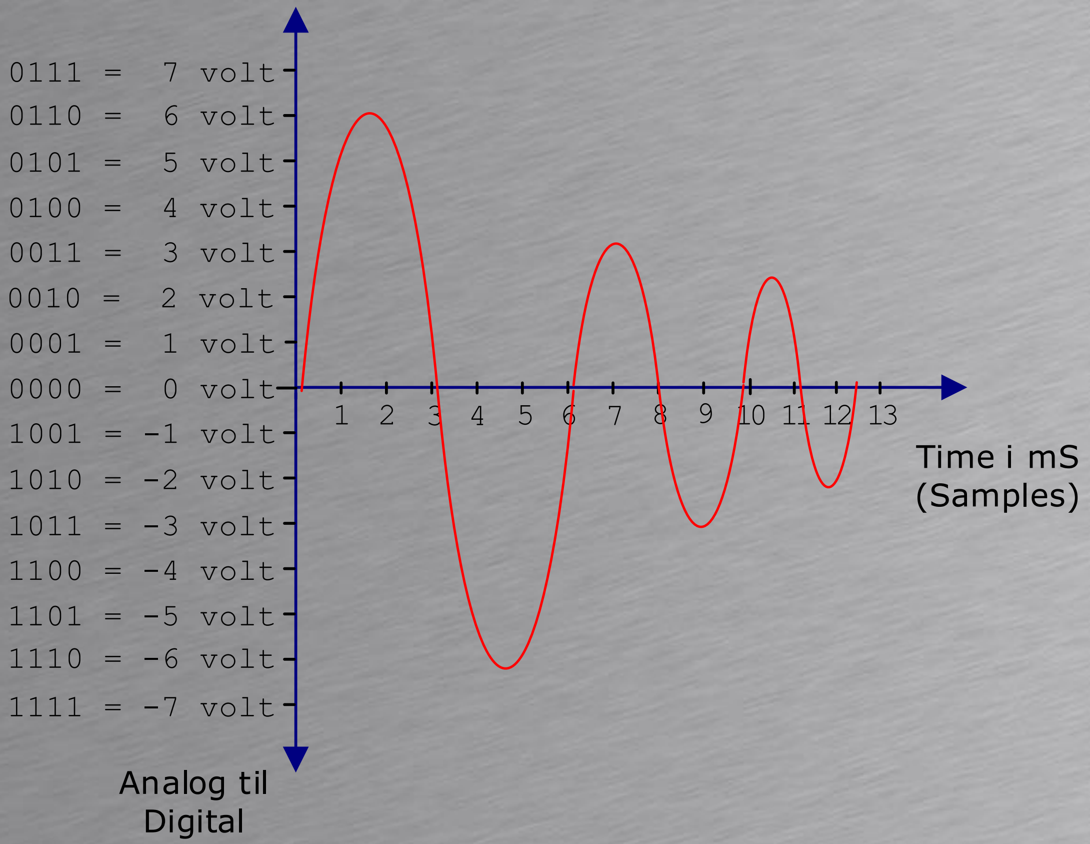
A/D converter

Analog to Digital converter

- Converts analog input to digital output
  - Digital output is a binary number representing the unknown analog input voltage
- Sample frequency
  - How many samples pr. Second the converter converts the unknown analog signal
- Resolution
  - How many bits the converter converts the unknown analog signal to
    - 4 bits sampling = 2^4 = 16 levels
    - 24 bits sampling = 2^{24} = 16.777.216 levels

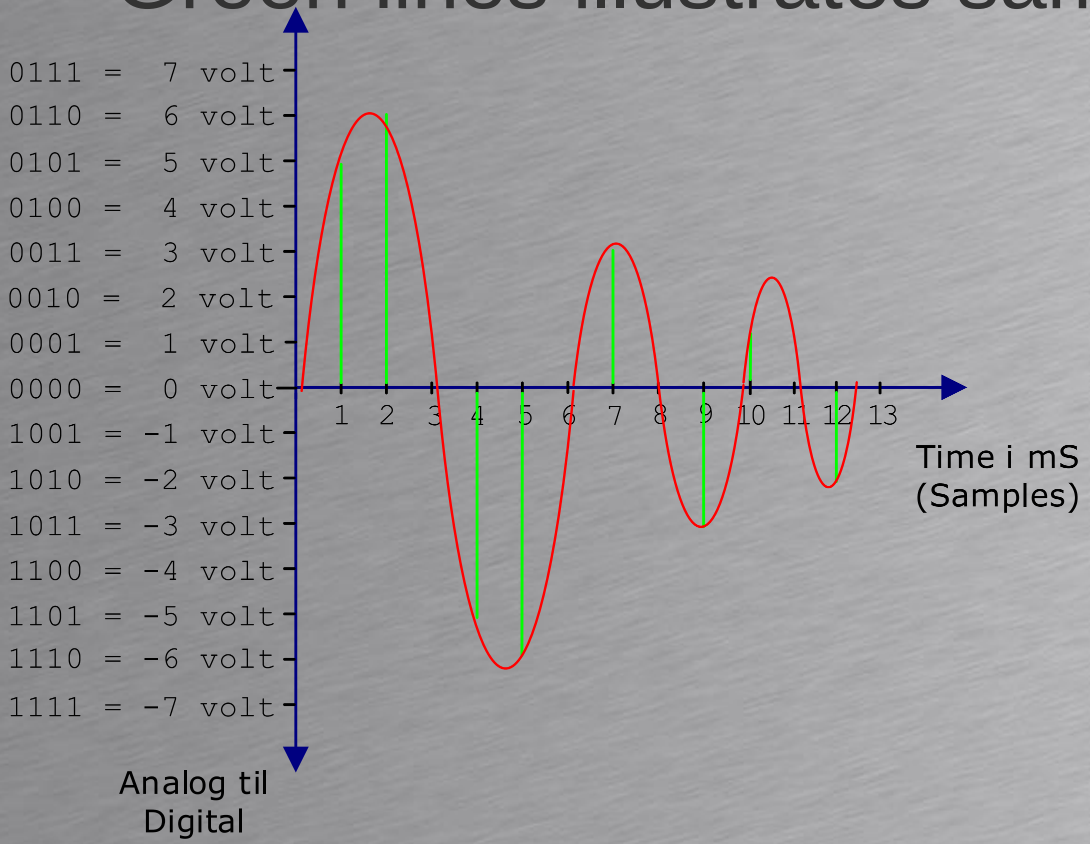
A/D converter

- X-axis is time
- Y-axis is signal amplitude



A/D converter - example

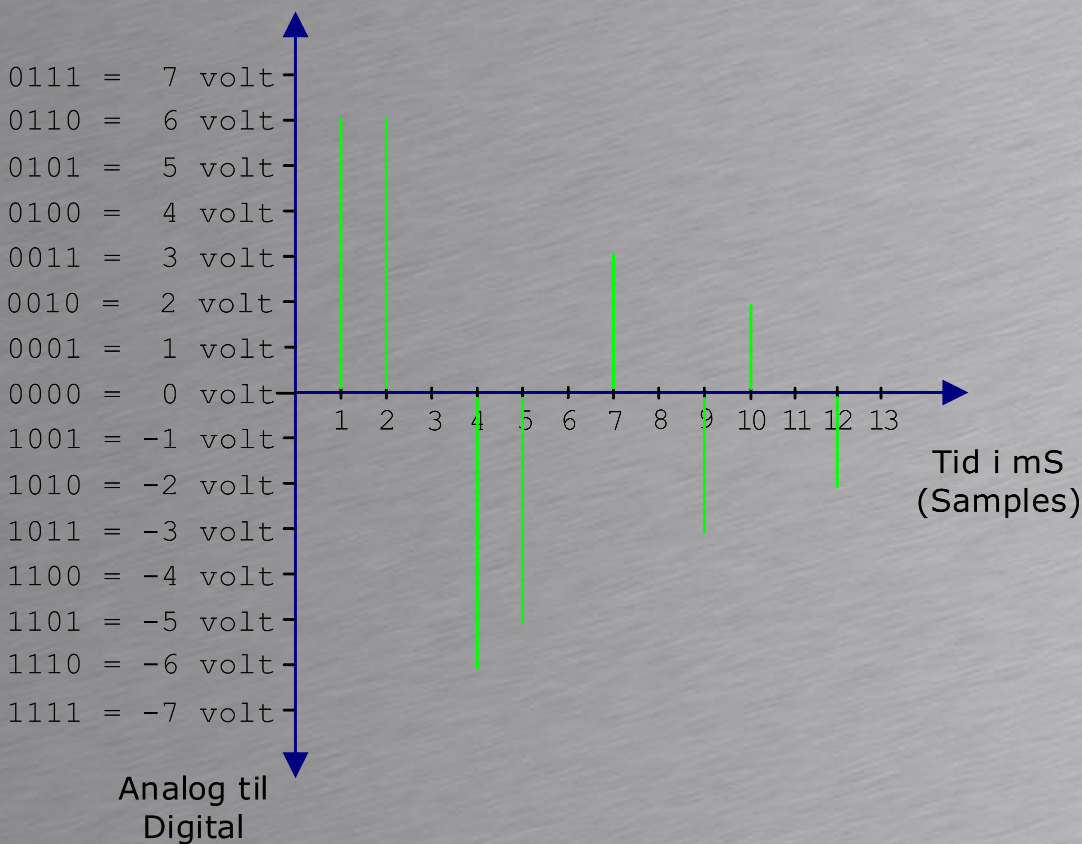
- Sample rate is 1000 times a second (1 Khz)
- Resolution is 4 bit
- Green lines illustrates samples



Sample	Value
1	0101
2	0110
3	0000
4	1101
5	1110
6	0000
7	0011
8	0000
9	1011
10	0010
11	0000
12	1010

D/A converter

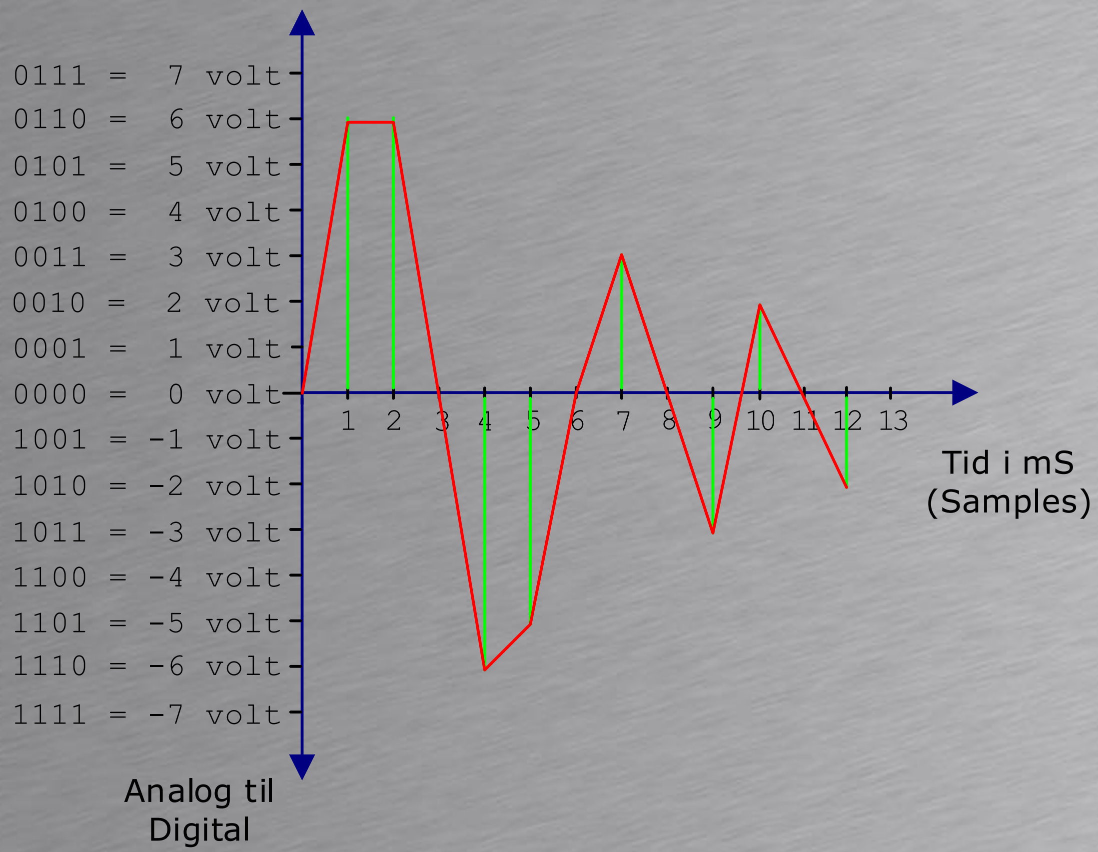
- D/A converter receives digital input
- Outputs analog value (Show in green)



Sample	Value
1	0110
2	0110
3	0000
4	1101
5	1110
6	0000
7	0011
8	0000
9	1011
10	0010
11	0000
12	1010
Sample	Value

D/A converter

- Approximated analog waveform at receiver
- Outputs analog value to speaker



Nyquist theorem

- Harry Nyquist
  - Swedish scientist working for AT&T
- In 1928 Nyquist stated

> To adequately represent an analog wave in digital form , you must sample the analog waveform at a rate at least twice that of the highest frequency to be transmitted

- To transfer voice in the range 300 to 3.400 Hz
  - Sample the signal at least 2 x 3.400 = 6.800 pr. Second
  - To avoid aliasing (distortion) a higher frequency is used

Codecs

- Common sampling rates

Codec	Use	Bandwidth	Data rate	Sample rate pr. second
G.711	Telephony	300-3.400 Hz	64 Kbps	8.000
G.722	Telephony	50-7.000 Hz	48, 56 or 64 Kbps	16.000
G.723	Telephony	300-3.400 Hz	24 or 40 Kbps	8.000
G.726	Telephony	300-3.400 Hz	16, 24 or 32 Kbps	8.000
G.729	Telephony	300-3.400 Hz	8 Kbps	8.000
Audio CD	Audio	20-20.000 Hz	1,411 Mbps	44.100
MP3	Audio	20-20.000Hz	128 to 320 Kbps	44.100

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

G.711 codecs

- G.711 codecs come in two flavours
  - µ-Law and A-law

Codec	Used in	Sampling rate	Data rate
µ-Law	USA and Japan	8 Khz	64 Kbps
A-Law	Europe and rest of the world	8 Khz	64 Kbps

- Data rates for both codes

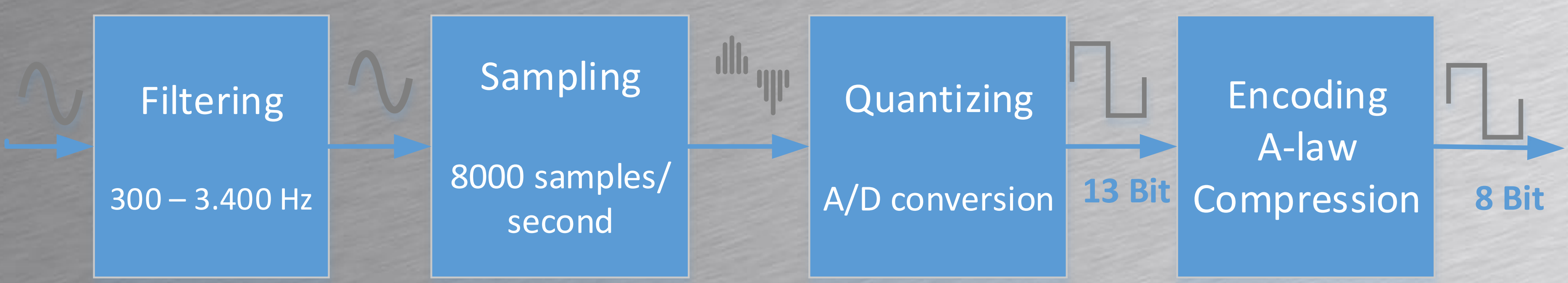
  8000 samples/second x 8 bit/sample = 64 Kbps

- µ-law and A-law are <u>not</u> compatible

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



G.711 codecs

- G.711 codecs come in two flavours
  - μ-Law and A-law

Codec	Used in	Sampling	Resolution	Compression	Data rate
μ-Law	USA and Japan	8 Khz	14 bit	to 8 bit/sample	64 Kbps
A-Law	Rest of the world	8 Khz	13 bit	to 8 bit/sample	64 Kbps

- Data rates for both codes

  8000 samples/second x 8 bit/sample = 64 Kbps

- μ-law and A-law are <u>not</u> compatible

ISDN

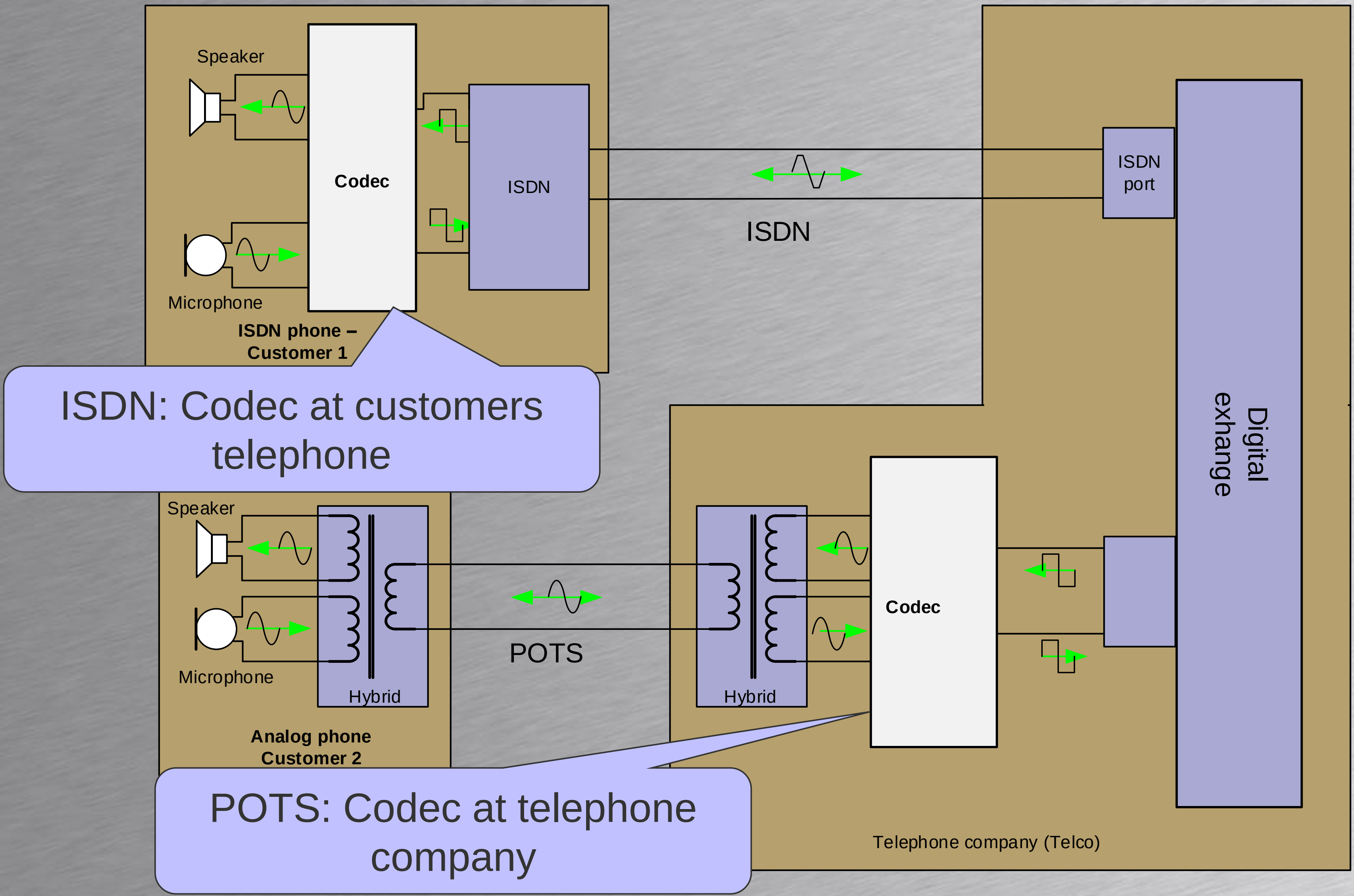
Integrated Services Digital Network

ISDN

Integrated Services Digital Network

- Defined in 1988
- Communication standards for
  - Carrying voice, video and data over PSTN
  - Digital transmission from end-points (Customer)
  - 64 Kbps channels for voice, data or video
    - called B-channels
  - 16 or 64 Kbps signaling channel or data
    - Called D-channel
  - Both B and D channels are full duplex
    - Simultaneous transmission in both directions

ISDN phone principle



ISDN

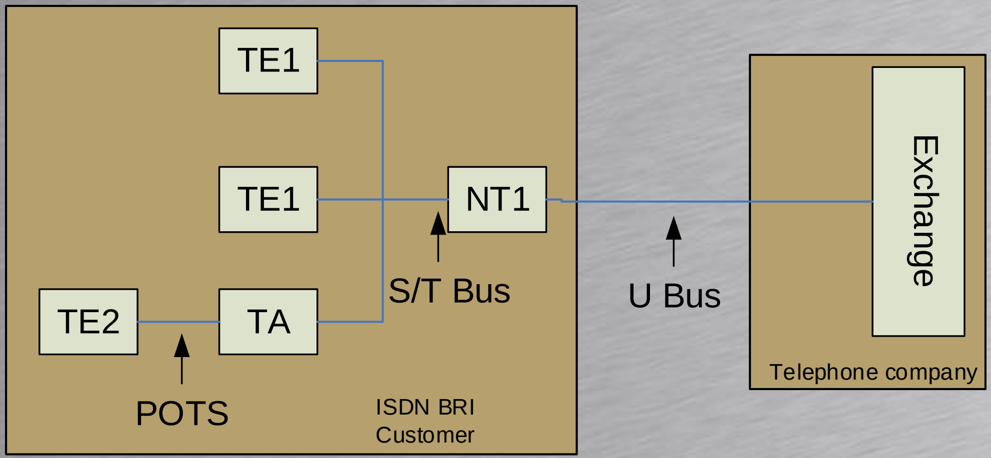
Integrated Services Digital Network

- Two rates available from providers
- ISDN BRI (Basic Rate Interface)
  - Two 64 Kbps B-channels for voice/data
  - One 16 Kbps D-channel for signaling
  - ISDN BRI
- ISDN PRI (Primary Rate Interface)
  - 23 x 64 Kbps B-channels (USA and japan)
  - 30 x 64 Kbps B-channels (Rest of the world)
  - One 64 Kbps D-channel for signaling

ISDN BRI

Basic Rate Interface

- U bus: Two wire local loop connection
  - Same cable used for POTS
- S/T bus: 2 pairs of wire including power
- NT1 – Network Terminator 1 – small box
- TE1 – Terminal Equipment (ISDN phone)
- TA – Terminal adapter. Converts ISDN to POTS
  - Also called AB ports .Analog to B-channel

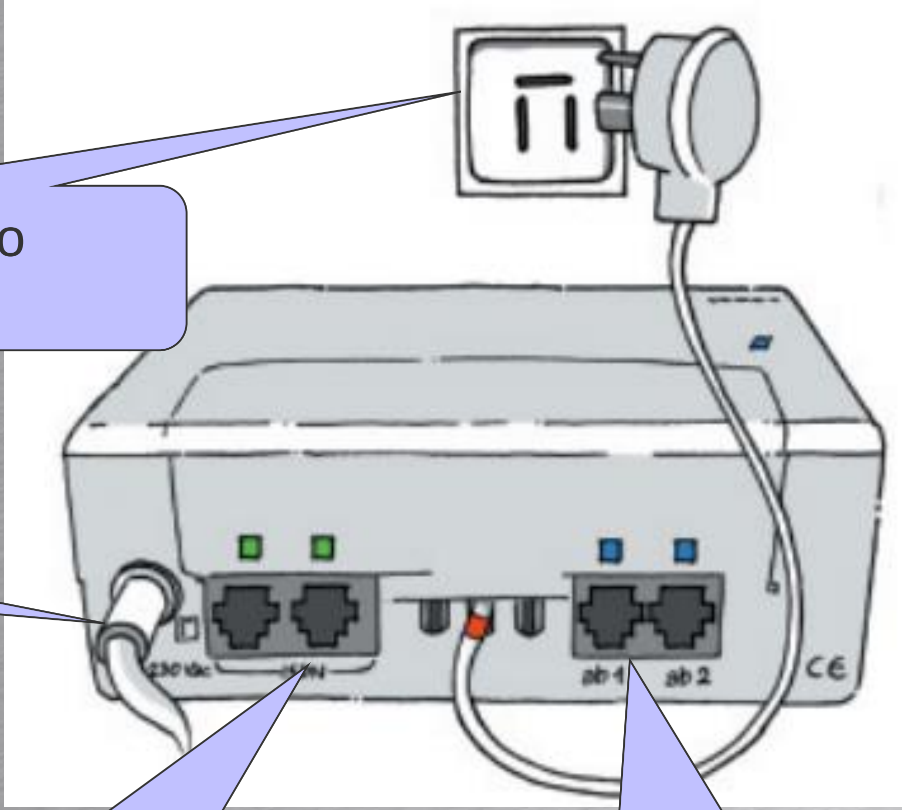


ISDN BRI NT1 box

- NT1 box

Connection to telco
One pair of wire

Mains power

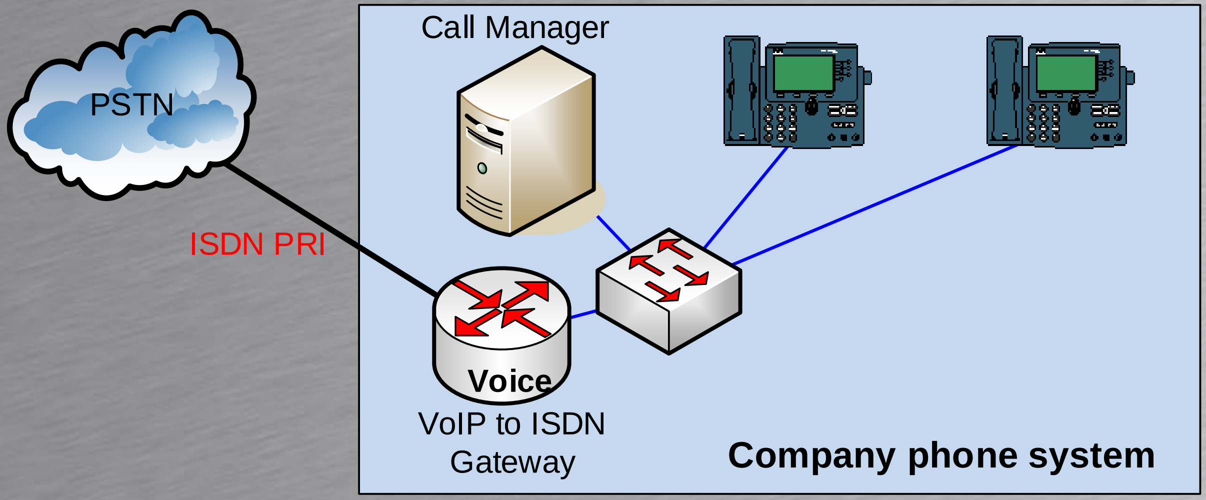


Connection to ISDN equipment

Connection to analog (POTS) equipment

ISDN PRI

- Mainly used to connect Private telephone exchanges or IP phone system to PSTN
  - Called PBX (Private Branch eXchange)
- For example dial 0 to get a public line
- Customer calls are routed through ISDN PRI



ISDN PRI

- Different Telco's ISDN products

ISDN standard	Sweden (Telia)	Norway (Telenor)	Denmark (TDC)
ISDN PRI	ISDN Multi	ISDN Proof/FlexiUT	ISDN Flex

Product	Active B channels
Telia ISDN Multi	30
Telenor Proof	2 – 10
Telenor FlexiUT	12 – 30
TDC Flex	8 – 30

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PBX with ISDN PRI

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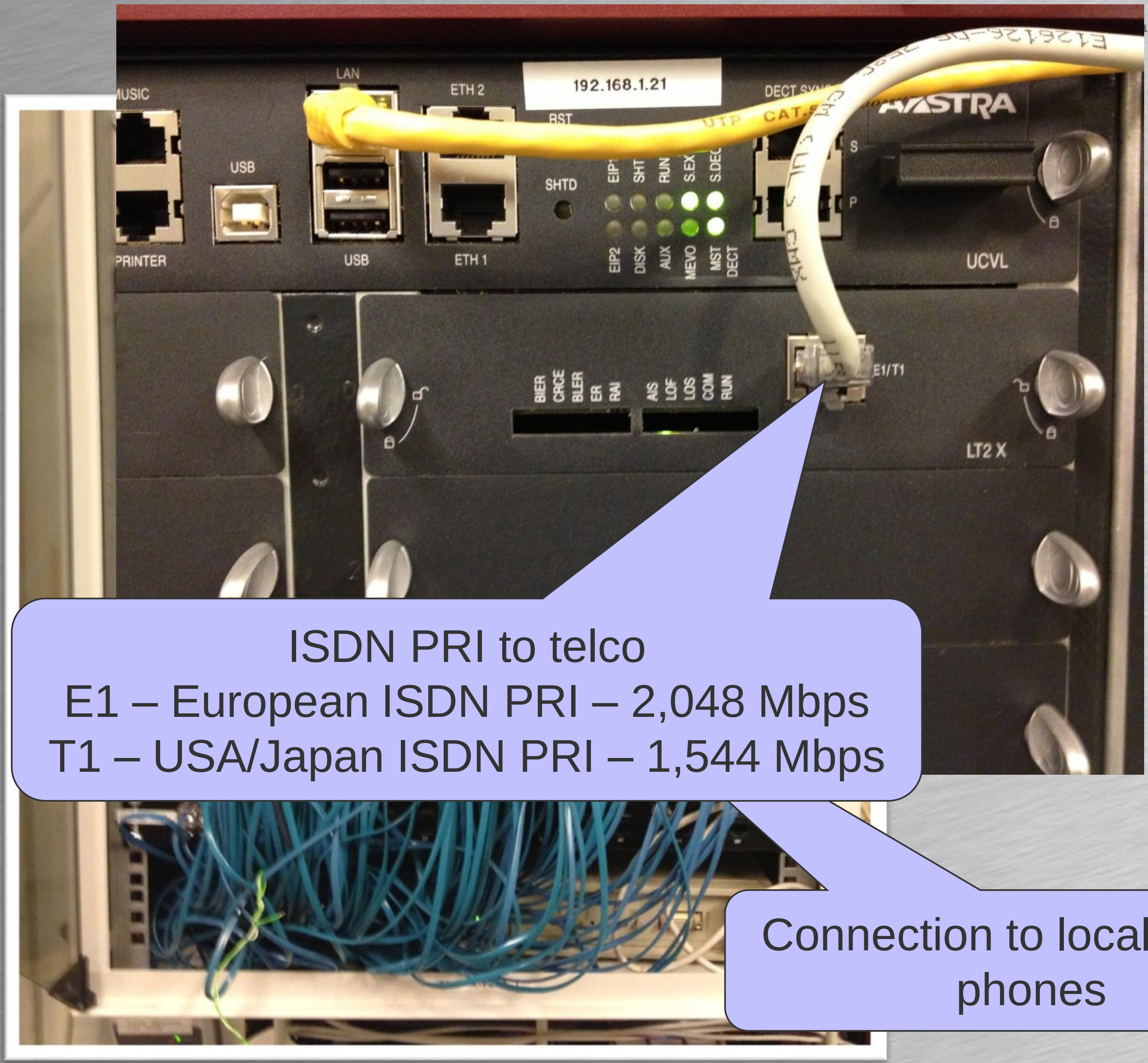


ISDN PRI to telco
E1 – European ISDN PRI – 2,048 Mbps
T1 – USA/Japan ISDN PRI – 1,544 Mbps

Connection to local analog phones

Q.931

- ISDN signalling protocol
- Used to establish, maintain and release connections between end-points. (Phones)
- Signalling carried in ISDN D channel
  - 16 Kbps ISDN BRI
  - 64 Kbps ISDN PRI

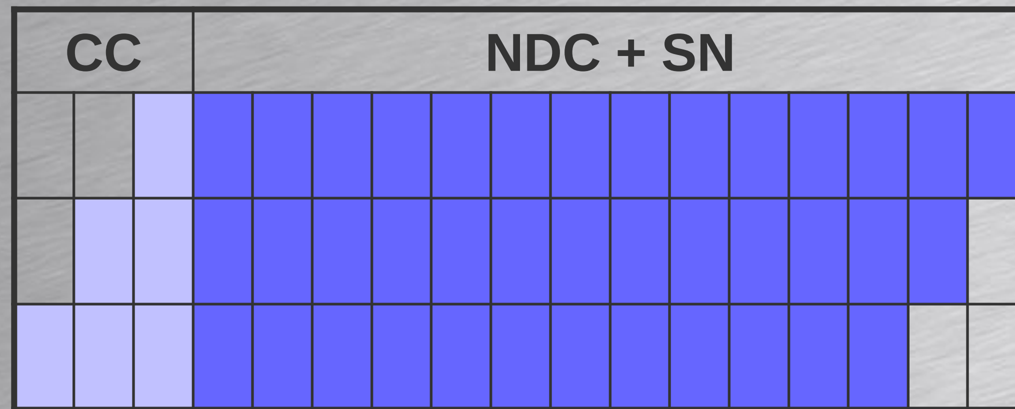
Q.931 signalling

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

ITU-T number plan E.164

- CC = Country Code
  - 1 = USA
  - 45 = Denmark
  - 46 = Sweden
  - 47 = norway
  - 299 = Greenland



Maximum 15 digits

- NDC = National Destination Code
  - Can be used to ”subnet” regions in a country (Area code)
- SN = Subscriber number
- + 45 3053 9361
  - 45 = Country code
  - 3053 9361 = NDC + SN
- www.numberingplans.com
- http://countrycode.org/ http://www.countrycallingcodes.com

ITU E.164

- International dialing

  +45 30539361 means

  \+ = International call

  45 = Country Code (Denmark)

  30539361 = national number = NDC+SN
- International calls +
  - From Europe dial 00 + CC + NDC + SN
  - From USA dial 011 + CC + NDC + SN
  - From Japan dial 010 + CC + NDC + SN
  - Exceptions

ITU-T number plan E.164

Examples

Country	CC	NDC + SN
USA	1	

Maximum 15 digits

ITU-T number plan E.164

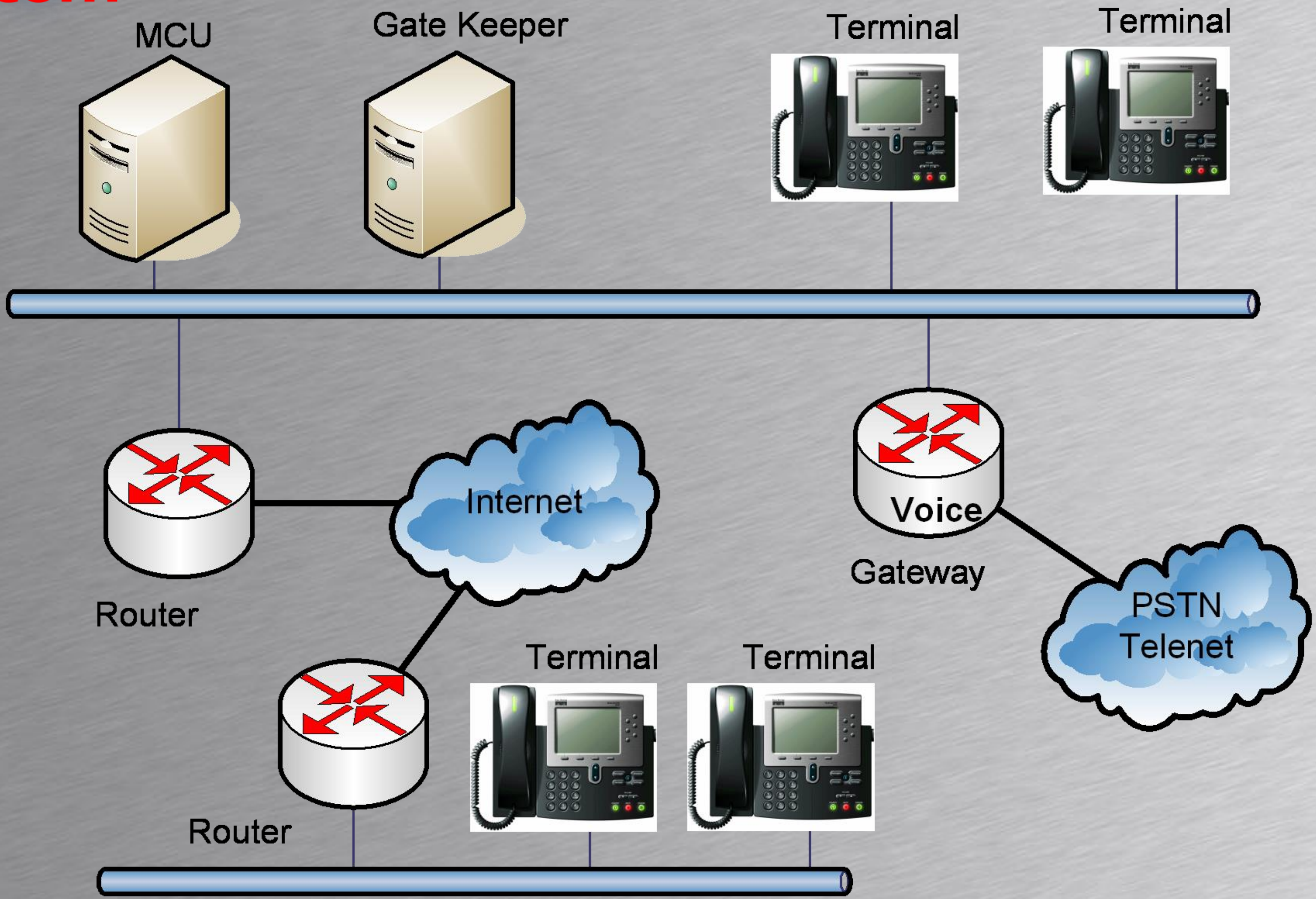
Examples

Country	Number	Country code	NDC (Area Code)	Subscriber number
USA (Texas)	+1-214-555-8283	1	214	5558283
Japan (Tokyo)	+81-3-5618-9876	81	3	56189876
Sweden (STH)	+46-08-685-9000	46	08	6859000
Norway	+47-2324-1245	47		23241245
Denmark	+45-3053-9361	45		30539361
Greenland	+299-981173	299		981173

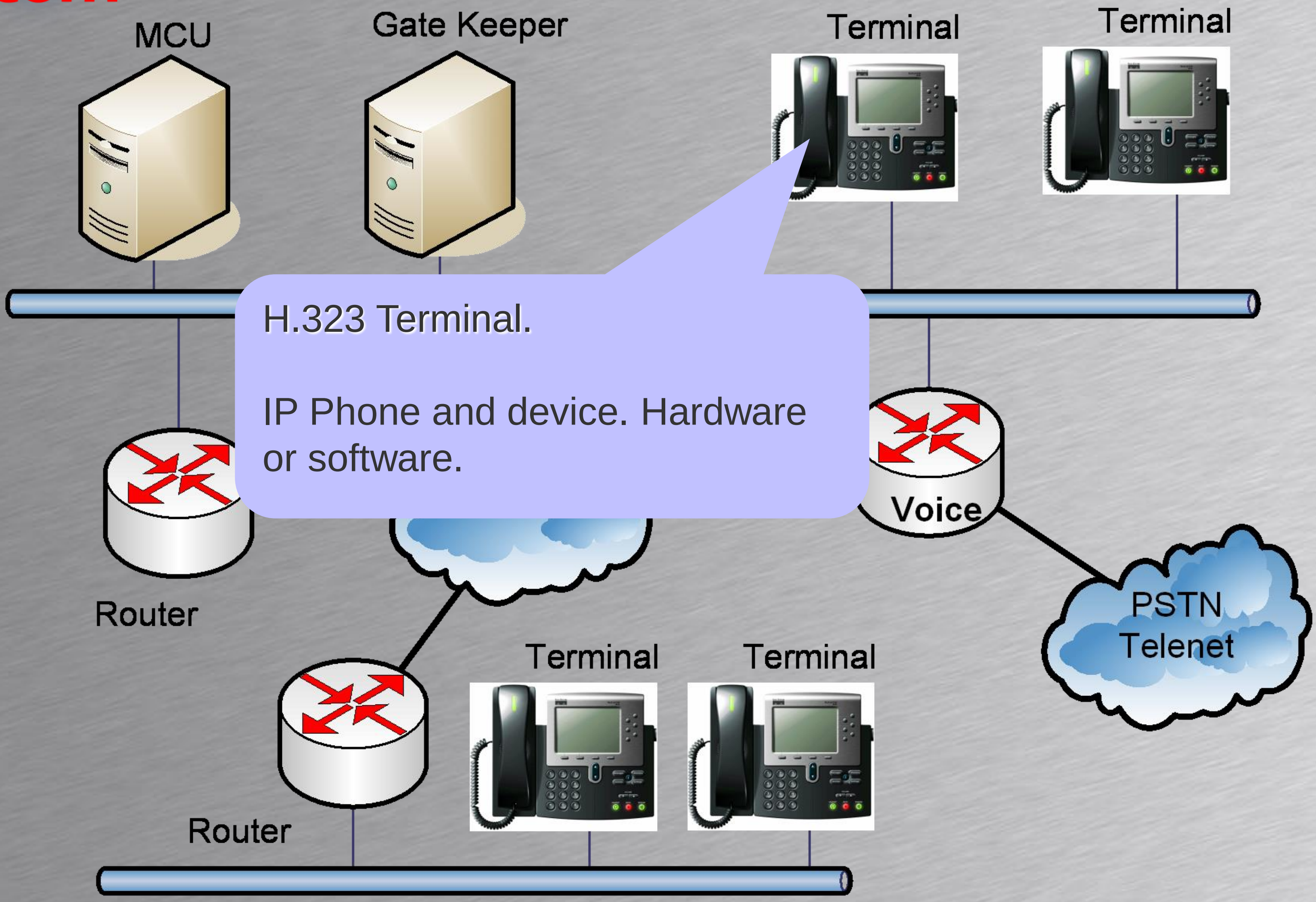
Task

- Groups of 3
  - Connect 2 phones(7940) to the network
  - Change the number from auto to a fixed number on the CallManager.
    - 10 Group 1
    - 11 Group 2
    - 20 Group 3
    - Etc…
  - Create a personal user for the phone.

H.323


MCU
Gate Keeper
Terminal
Terminal
Router
Internet
Voice
Gateway
PSTN
Telenet
Terminal
Terminal
Router

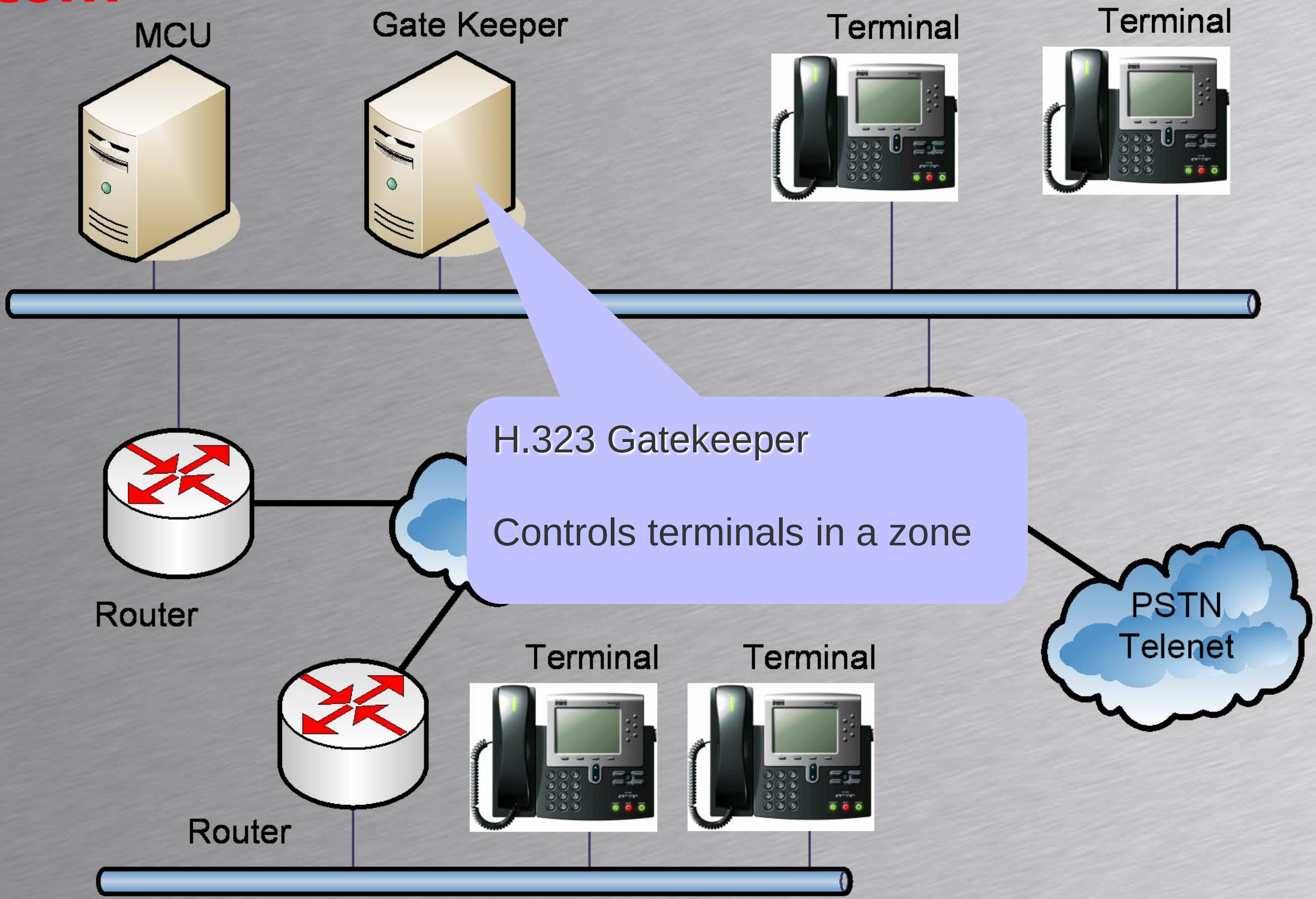
H.323:Terminal



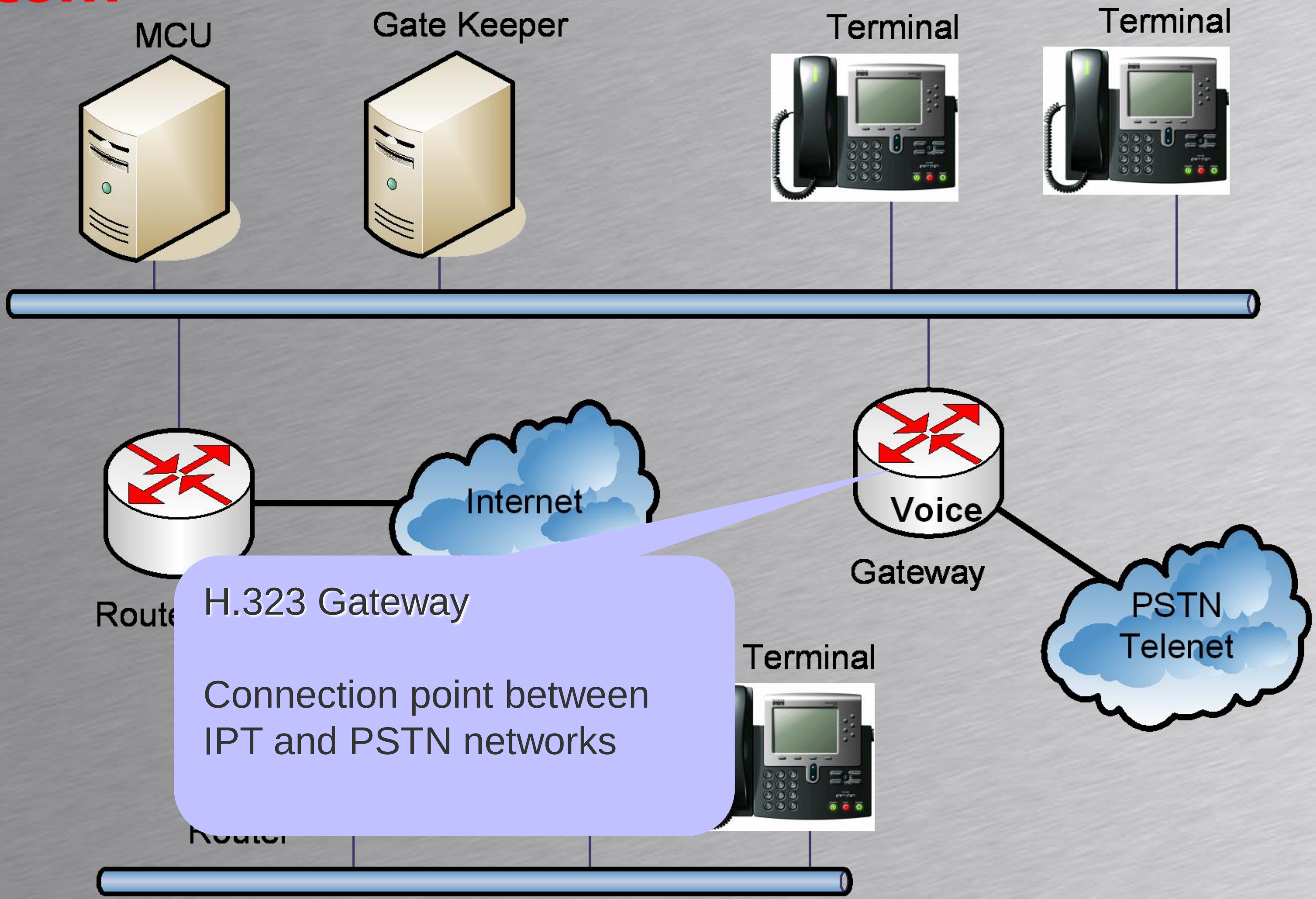
H.323 Terminal.

IP Phone and device. Hardware or software.

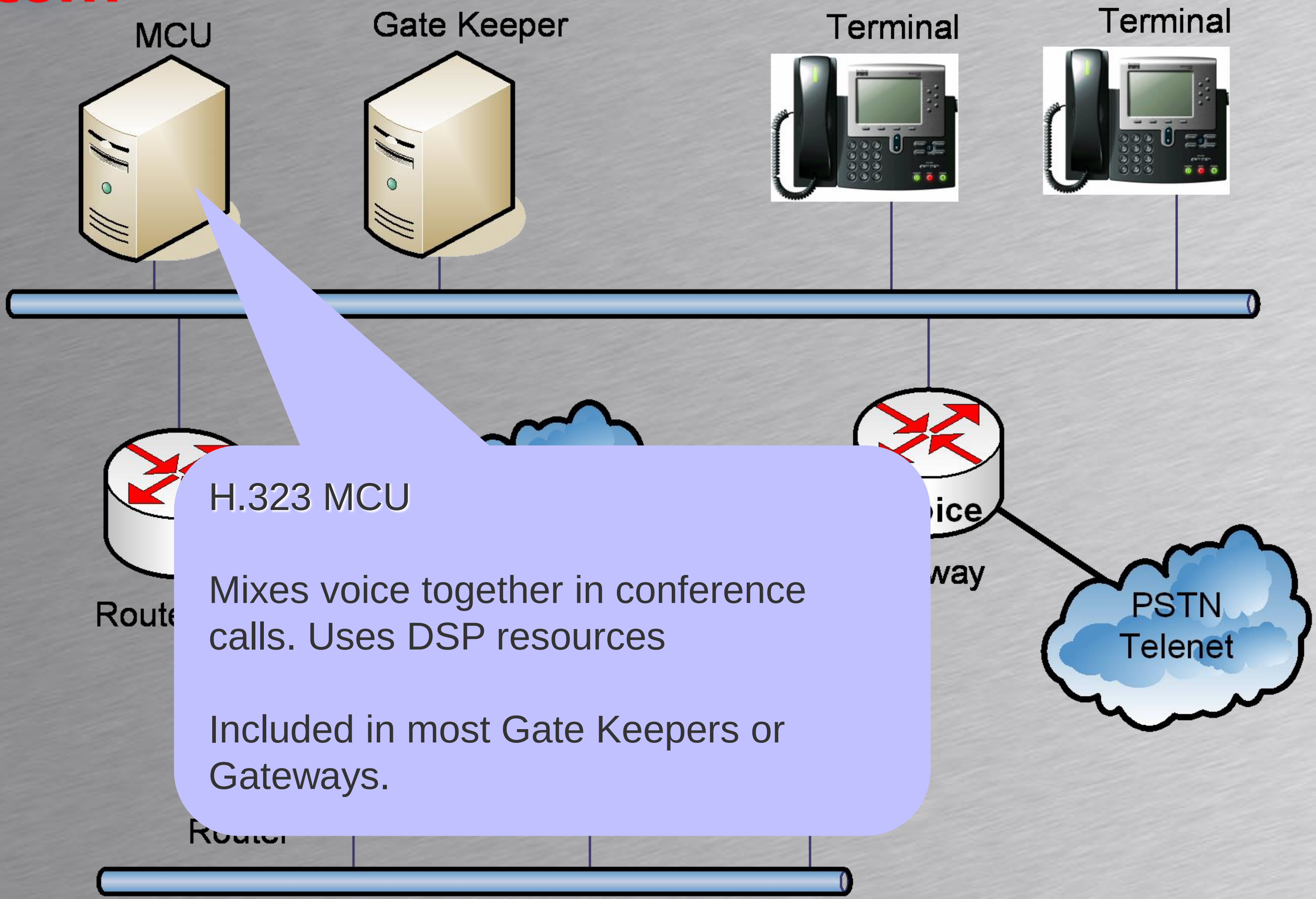
H.323: Gatekeeper



H.323: Gateway



H.323: Multi-point Control Unit

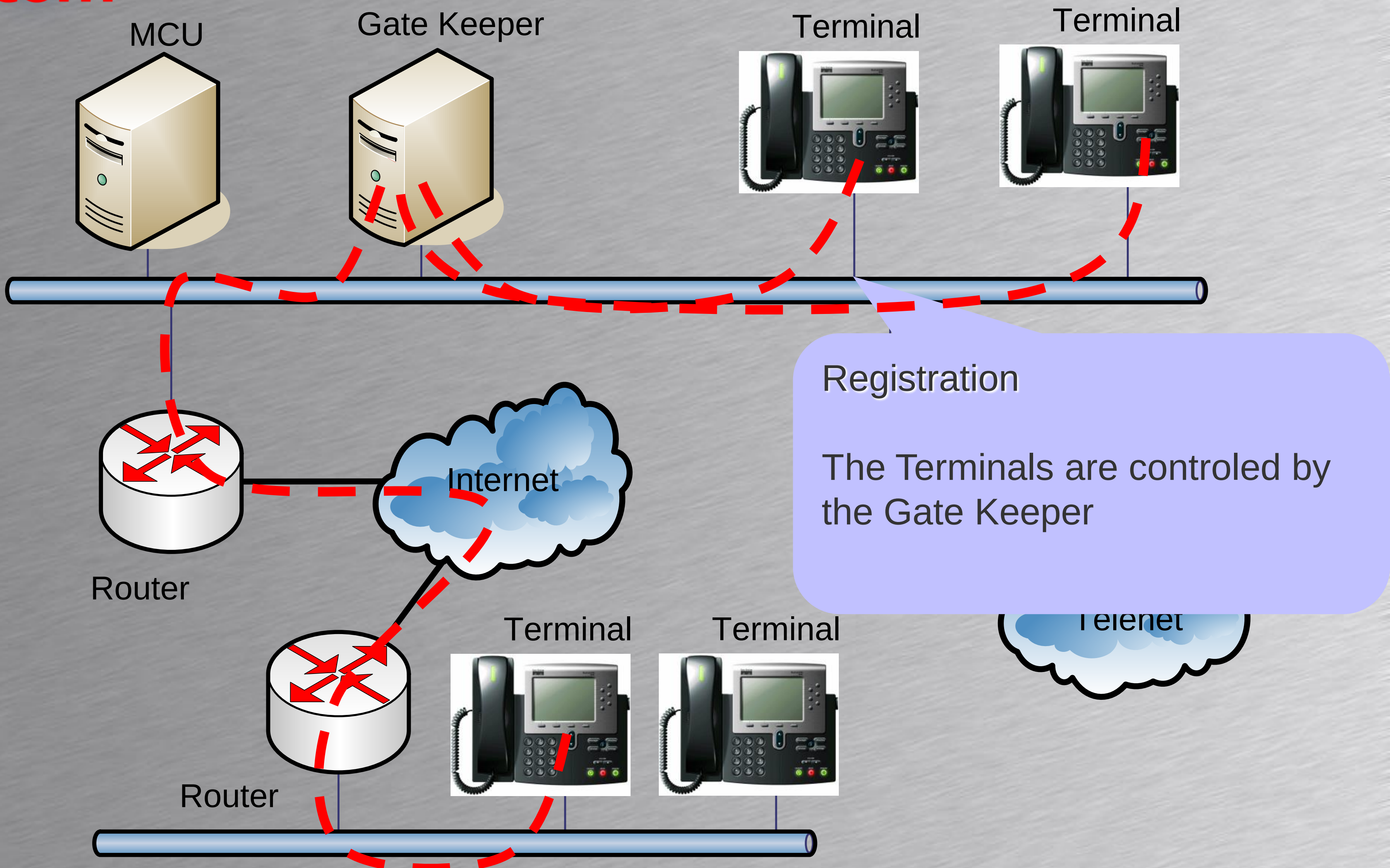


H.323 MCU

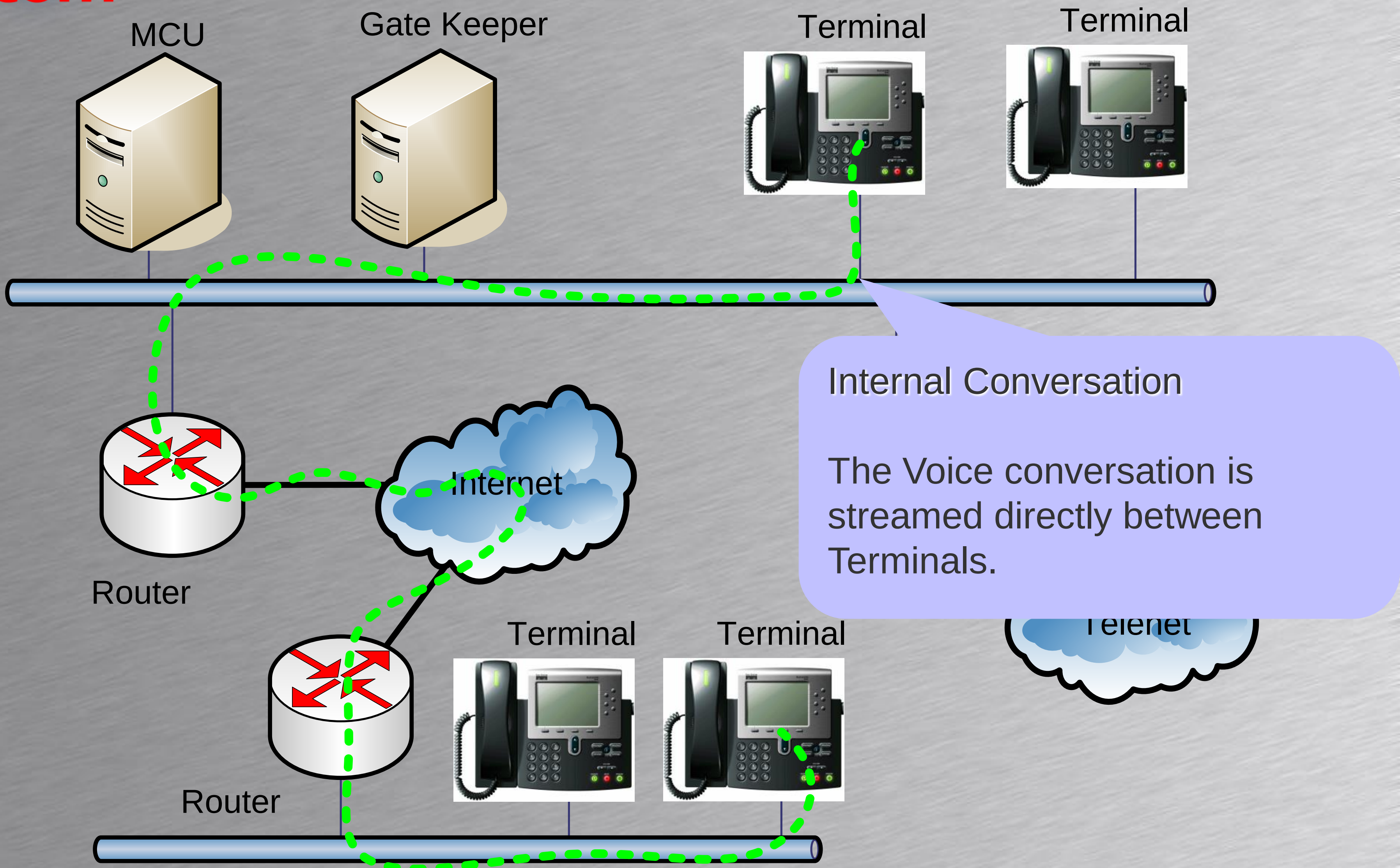
Mixes voice together in conference calls. Uses DSP resources

Included in most Gate Keepers or Gateways.

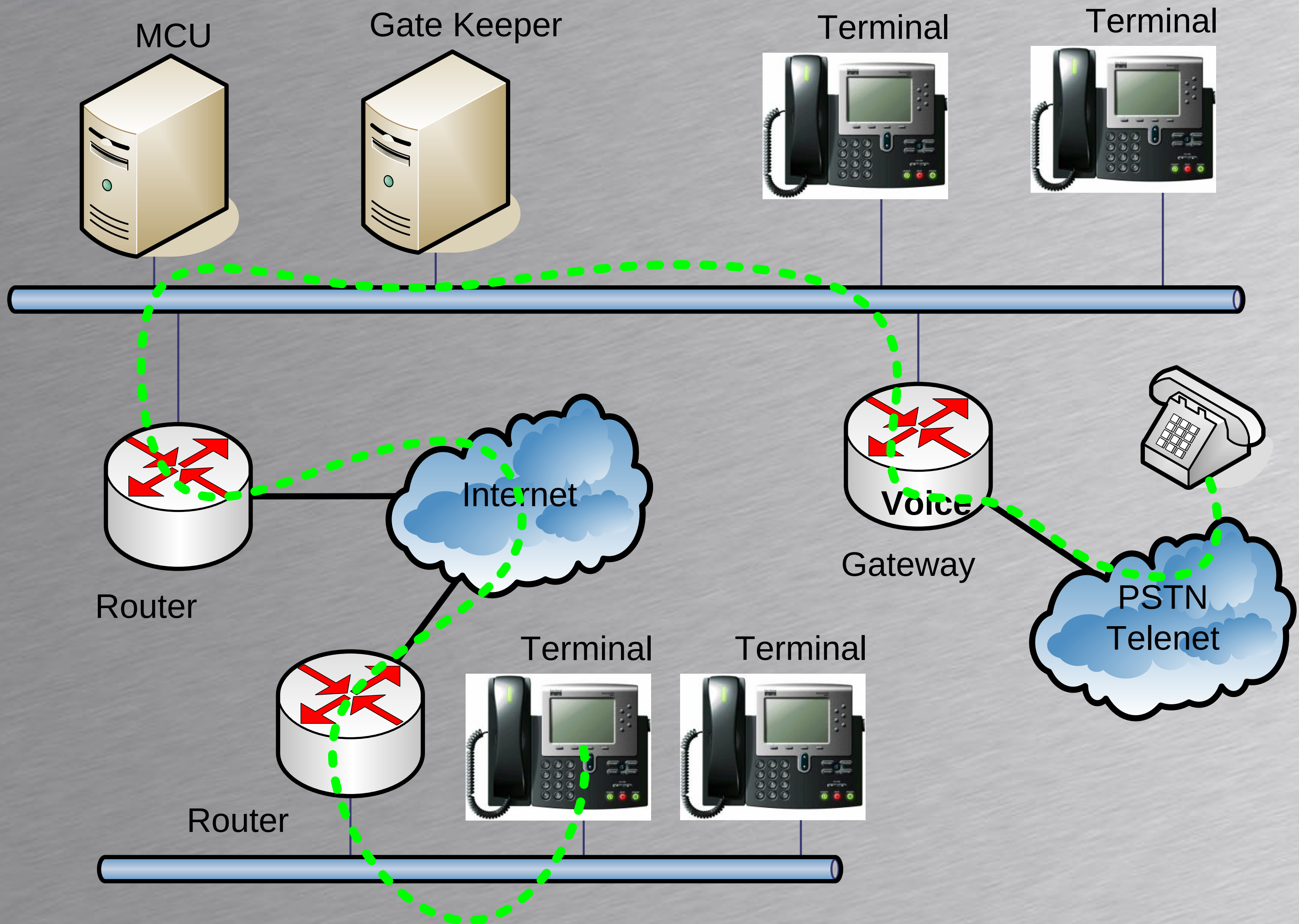
Gatekeeper controls the terminals



Gatekeeper controls the terminals



Terminal to Gateway



WIRESHARK

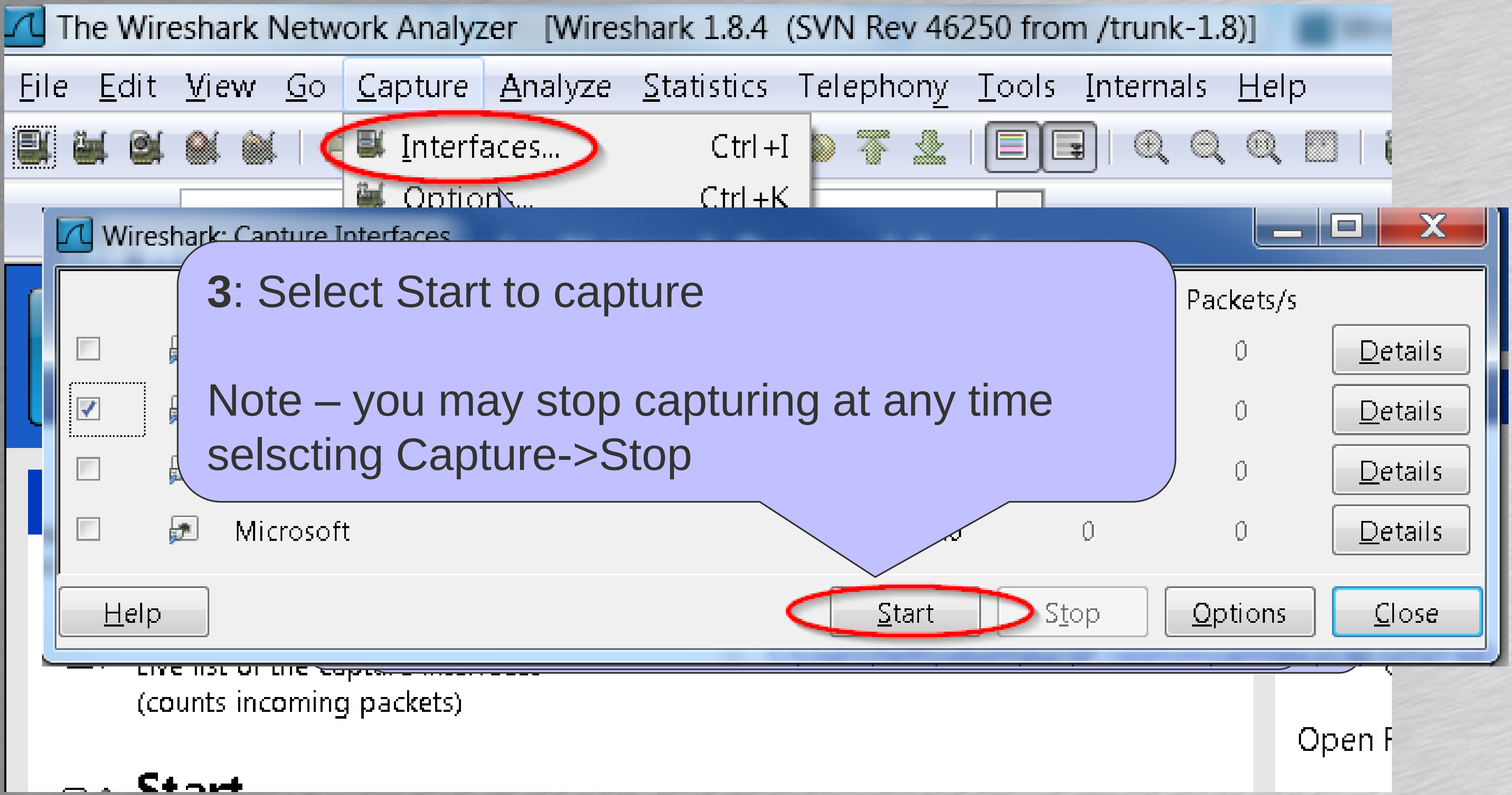
WIRESHARK

Packet analyzer

Wireshark

- Wireshark is free and open source network packet analyzer
- Captures packets from wired or wireless networks for analyze
- Useful for troubleshooting
- Useful for understanding protocols
  - Primary use in this course module
- Download from http://www.wireshark.org/

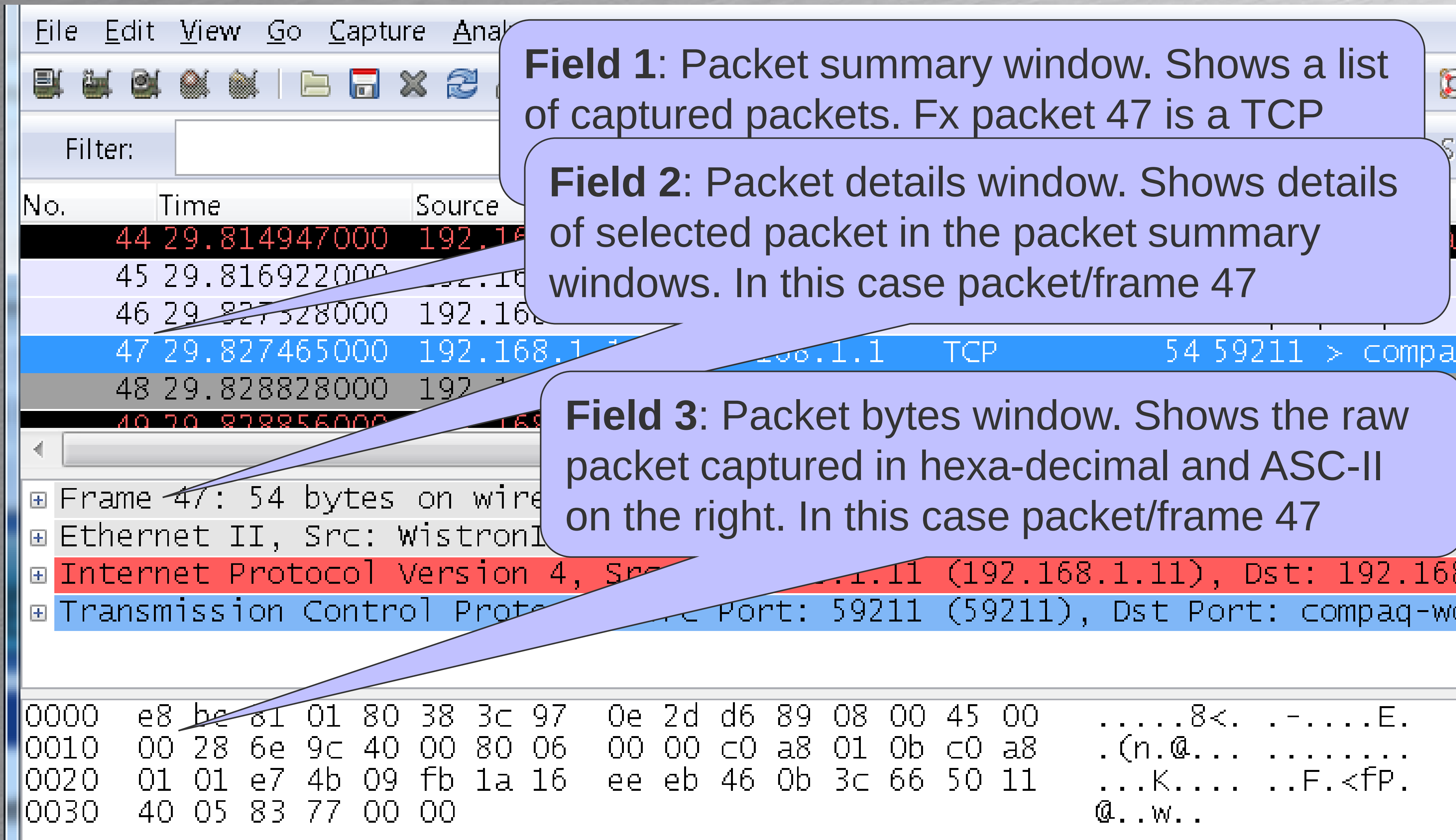
Capturing packets



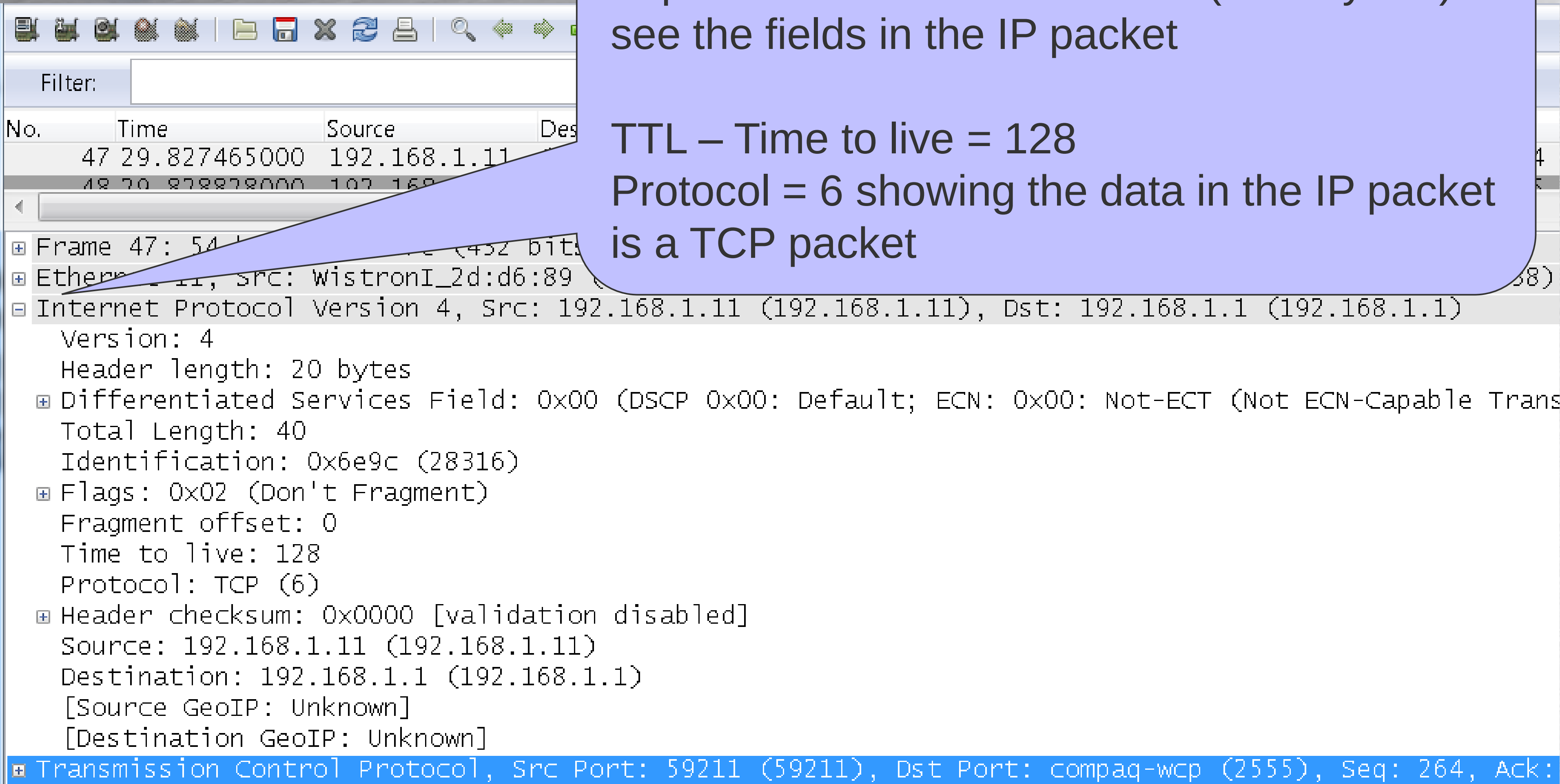
3: Select Start to capture

Note – you may stop capturing at any time selscting Capture->Stop

Capturing packets



Dissection of packets



Exploded the Internet Protocol (OSI layer 3) to see the fields in the IP packet

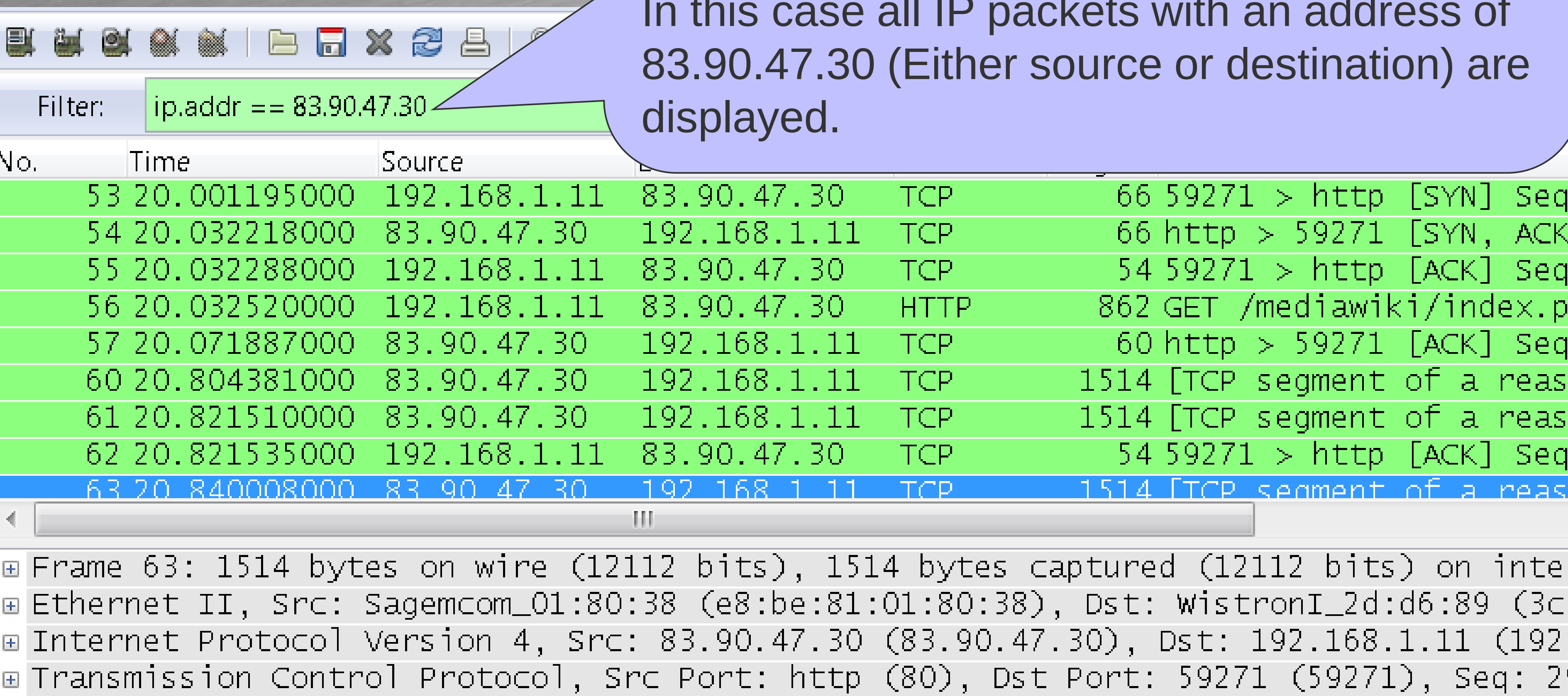
TTL – Time to live = 128

Protocol = 6 showing the data in the IP packet is a TCP packet

Packet filters

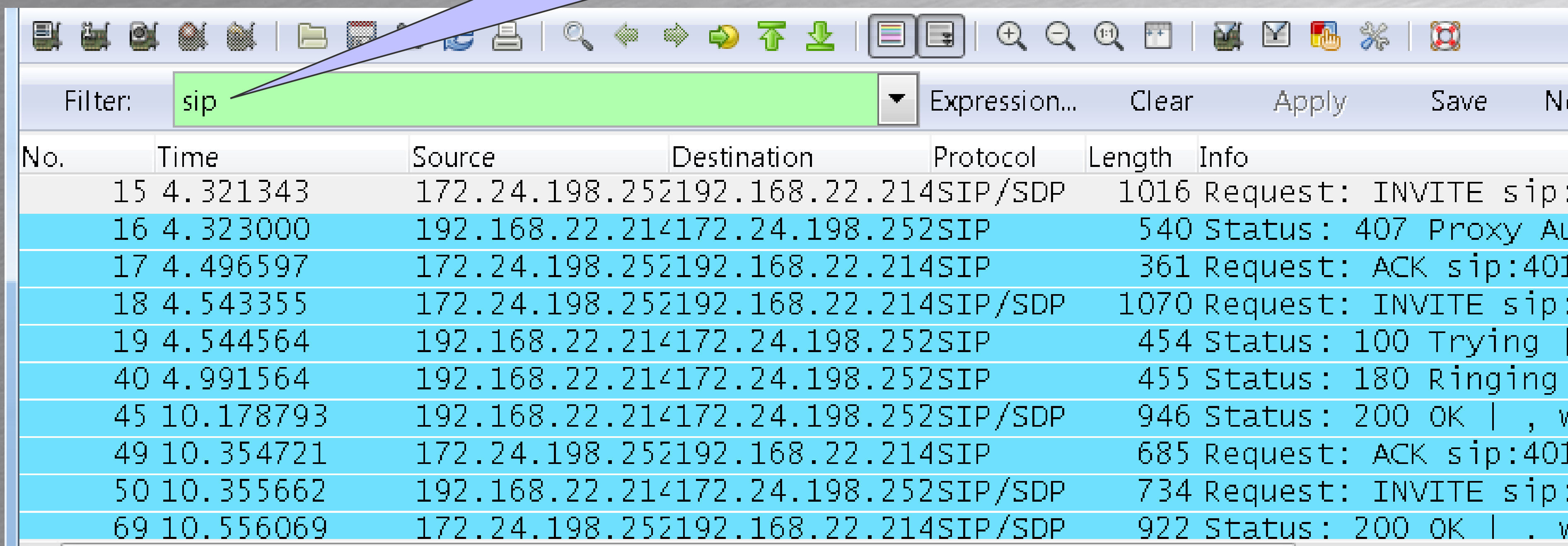
In the packet filter window it is possible to write an expression specifying which packets that should be displayed.

In this case all IP packets with an address of 83.90.47.30 (Either source or destination) are displayed.



Packet filters

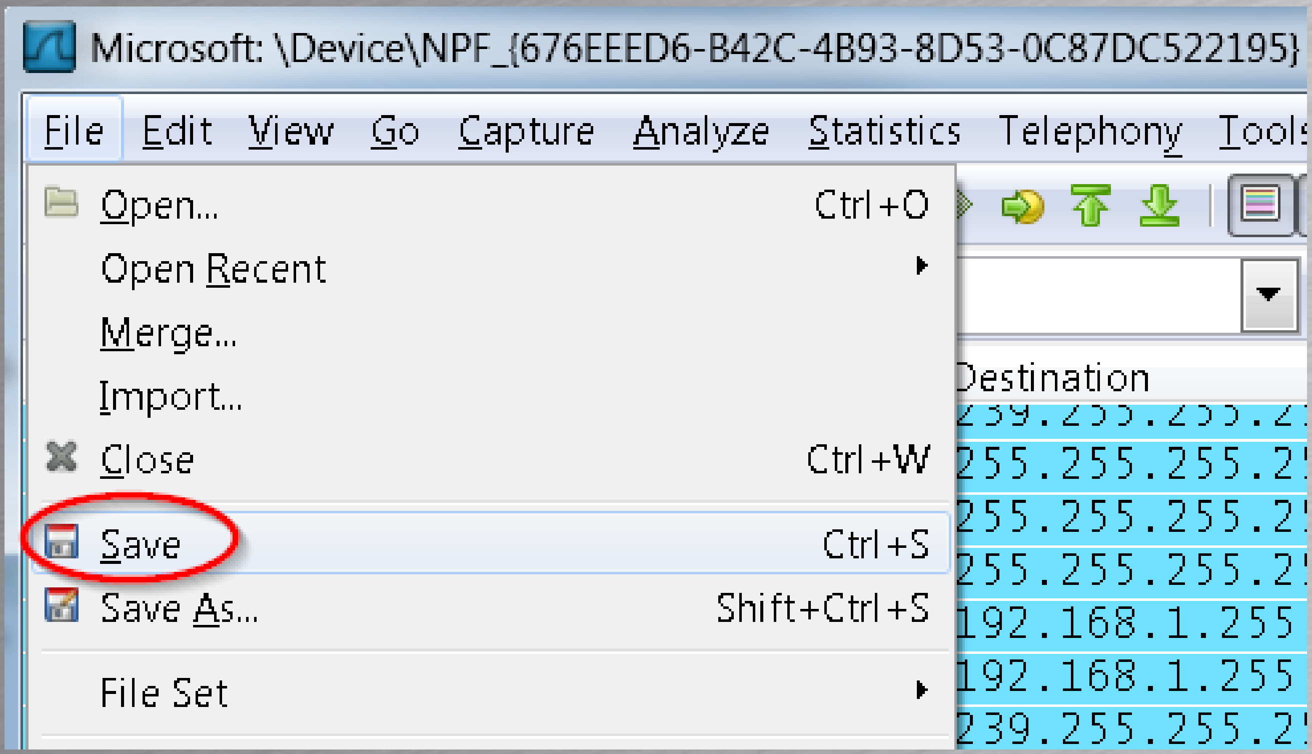
SIP packet filter –showing all SIP packets



No.	Time	Source	Destination	Protocol	Length	Info
15	4.321343	172.24.198.252	192.168.22.214	SIP/SDP	1016	Request: INVITE sip:
16	4.323000	192.168.22.21	172.24.198.252	SIP	540	Status: 407 Proxy Au
17	4.496597	172.24.198.252	192.168.22.214	SIP	361	Request: ACK sip:401
18	4.543355	172.24.198.252	192.168.22.214	SIP/SDP	1070	Request: INVITE sip:
19	4.544564	192.168.22.21	172.24.198.252	SIP	454	Status: 100 Trying |
40	4.991564	192.168.22.21	172.24.198.252	SIP	455	Status: 180 Ringing
45	10.178793	192.168.22.21	172.24.198.252	SIP/SDP	946	Status: 200 OK | , w
49	10.354721	172.24.198.252	192.168.22.214	SIP	685	Request: ACK sip:401
50	10.355662	192.168.22.21	172.24.198.252	SIP/SDP	734	Request: INVITE sip:
69	10.556069	172.24.198.252	192.168.22.214	SIP/SDP	922	Status: 200 OK | , w

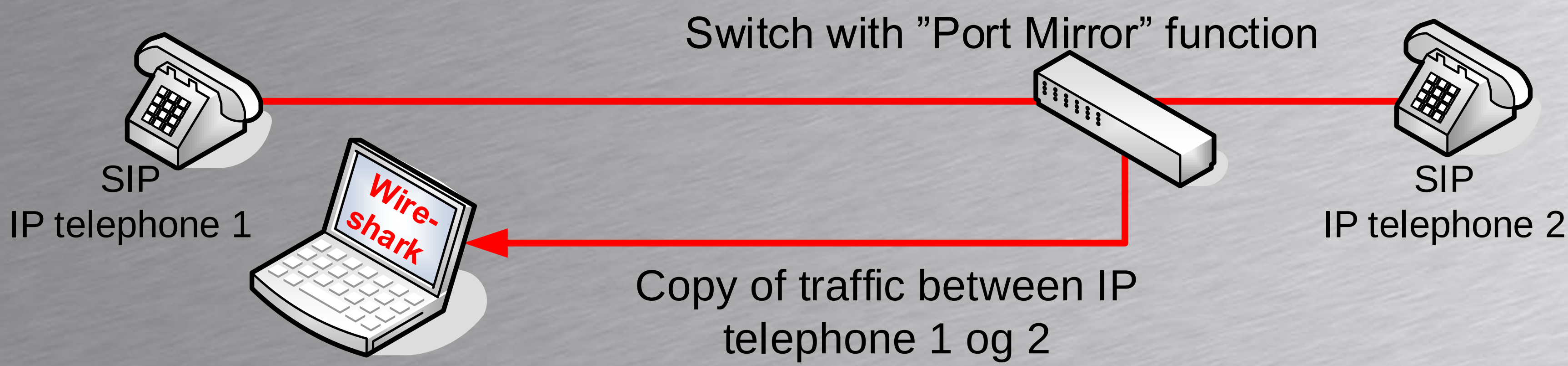
Saving captured packets

- When troubleshooting you can save captured packets and mail to expert
- Collect information for later analyze



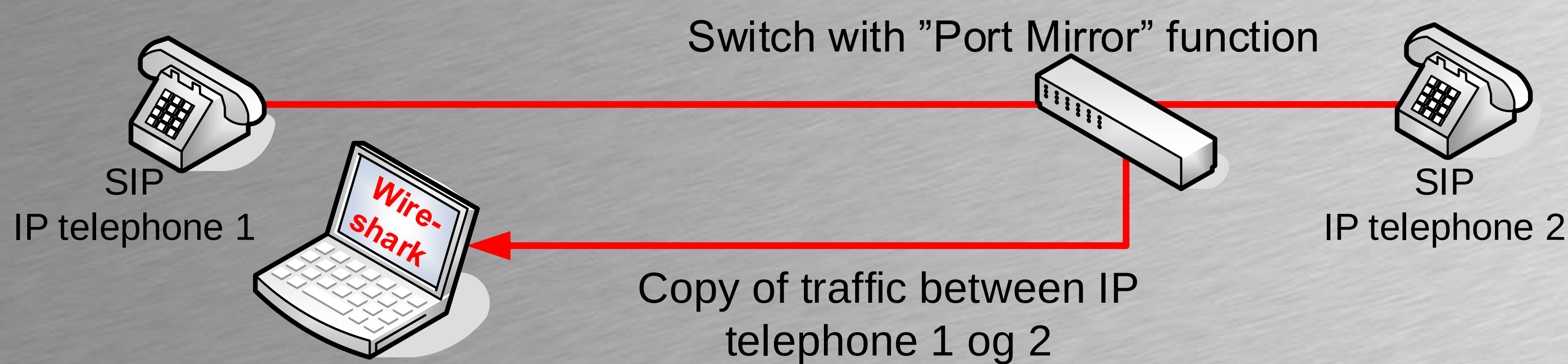
Access to packets

- For Wireshark to capture the packets – the packets must be received by your PC
  - Wired or wireless
- Capturing wired traffic may require additional setup



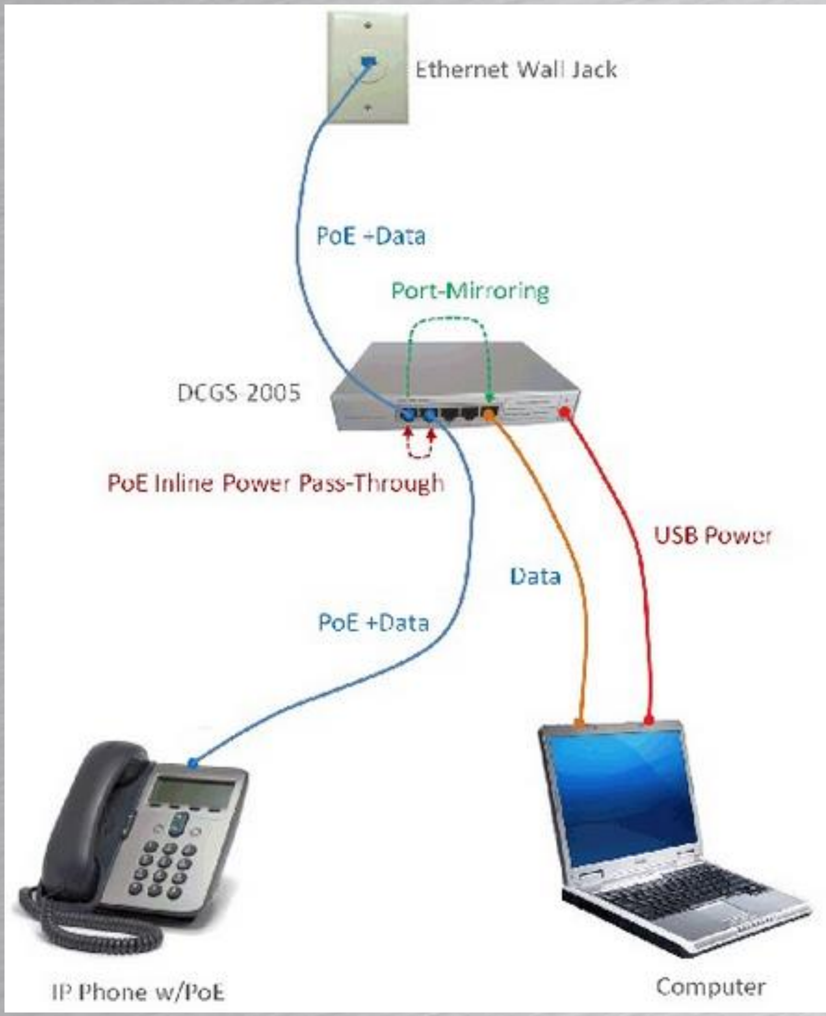
Mirror function

- Not all switches support mirror function
  - Zyxel name: Port mirroring
  - 3COM name: Roving Analysis Port (RAP)
  - Cisco name: Switched Port Analyzer (SPAN)



Network tap

- A small dedicated switch with build in mirroring
- Example from
  - Dualcomm
  - DCGS-2005



Task

- Capture SCCP and RTP Packets from the phone
- Create a visual document explaining the packets and the flow in a conversation.