



UNIVERSITA' DEGLI STUDI DI PARMA  
Dipartimento di Ingegneria dell'Informazione

## IP Telephony

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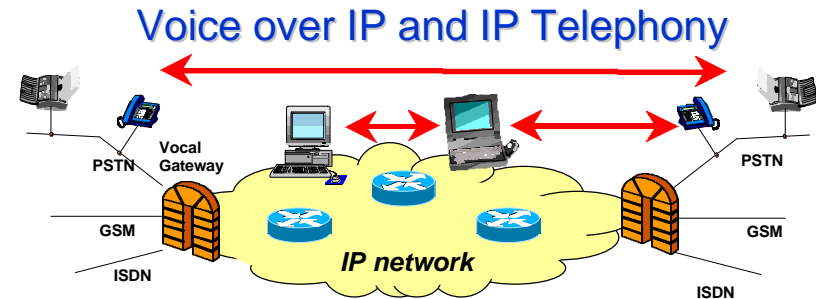
Corso di Reti di Telecomunicazioni C, a.a. 2004/2005

<http://www.tlc.unipr.it/veltri>



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IP Telephony



- The two terms are often used as synonymous...
- “Voice over IP”: support for voice communications using the IP Protocol over Internet/Intranet  
The phone path may be entirely or partially over IP network
- “IP telephony” and “Internet telephony” focus on the service...  
at least one terminal should be IP: the service is not PSTN
  - “IP telephony” is not only Telephony over IP
  - “IP telephony” includes conferencing, video communication, new services

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IP Telephony

## VoIP (initial) motivations

### Billing

- Initial stimulus
- IP is generally considered “free”; long distance phone calls were expensive

### Regulations

- Laws and authorities... deregulation

### Costs

- VoIP Gateways (GWs) are cheaper than in the past
- Availability of very small GWs (i.e. 2 ports)

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IP Telephony

## VoIP motivations

### Integration (network/transport aspect)

- the IP data traffic is dramatically growing with reference to voice traffic
- In the backbones: the future highways are built for IP traffic!
- Fax services over IP (fax traffic today considered as telephone traffic will be likely replaced by email)
- For the ISPs: NAS with VoIP GW functions (Internet Telephony Service Provider - ITSP)

### Integration (operation and management aspect)

- an integrated Voice/Data network can represent a large cost saving from the operational point of view
- for example within an enterprise the same people could handle the Data and Voice network... as the technology and the knowledge will be in common

### Services

- the Voice/Data integration over IP networks opens the door to new advanced services

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## The main players

### The traditional manufactures

Ericsson, Lucent, Nortel,

...

### Data manufactures

Cisco, Ascend, 3Com, ...

### New Comers

Vocaltec, NetSpeak, E-Fusion, ViennaSys, ...

### PSTN Operators

AT&T, DT, Telekom Finland, ...

### New Comers

Delta3, ITXC, GRIC, OzMail, Qwest, USA

Talks, ...

### ISP and ITSP

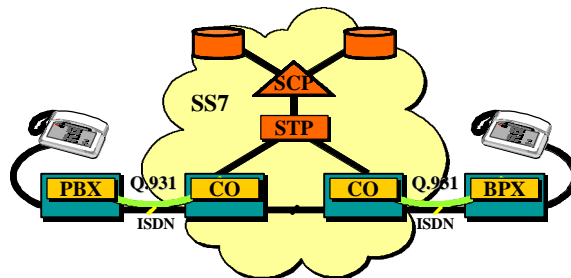
AOL, GEOnet, IDT ...

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## The PSTN – a reminder

## Once upon a time... the PSTN

- Circuit switching
  - PSTN (Public Switched Telephone Network),
  - ISDN (Integrated Services Digital Network)
- Main characteristics:
  - circuit setup
  - fixed bandwidth
  - fixed delay



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## PSTN services

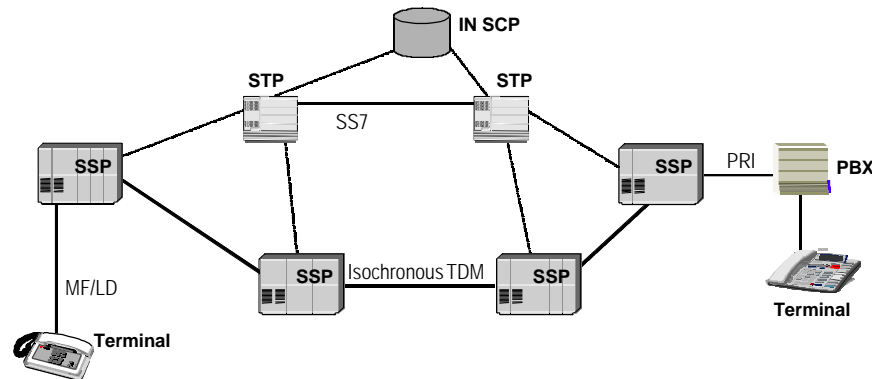
- Basic POTS (Plain Ordinary Telephone Service)
  - augmented with class services
  - call back when free, three-way calling, etc.
- Number Translation Services
  - (800-xxx) freephone, local call, premium call
- Virtual Private Networking, Centrex
  - aimed at corporates, integration with PBX
- Value-Added Services
  - calling card
  - voicemail

The voice community has always understood that the voice business lies in advanced services, not just the bearer service of voice connectivity (POTS)

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## PSTN Generic Architecture

- Dumb terminals (don't run IP - so can't be hosts)
- Two networks - a connection-oriented one for TDM voice and a separate connectionless network for signalling.
- Call Model runs in the SSP (ISUP & TCAP).



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## SS7 performance

- Less than 1 in 10 million messages lost ( $\leq 1$  in  $10E+7$ ).
- Less than 1 in 10 billion messages out of sequence, or duplicated ( $\leq 1$  in  $10E+10$ ).
- Less than 1 in a billion message errors should go undetected ( $\leq 1$  in  $10E+9$  undetected).
- End-to-end route availability more than 99.998% (less than ten minute downtime per year).
- Example end-to-end permitted delay for an "average" country:  
simple message = 390 ms; complex message = 600 ms.

This is quite challenging for an IP network – we will look at "signalling transport" later

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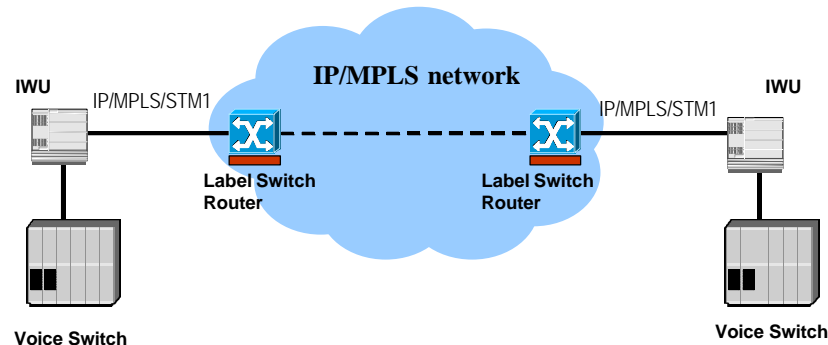
## PSTN Summary

- The Public Switched Telephone Network is a global network which has evolved over the decades to be a highly-engineered real-time system of great reliability.
- It is, however a network which is specialised to do only a few things very well – highly vertically integrated.
- The IP challenge is to replicate and transcend the engineering requirements of PSTN-grade voice in an open and extensible IP-based architecture.

## Different VoIP views

## Voice Transport

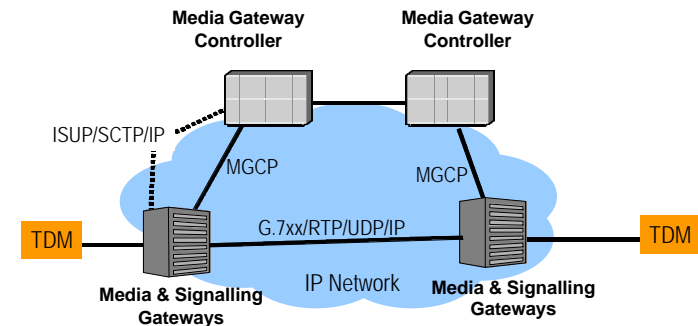
- Voice DS0 and signalling flows are carried transparently on PVCs (or LSPs)
- This view was most popular a few years ago



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## MEGACO - voice peering to the TDM world

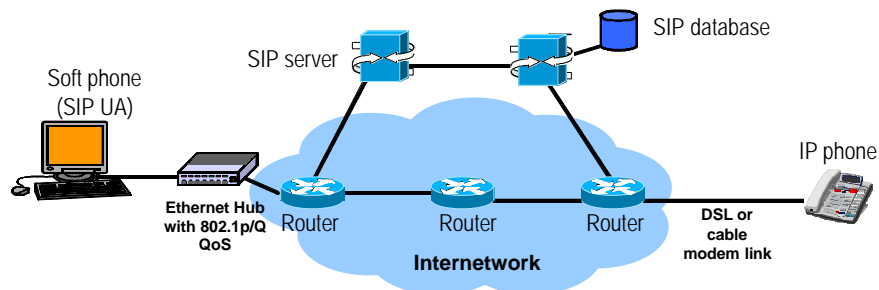
- The IP network emulates a PSTN switching peer
- It is the way IP-carriers are going to handle traditional TDM voice customers
- Interconnects partner services and enterprise (PBX) voice



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## IETF view of voice - Session Initiation Protocol

- Call model runs in intelligent end-devices
- Network Services run on network servers
- SIP protocol makes it all work (leveraging other protocols)
- Voice is special case of multiparty, multimedia session
- Similar to H.323 but faster & more flexible & within IETF paradigm.



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## Voice Transport over IP

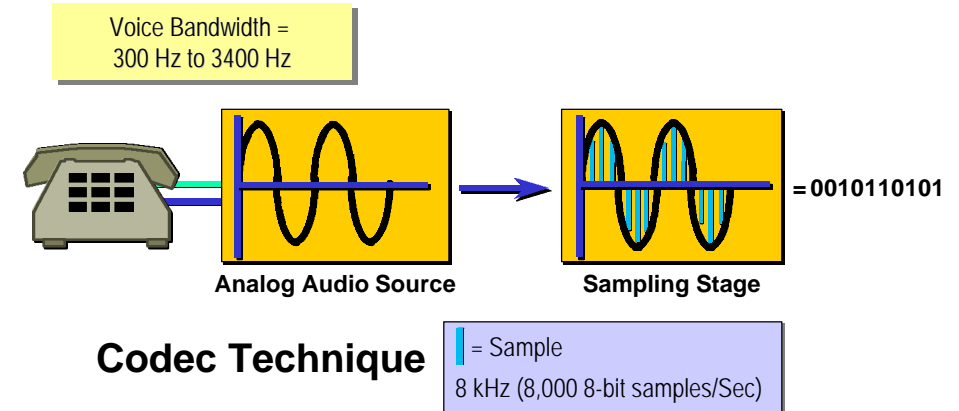
## Introduction: Packet telephony

- Once voice has been digitized, a number of processing and transmission steps occur before it reaches the eventual user, at the other end of the transmission link
  - digitization
  - compression methods
  - streaming and transmission
- Most of these steps result in delay, delay variation and loss

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## Analog to Digital

Pulse Code Modulation—"Nyquist Theorem" (Sample at twice the Frequency)



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## ITU-T G-series Voice coding

- The ITU-T G-series recommendations standardized different coding techniques
- The most popular coding standards are:
  - **G.711**—Describes the 64-kbps PCM voice coding technique. In G.711, encoded voice is already in the correct format for digital voice delivery in the public switched telephone network (PSTN) or through PBXes.
  - **G.723.1**—Describes a compression technique that can be used for compressing speech or audio signal components at a very low bit rate as part of the H.324 family of standards. This CODEC has two bit rates associated with it: 5.3 and 6.3 kbps. The higher bit rate is based on ML-MLQ technology and provides a somewhat higher quality of sound. The lower bit rate is based on CELP and provides system designers with additional flexibility.
  - **G.726**—Describes ADPCM coding at 40, 32, 24, and 16 kbps. ADPCM-encoded voice can be interchanged between packet voice, PSTN, and PBX networks if the PBX networks are configured to support ADPCM.
  - **G.728**—Describes a 16-kbps low-delay variation of CELP voice compression. CELP voice coding must be translated into a public telephony format for delivery to or through the PSTN.
  - **G.729**—Describes CELP compression where voice is coded into 8-kbps streams. There are two variations of this standard (G.729 and G.729 Annex A) that differ mainly in computational complexity; both provide speech quality similar to 32-kbps ADPCM.

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## ITU-T G-series Voice coding

Compression Method	Bit Rate (kbps)	Framing Size (ms)	MOS Score
<b>G.711 PCM</b>	<b>64</b>	<b>0.125</b>	<b>4.1</b>
<b>G.726 ADPCM</b>	<b>32</b>	<b>0.125</b>	<b>3.85</b>
<b>G.728 LD-CELP</b>	<b>16</b>	<b>0.625</b>	<b>3.61</b>
<b>G.729 CS-ACELP</b>	<b>8</b>	<b>10</b>	<b>3.92</b>
<b>G.729 x 2 Encodings</b>	<b>8</b>	<b>10</b>	<b>2.68</b>
<b>G.729 x 3 Encodings</b>	<b>8</b>	<b>10</b>	<b>2.68</b>
<b>G.729a CS-ACELP</b>	<b>8</b>	<b>10</b>	<b>3.7</b>
<b>G.723.1 MP-MLQ</b>	<b>6.3</b>	<b>30</b>	<b>3.9</b>
<b>G.723.1 ACELP</b>	<b>5.3</b>	<b>30</b>	<b>3.65</b>

Table taken from Cisco Systems:  
[http://www.cisco.com/univercd/cc/td/doc/product/software/ios113ed/113t/113t\\_1/voip/voipover.htm](http://www.cisco.com/univercd/cc/td/doc/product/software/ios113ed/113t/113t_1/voip/voipover.htm)

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## Voice coding (ITU-T + ETSI)

Standards Body	ITU-T	ITU-T	ITU-T	ITU-T	ITU-T	ETSI	ETSI	ETSI
Recc.	G.726	G.728	G.729	G.729A	G.723	GSM (FR)	GSM (HR)	GSM (EFR)
Year	1990	1992	1995	1996	1995	1987	1994	1995
Bit Rate Kbit/s	16-40	16	8	8	6.3 & 5.3	13	5.6	12.2
Quality	≤PSTN	PSTN	PSTN	PSTN	≤ PSTN	< PSTN	=GSM	PSTN
Frame Size	0.125 ms	0.625 ms	10 ms	10 ms	30 ms	20 ms	20 ms	20 ms
Look Ahead	0	0	5 ms	5 ms	7.5 ms	0	4.4 ms	0
MOS	4.1	4.0	3.9	< 3.9	3.8	3.5	3.5	> 4

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## Bandwidth requirements

Encoding/ Compression	Result Bit Rate
G.711 PCM A-Law/u-Law	64 kbps
G.726 ADPCM	16, 24, 32, 40 kbps
G.727 E-ADPCM	16, 24, 32, 40 kbps
G.729 CS-ACELP	8 kbps
G.728 LD-CELP	16 kbps
G.723.1 CELP	6.3/5.3 kbps Variable

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## G723.1 - 5333 bit/s

frame [bytes]	frame [ms]	frames/ packet	packets/ sec	payload [bytes]	packet [bytes]	bits/sec	% optimal	latency [ms]
20	30	1	33,33	20	60	16000	300%	30
20	30	2	16,67	40	80	10667	200%	60
20	30	3	11,11	60	100	8889	167%	90
20	30	4	8,33	80	120	8000	150%	120
20	30	5	6,67	100	140	7467	140%	150
20	30	6	5,56	120	160	7111	133%	180
20	30	7	4,76	140	180	6857	129%	210
20	30	8	4,17	160	200	6667	125%	240
20	30	9	3,70	180	220	6519	122%	270
20	30	10	3,33	200	240	6400	120%	300
20	30	11	3,03	220	260	6303	118%	330
20	30	12	2,78	240	280	6222	117%	360

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## G723.1 - 6400 bit/s

frame [bytes]	frame [ms]	frames/ packet	packets/ sec	payload [bytes]	packet [bytes]	bits/sec	% optimal	latency [ms]
24	30	1	33,33	24	64	17067	267%	30
24	30	2	16,67	48	88	11733	183%	60
24	30	3	11,11	72	112	9956	156%	90
24	30	4	8,33	96	136	9067	142%	120
24	30	5	6,67	120	160	8533	133%	150
24	30	6	5,56	144	184	8178	128%	180
24	30	7	4,76	168	208	7924	124%	210
24	30	8	4,17	192	232	7733	121%	240
24	30	9	3,70	216	256	7585	119%	270
24	30	10	3,33	240	280	7467	117%	300
24	30	11	3,03	264	304	7370	115%	330
24	30	12	2,78	288	328	7289	114%	360

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## G.711 - 64000 bit/s

frame [bytes]	frame [ms]	frames/ packet	packets/ sec	payload [bytes]	packet [bytes]	bits/sec	% optimal	latency [ms]
240	30	1	33,33	240	280	74667	117%	30
240	30	2	16,67	480	520	69333	108%	60
240	30	3	11,11	720	760	67556	106%	90
240	30	4	8,33	960	1000	66667	104%	120
240	30	5	6,67	1200	1240	66133	103%	150
240	30	6	5,56	1440	1480	65778	103%	180
240	30	7	4,76	1680	1720	65524	102%	210
240	30	8	4,17	1920	1960	65333	102%	240
240	30	9	3,70	2160	2200	65185	102%	270
240	30	10	3,33	2400	2440	65067	102%	300
240	30	11	3,03	2640	2680	64970	102%	330
240	30	12	2,78	2880	2920	64889	101%	360

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## LPC10 - 2489 bit/s

frame [bytes]	frame [ms]	frames/ packet	packets/ sec	payload [bytes]	packet [bytes]	bits/sec	% optimal	latency [ms]
7	22,5	1	44,44	7	47	16711	671%	22,5
7	22,5	2	22,22	14	54	9600	386%	45
7	22,5	3	14,81	21	61	7230	290%	67,5
7	22,5	4	11,11	28	68	6044	243%	90
7	22,5	5	8,89	35	75	5333	214%	112,5
7	22,5	6	7,41	42	82	4859	195%	135
7	22,5	7	6,35	49	89	4521	182%	157,5
7	22,5	8	5,56	56	96	4267	171%	180
7	22,5	9	4,94	63	103	4069	163%	202,5
7	22,5	10	4,44	70	110	3911	157%	225
7	22,5	11	4,04	77	117	3782	152%	247,5
7	22,5	12	3,70	84	124	3674	148%	270

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## Voice Transport Requirements

- The impairments in the transport of Voice over IP are:
  - loss
  - delay
  - delay variation (jitter)
- This transport impairments impact the perceived voice quality
- A subjective measurement of voice quality is still used, called Mean Opinion Score (MOS)

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## Speech quality and Tradeoff

- Each CODEC provides a certain quality of speech
  - The quality of transmitted speech is a subjective response of the listener
- A common benchmark used to determine the quality of sound produced by specific CODECs is the mean opinion score (MOS)
  - wide range of listeners judge the quality of a voice sample (corresponding to a particular CODEC) on a scale of 1 (bad) to 5 (excellent)
  - the scores are averaged to provide the mean opinion score for that sample
- From a financial standpoint it might seem logical to convert all calls to low-bit rate CODECs
- One of the drawbacks compressing voice is signal distortion due to multiple encodings (called tandem encodings)
  - For example, when a G.729 voice signal is tandem encoded three times, the MOS score drops from 3.92 (very good) to 2.68 (unacceptable)
- Another drawback is CODEC-induced delay with low bit-rate CODECs

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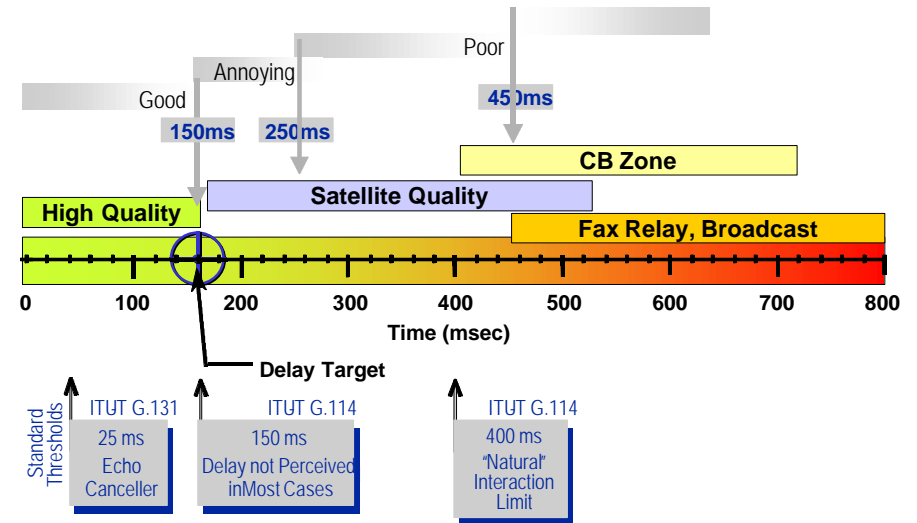
## Mean Opinion Score (MOS)

Rating	Speech Quality	Level of Distortion
5	Excellent	Imperceptible
4	Good	Just perceptible, not annoying
3	Fair	Perceptible, slightly annoying
2	Poor	Annoying but not objectionable
1	Unsatisfactory	Very annoying, objectionable

**MOS of 4.0 = Toll Quality**

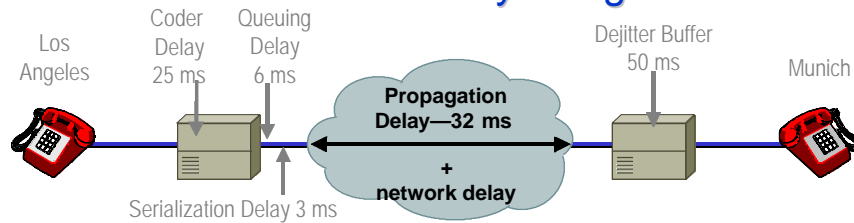
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## Cumulative One-Way Transmission Delay



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## Calculate Delay Budget



	Fixed Delay	Variable Delay
Coder Delay G.729 (10 ms + 5 ms look ahead)	15 ms	
Coder Delay G.729 (10 msec per frame)	10 ms	
Packetization Delay—Included in Coder Delay	~ 0	
Queuing Delay 64 kbps Trunk		6 ms
Packet Serialization Delay 64 kbps Trunk (*)	3 ms	
Propagation Delay	32 ms	
Network Delay		?
Dejitter Buffer	50 ms	
Total	110 ms	

Delay budget:  
if the maximum tolerable end to end delay is 150 ms, the fixed network delay must be < 40 ms the variable network delay must be < dejitter delay

(\*) compressed packet header (8 bytes) and 20 byte payload

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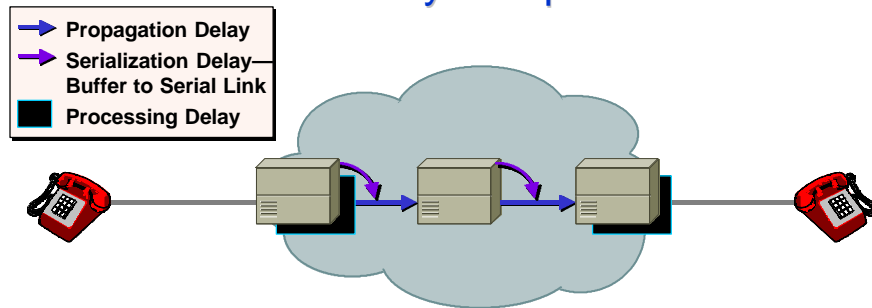
## End-to-end delay components

- End-to-end delay is given by:
  - Fixed component:
    - voice processing delay (voice coding and packetization)
    - propagation delay
    - serialization delay
  - Variable component
    - Delay introduced by the network (queuing delay, packet processing)
    - Variable packet sizes
- "Dejitter" buffers compensate the variable component

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## Fixed Delay Components



- Processing
  - Coding/compression/decompression/decoding
  - Packetization
- Propagation—six microseconds per kilometer
- Serialization

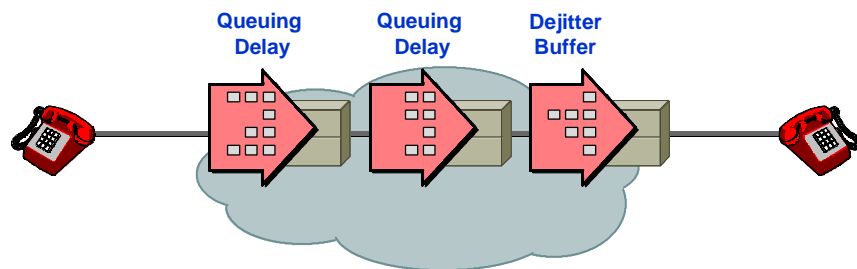
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## Serialization Delay

	Frame Size						
	1 Byte	64 Bytes	128 Bytes	256 Bytes	512 Bytes	1024 Bytes	1500 Bytes
Link Speed	56kbps	143us	9ms	18ms	36ms	72ms	214ms
	64kbps	125us	8ms	16ms	32ms	64ms	187ms
	128kbps	62.5us	4ms	8ms	16ms	32ms	93ms
	256kbps	31us	2ms	4ms	8ms	16ms	46ms
	512kbps	15.5us	1ms	2ms	4ms	8ms	23ms
	768kbps	10us	640us	1.28ms	2.56ms	5.12ms	15ms
	1536kbs	5us	320us	640us	1.28ms	2.56ms	7.5ms

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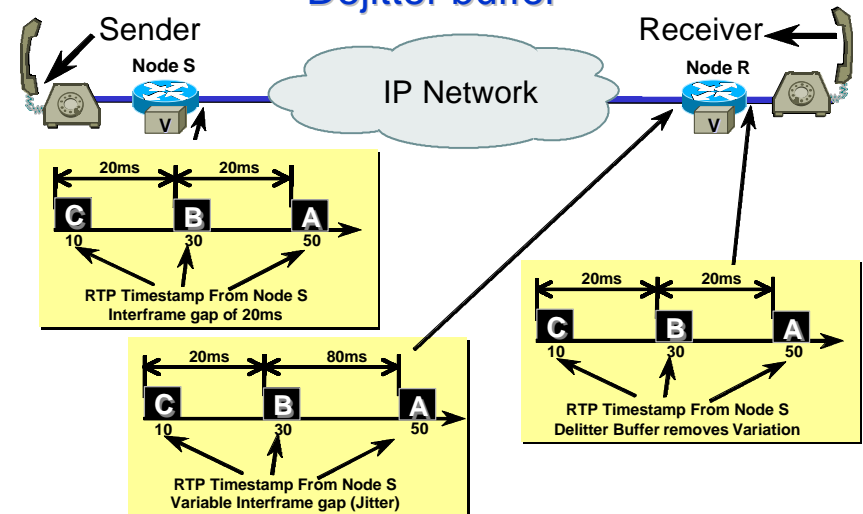
## Variable Delay Components



- Queuing delay
- Variable packet sizes

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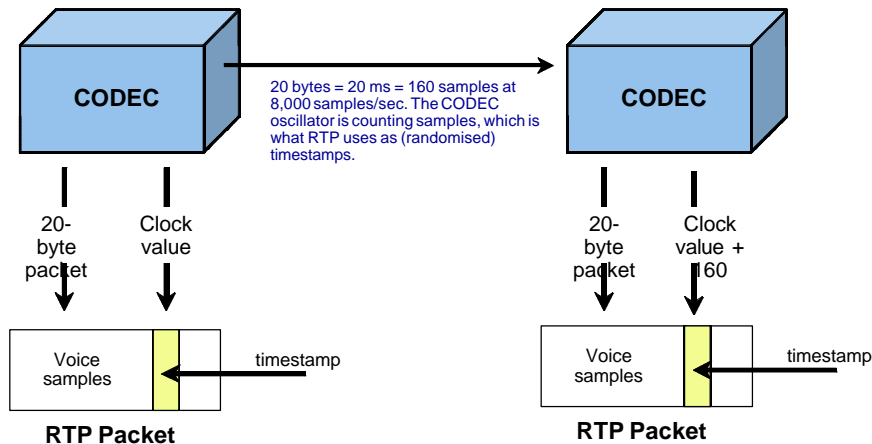
## Dejitter buffer



- Large packets can cause playback buffer underrun, resulting in slight voice degradation
- Jitter or playback buffer can accommodate some delay/delay variation

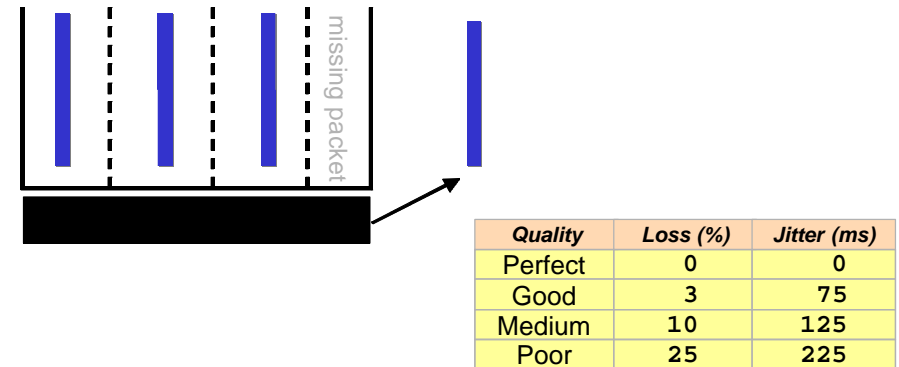
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## The mechanism of RTP "timing"



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## VoIP and Packet Loss



- The friendly "retransmission" is of no use in the voice world... late is as good as never
- VoIP coding is tolerant to occasional loss
- Complex strategy in algorithm can interpolate lost packets based on context

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## QoS parameters in the global IP network

- As an IP packet enters a Service Provider IP network, the major impairments it can suffer are the following:
  - **Unavailability of service**
    - In the case of a fibre breakage, or equipment failure, throughput can vanish until recovery mechanisms complete.
  - **Delay**
    - Packets are delayed when they are processed (e.g. through encryption gateways), when they encounter non-empty queues in routing devices, and in playback buffers (jitter-removal)
  - **Jitter**
    - Packets accumulate jitter when they encounter varying router queue occupancies on their path through the network. As a result, they incur a different overall delay than that of their predecessors or successors. This skews the timing relationships between successive packets
  - **Packet Loss**
    - Packets are lost when they encounter a queue in a router which is completely full, or when they are subject to policy-based discard - e.g. they are out-of-profile of their SLA

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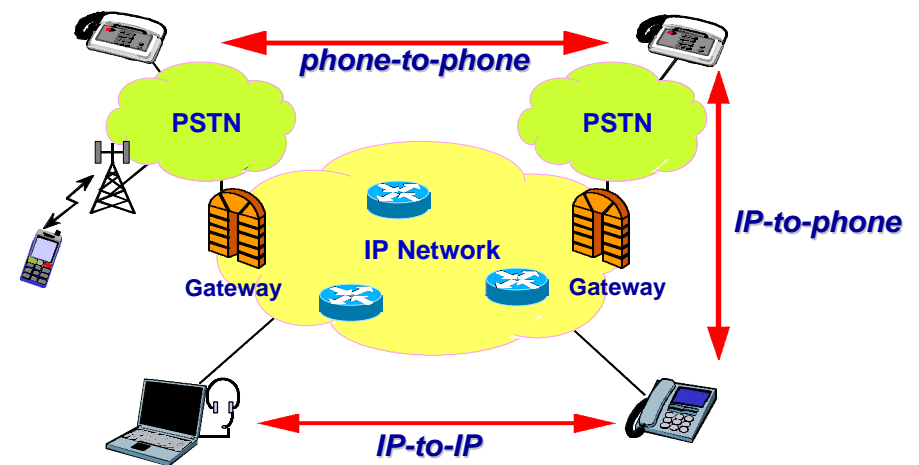
## Typical view of QoS parameters

- **Availability:**
  - 99.99% = 4 mins downtime/month; 99.999% = 25 seconds downtime/month
  - Enterprise voice: 99.999% desktop-to-desktop. For enterprise customers, planned downtime in slack periods may be permissible
  - Public telephony voice: 99.999% Media Gateway-to-Media Gateway. No planned outages in service
- **Resilience:**
  - Node or link failure recovered within 50 ms to avoid impairments to voice calls and Internet TV
- **Delay, or Latency (maximum):**
  - 70 ms PoP-PoP network RTT delay for IP transit services
  - 100 ms desktop to network to server RTT for transactional ASP services
  - 150 ms network and application one-way delay for voice/multimedia (G.114)
- **Jitter:**
  - Less than 2% of one-way delay as seen by the customer
- **Packet loss:**
  - < 1.5% "usually" but we need to specify time-dynamics
  - TCP traffic (web, email) is resilient to episodic packet loss (triggering rate-adaptive behaviour)
  - RTP/UDP traffic (voice, video) is resilient to small packet loss, and brittle to larger packet loss

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## VoIP/IP Telephony

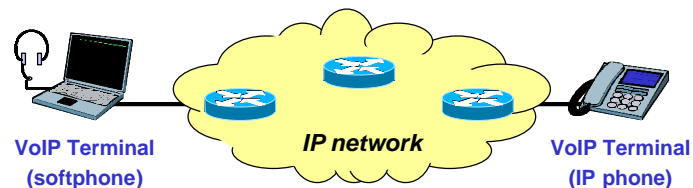
### VoIP Scenarios



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### VoIP Scenarios: IP-to-IP

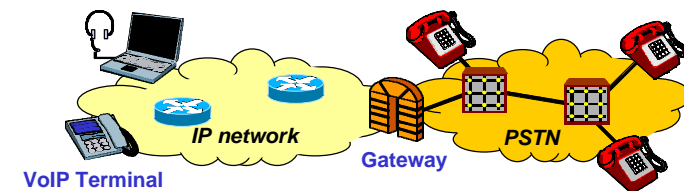
- IP-to-IP
  - The PC-to-PC architecture was historically the first to be implemented (Vocaltec, "Internet Phone", 1995)
  - Need a host with audio card, mic, + telephony software (Netmeeting, etc)
  - Examples: Internet phone, corporate telephony service
  - Optional services: video, chat, data sharing...



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### Scenarios: IP to Phone

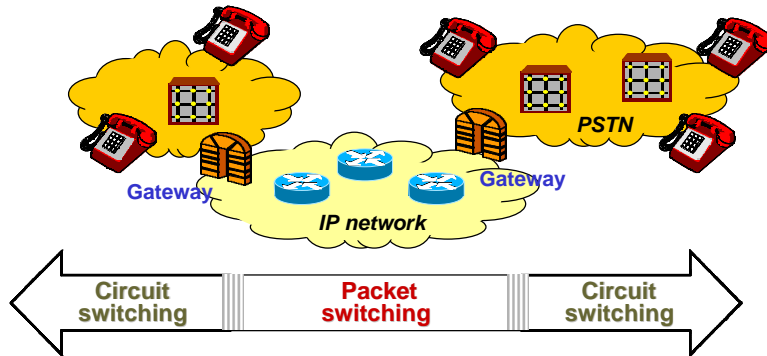
- IP-to-SCN/PSTN
  - Need a gateway that connects IP network to phone network (also called IWU InterWorking Unit)
  - It realizes the interworking: voice signal, service control (signaling, routing, ...)



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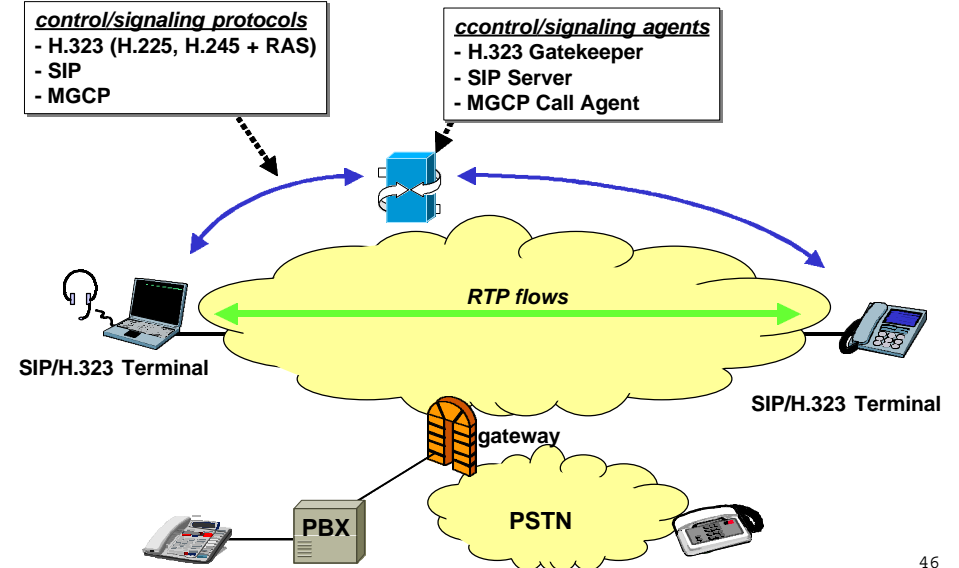
## VoIP Scenarios

- SCN-to-IP-to-SCN (phone-to-phone via an IP network)
  - Need more gateways
  - The IP network could be dedicated intranets or the Internet
  - The phone networks could be intra-company PBXs or the carrier switches



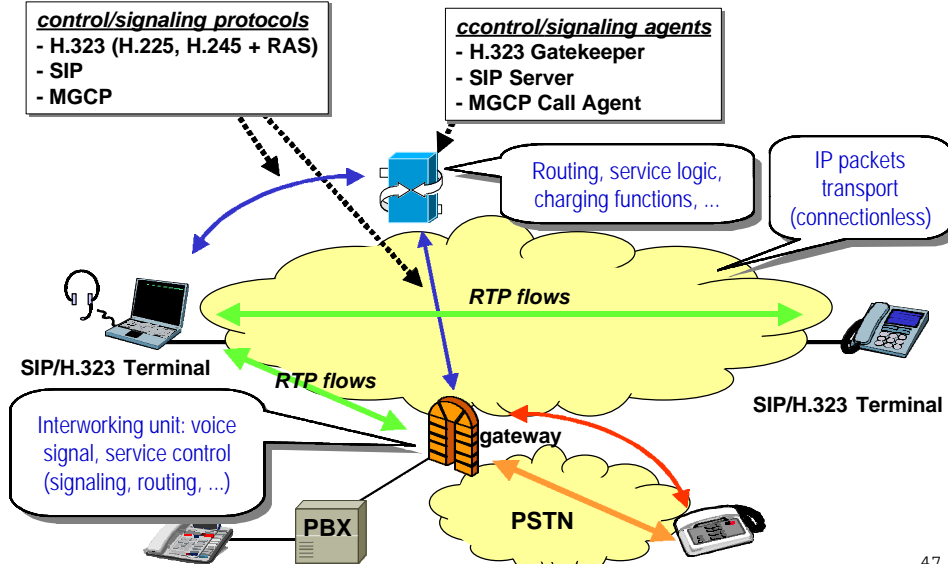
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## VoIP/IP Telephony service basics



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## VoIP general architecture



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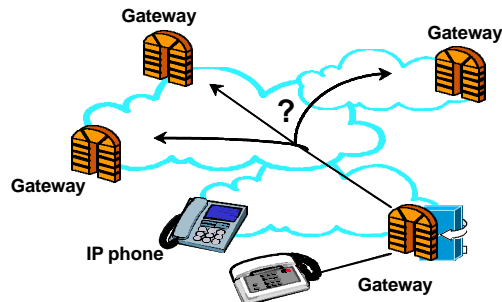
## The Gateway

- It is the IWU (inter-working unit) between an IP network and a Switched Circuit Network (SCN) (ex.: PBX, ISDN, POTS)
- A Gateway takes a phone call from the SCN, digitizes it if not already digital, compresses and packetizes it into IP datagrams
- Reverse for the other way (full-duplex)
- It acts as gateway for both:
  - user plane (data)
  - control plane (signaling)
- Gateways are often based on workstation platforms

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## Gateway Location Problem

- When
  - PC to PSTN Calling
  - PSTN to IP to PSTN calling
- Must locate a remote IP-PSTN gateway
  - Need to "learn" about remote gateways
  - Gateways can be also within another administrative domain



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## Markets

- IP-to-IP over Internet
- Corporate IP Telephony in place of PBX
- Corporate toll-bypass
- IP based public phone service
  - PC-to-phone
  - phone-to-phone
- New services..

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## Technical issues

- Large delay and delay jitter
- Packet length
- Voice compression and silence suppression
- Lost of packets
  - replacement of lost packets by silence, extrapolate previous waveform, etc.
- Echo cancellation
- Address translation: Phone # to IP; directory servers
- Telephony signaling: Different PBXs may use different signaling methods
- Multiplexing
  - Subchannel multiplexing → Multiple voice calls in one packet
- Security
  - Firewalls may not allow incoming IP traffic
  - Insecurity of Internet
- Charging
  - Not a technical issue.. ;-)

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## The phone works — why bother with VoIP?

### User prospective

- Voice over IP
  - variable compression/quality
  - low cost calls
  - no local access fees
- security through encryption
- caller, talker identification
- better user interface
- video, whiteboard, and new services

### Carrier prospective

- voice and silence suppression → low traffic
- integration of data and voice switching (only one network)
- operational advantages
- integration of data and voice services
- new services

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## IP Telephony basic components

- Codec/decoder (audio, video)
- User plane
  - RTP/RTCP/UDP/IP
  - IP Multicast
- Signaling: Call Setup, Tear-down
  - H323 (ITU-T standard includes H.225.0, H.245, Q.931)
  - Session Initiation Protocol (SIP) (IETF)
  - Media GW control (GCP) (IETF/ITU-T)
- Gateways (H.323-to-PSTN, SIP-to-PSTN, etc.)
- QoS: Resource Reservation
  - RSVP
  - Differentiated Services
- Policy issues: billing, firewall access, AAA (Authentication, Authorization, Accounting)

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## Other multimedia components

- SDP - Session Description
  - Describes the characteristics of a multimedia session such as time, format, address/ports, purpose, etc.
  - SDP may distributed using SAP, e-mail or the Web
- Call processing scripts
  - Specifies behaviour of a server or terminal in responding to incoming and outgoing calls
- SAP - Session Announcement
  - announcement protocol for multimedia sessions
  - uses IP multicast to a well-known address/port
  - SAP/SDP are used to advertise multimedia session information to a set of potential participants
- RTSP - Media Server Control
  - VCR like controls
    - Fast-forward, play, rewind, pause, record

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## VoIP protocol stack

<i>media</i>	<i>session control</i>
audio/video codec	H.323, SIP, SDP, alt.
RTP/RTCP	
UDP	TCP or UDP
IP ( + QoS/RSVP/Diffserv )	
layer 2 technologies (MPLS, ATM, SDH, ...)	

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## ITU-T Study Group 16 Multimedia Services and Systems

- Responsible for studies relating to multimedia service definition and multimedia systems, including the associated terminals, modems, protocols and signal processing
- Activities
  - H.323 (signaling protocol between terminals, gateways and gatekeepers)
  - H.225 and H.245 (Media Capabilities)
  - H.248 (protocol between MG and MGC), in conjunction with IETF MEGACO
  - G.729 (8 kbit/s voice codec)

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## Main IETF Working Groups (WGs)

- **AVT** - Audio/Video Transport
  - formed to specify a protocol for real-time transmission of audio and video (RTP)
  - the associated profile for audio/video conferences and payload formats (e.g. MPEG-4, PureVoice)
- **SIGTRAN** - Signaling Transport
  - transport of packet-based PSTN signaling (such as Q.931 or SS7 ISUP) over IP Networks, between IP nodes such as a Signaling Gateway and Media Gateway Controller or Media Gateway
- **MEGACO** - Media Gateway Control
- **MMUSIC** - Multiparty Multimedia Session Control
  - to develop Internet standards track protocols to support Internet teleconferencing sessions
- **IPTEL** - IP Telephony
  - to develop protocols for Internet telephony (ex. signaling and capabilities exchange)
  - e.g. specification of the Call Processing syntax, specification of the service model, protocols, etc.
- **SIP** - Session Initiation Protocol  
Sviluppo ed estensione di SIP

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## IETF AVT - Audio/Video Transport

- The Audio/Video Transport Working Group was formed to specify a protocol for real-time transmission of audio and video over UDP and IP multicast
- This is the Real-time Transport Protocol, RTP, together with its associated profile for audio/video conferences and payload formats
- The payload formats currently under discussion include a number of media specific formats (MPEG-4, H.264, etc.)

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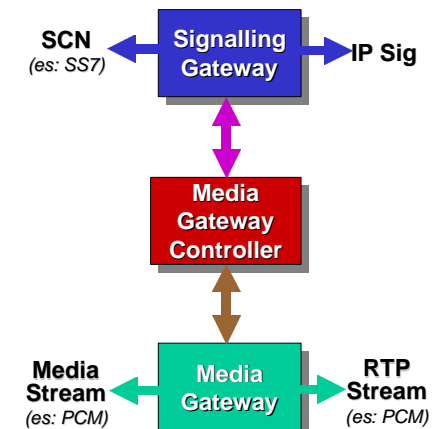
## IETF SIGTRAN - Signaling Transport

- The primary purpose of this working group is to address the transport of packet-based PSTN signaling over IP Networks, taking into account functional and performance requirements of the PSTN signaling
- For interworking with PSTN, IP networks will need to transport signaling such as Q.931 or SS7 ISUP messages between IP nodes such as a Signaling Gateway (SGW) and Media Gateway (MGW) or MGW Controller

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## IETF MEGACO - Media Gateway Control

- The working group develops an architecture for controlling Media Gateways (MGW) from external control elements such as a MGW Controller
- Megaco Working Group in close cooperation with ITU-T Study Group 16 developed the Gateway Control Protocol Version 1 (RFC 3525)
  - ex MEGACO/H.248, ex MGCP



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## IETF MMUSIC - Multiparty Multimedia Session Control

- The Multiparty MULTimedia Sesslon Control (MMUSIC) Working Group was chartered to develop protocols to support Internet teleconferencing and multimedia communications
- The group is now focussed on the revisions of these protocols in the light of implementation experience and additional demands that have arisen from other WGs (such as AVT, SIP, SIPPING, and MEGACO)
- MMUSIC WG also maintains and revises the specification of:
  - **Session Description Protocol (SDP)**
  - **Real Time Streaming Protocol (RTSP)**

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## IETF IPTEL - IP Telephony

- The primary purpose of this working group is to develop protocols for Internet telephony (ex. signaling and capabilities exchange)
- The focus is on the problems related to naming and routing for Voice over IP (VoIP) protocols

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## IETF SIP - Session Initiation Protocol

- The Session Initiation Protocol (SIP) working group is chartered to develop SIP protocol, currently specified by RFC 3261
- SIP is a text-based protocol, similar to HTTP and SMTP, for initiating interactive communication sessions between users
- Such sessions include voice, video, chat, interactive games, and virtual reality
- The SIP working group will concentrate on the specification of SIP and its extensions, and will not explore the use of SIP for specific environments or applications

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## 3GPP

- Third Generation Partnership Project
  - **Standardization of UMTS**
  - **IP Multimedia Subsystem (IMS)**

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