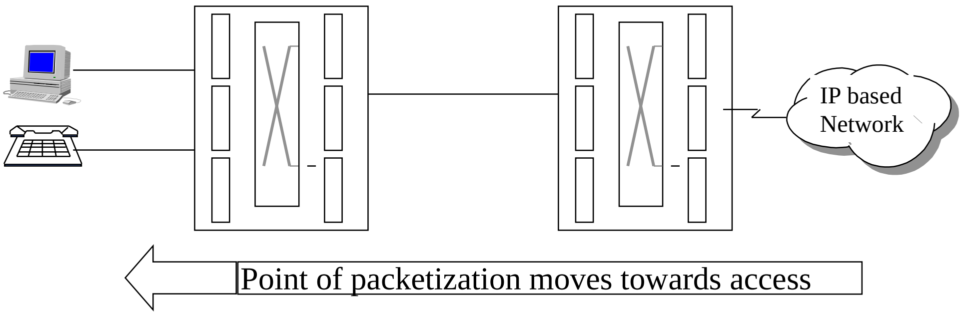
IP Telephony

Overview of IP Telephony

RTP, RTCP

Quality of Service

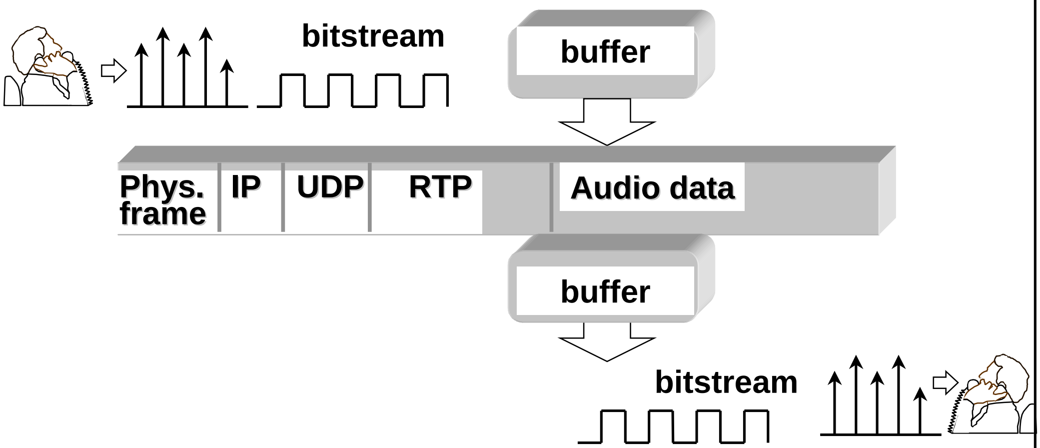
Data traffic will dominate voice in volume. Therefore Data will drive the Network Architecture.



- Broadband Networks will be based on packet switching
- BB network emerges from the existing Internet
- Each step of Development pays for itself.

VoIP in action

Coded samples (G.711, G.729B, G.723.1)

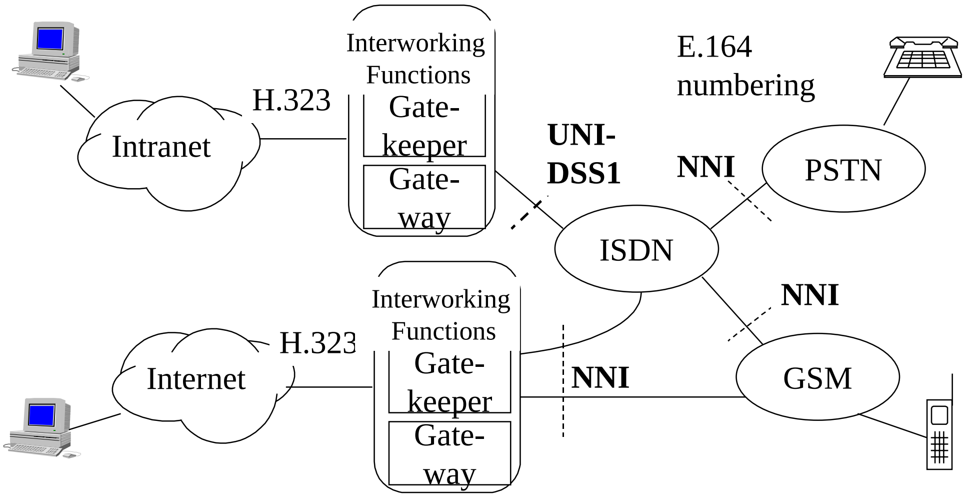


IP Telephony Standardization is active on de-jure and de-facto fora

- ITU-T - H.3xx, H.2xx series
- ETSI - TIPHON project - Telecommunications and Internet Protocol Harmonisation over Networks

- IPTEL and PINT - WGs of the IETF
- MMUSIC - WG of the IETF (Multiparty Multimedia Session Control)
- VOIP - Voice over IP by IMTC - Int'l Multimedia Teleconferencing Consortium

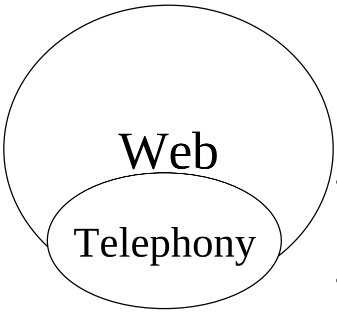
TIPHON specifies IP Voice to PSTN/ISDN/GSM Interworking



H.323 products are available

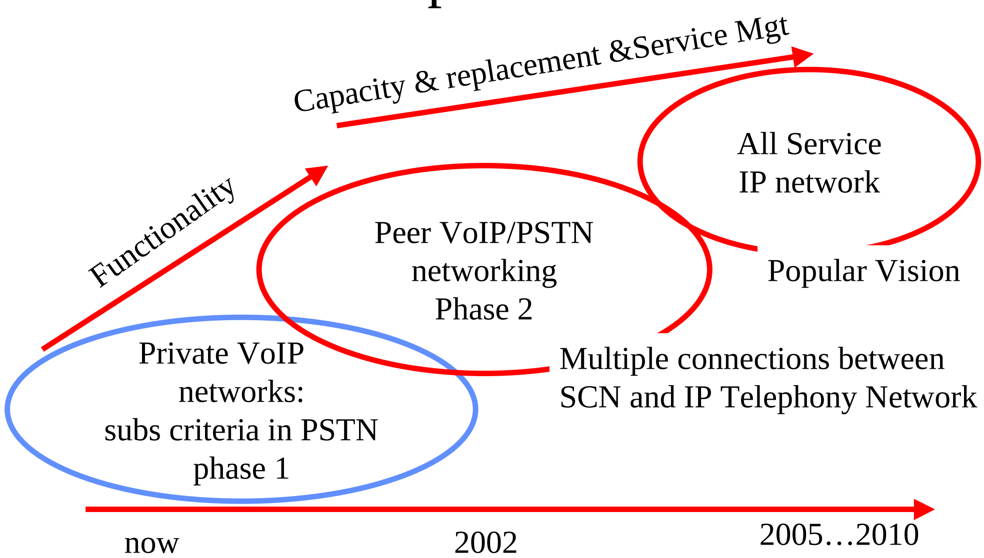
- ITSPs are strongly committed to H.323
- MS Netmeeting, Intel Videophone, Netscape Conference are examples of H.323 clients
- H.323 version 2 products are available
- Gateways and Gatekeepers/Call managers are available

IETF alternatives to H.323 pursue Integration of Telephony to the Web



- PINT works on Click-to-Dial, Click-to-Fax, Click-to-Fax-Back “www-buttons”. The idea is to integrate www to IN
- MMusic work on SIP - idea is to use web-technology to absorb signaling.
- CTI Simple Computer-Telephony-Protocol is based on similar ideas as SIP.

Roadmap to the Future



Interoperability Issues

- Signaling and Call control
- Quality of Service

Phase 1 --->

- Telephony Routing and addressing
  - Input Information gathering
  - Alternative routing over IP

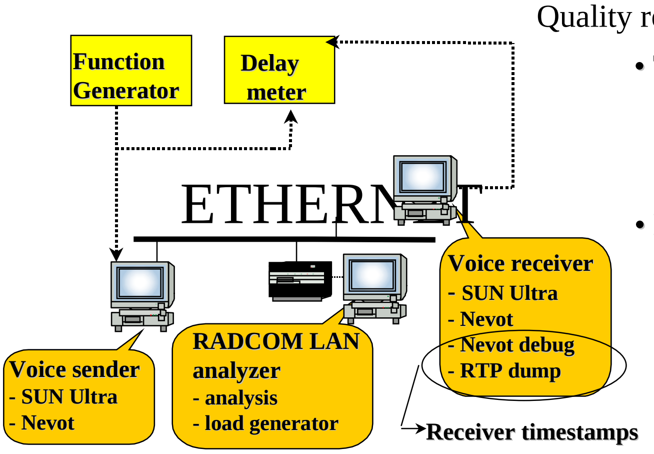
Phase 2 -->

- Service Management in the hybrid network

Phase 3

IP Voice in Ethernet - Delay is in the Workstation (IPANA -97)

Quality requires < 200 ms delay.



- **Terminal delay:**

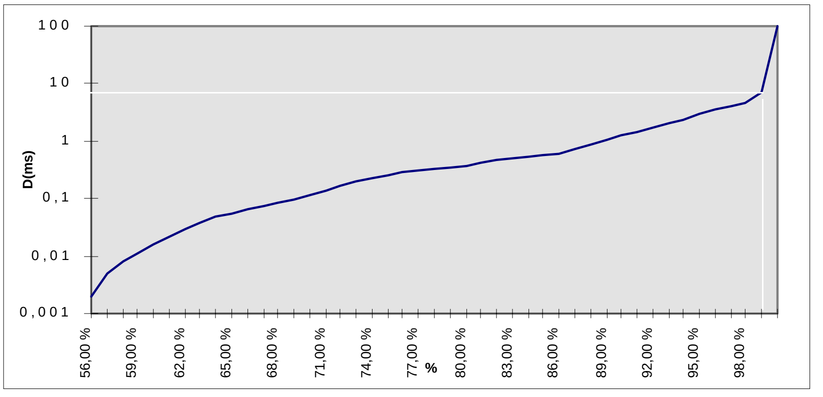
	Delay
HW	**8.9 ms**
VoIP Client	**103.9 ms**

- **End-to-end delay**

Packet lenght	**Delay**
0.02 s	**104.5 ms**

Difference = network delay

Packet spacing difference in a campus network



- In the public Internet lack of bandwidth, congested routes/links and underdeveloped charging are blockers to IP Voice.

Delay breakdown in a Nevot SunOS Workstation

- End to end delays of 30…40 ms in a campus intranet are achievable (IPANA -98).
- A buffering bug caused most of the 100ms in previous slide.
- Processing delay is 1 - 10% of CPU time depending on the coder.

Real time Services in IP

RTP (RFC 1889)

RTCP - “ -

Telephony over IP

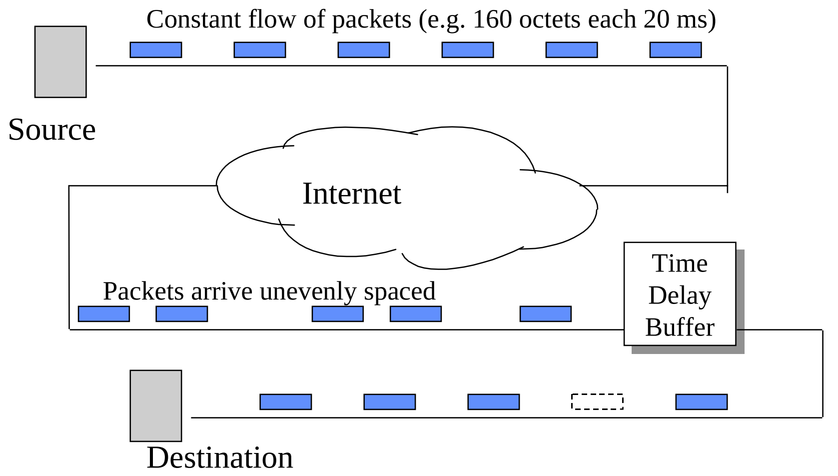
TCP is not suitable for real time services

Applications include

- Audio and video conferencing
- Shared workspaces
- Telephony
- Games
- Remote medicine
- ...

- TCP is point-to-point - not suitable for multicast
- TCP has retransmission for lost segments --> out of order delivery
- No mechanism for associating timing info with segments

Variable delay has to be compensated at reception by delay buffer



Packet arrival process is characterised by delay jitter and packet spacing difference

Delay jitter = Maximum variance in packet delay in a session

Example: fastest packet arrive in 1 ms
slowest arrive in 8 ms.
Delay jitter is 7 ms.

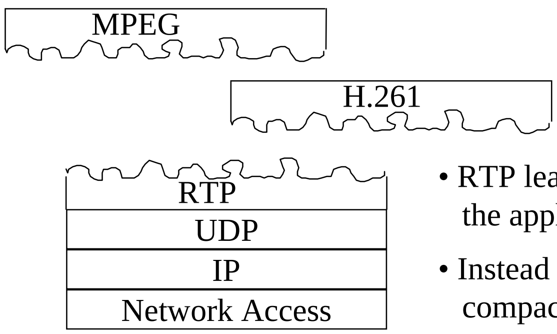
Packet spacing difference is measured based on receiver clock only:

Spacing difference = $[(t_i - t_{i-1}) - (t_j - t_{j-1})]$

Soft real time communications tolerate some loss but need the following

- Low jitter and Low latency
- Ability to integrate real-time and non-real-time services
- Adaptability to changing network and traffic conditions
- Performance for large networks and large nrof connections
- Modest buffering requirements in the network
- Effective capacity utilisation
- Low processing overhead per packet

RTP - Real time protocol is a "sublayer" library on top of UDP

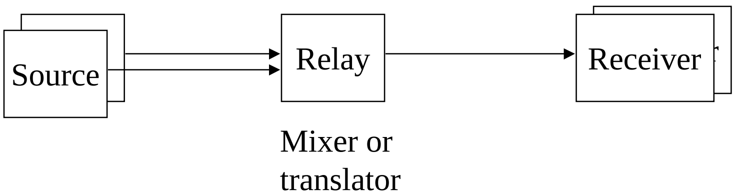


- RTP leaves recovery from loss to the application
- Instead of retransmission e.g. more compact coding may be chosen
- RTP provides sequencing

RTP supports the transfer of real time data among participants of a session

- Session is defined by
  - RTP port number (dest port in UDP header of all receivers
  - RTCP - Real time control protocol port number
  - Participant IP addresses - multicast address or a set of unicast addresses
- For session set-up e.g H.323 or SIP - Session Initiation Protocol can be used

RTP transport model includes sources, relays and receivers



- A mixer will combine sources - e.g. add voice signals from all conference participants
- A translator may translate from one video format to another
- The relay will mark itself as the synchronisation source

RTP header

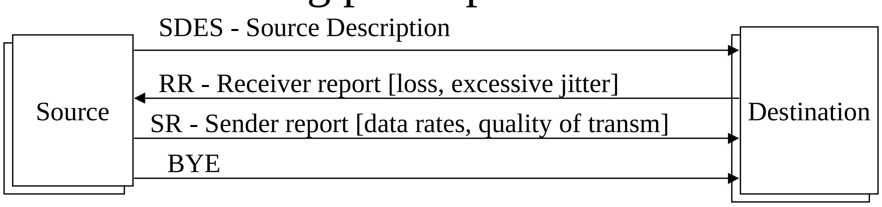
V=2 = version

0 4 8 9 16 31

V	P	X	CC	M	Payload type	Sequence number
Timestamp						
Synchronisation source (SSRC) identifier						
Contributing source (CSRC) identifier						
Contributing source (CSRC) identifier						

P - Padding - indicates that last octet of payload = nrof preceeding padding octets
X - Extension - there is an experimental extension header
CC - CSRC count - Nrof CSRC identifiers following the fixed header
M - Marker - e.g. End of video frame, Beginning of talk spurt
Payload type - format of RTP payload.
Seq. nr - each source starts at a random nr and =+1 for each packet - determines order of packets with the same timestamp
Timestamp - value of local clock at source at generation of first octet of payload
SSRC and CSRC identifiers are generated at random

RTCP - RTP Control Protocol provides feedback among participants of the session



- RTCP packets may be multicast in parallel to RTP using another UDP port
- RTCP source is identified by plain text
- Few participants: RTCP reports are sent once in 5s Rate of reports is reduced to max 5% of session traffic if there are more participants

RTCP fixed header is

0	4		8	16	31
V	P	RC/SC	PT	Length	
SSRC of sender (or CSRC)					

V =2 = version, P - Padding, same as RTP

RC - Reception report block count in SR or RR

SC - Source item count in SDES or BYE

PT - RTCP packet type [RR, SR, SDES, BYE]

Length - lenght of this packet in 32 bit words - 1

SSRC - same as in RTP

Sender Report carries sender info and reception report blocks

Sender information is

NTP timestamp (MS word)
NTP timestamp (LS word)
RTP timestamp
Sender’s packet count
Sender’s octet count

NTP is the wallclock time when sending this report (used for round-trip time measurement)

RTP timestamp let relate this report to RTP stream

Packet and octet counts run from beginning of session

Reception report block

SSRC_i (SSRC of source)	
Fraction lost	Cum nrof packets lost
Ext highest seq nr received	
Interarrival jitter	
Time of last sender report	
Delay since last sender report	

SSRC identifies source

Fraction lost since last SR or RR, Cum loss is for the whole session

16 LS bits= highest RTP seq nr. 16 MS bits=nrof times seq nr has wrapped back to zero

SR is sent by party who is both sender and receiver !

Average interarrival jitter for a source is estimated as follows

S(i) = Timestamp from RTP datapacket i
R(i) = Time of arrival of datapacket i in RTP timestamp units
D(i) = (R(i) - R(i -1)) - (S(i) - S(i -1))
J(i) = Estimate of Interarrival jitter up to the receipt of RTP packet i

$$J(i) = 15/16 * J(i-1) + 1/16 * | D(i) |$$

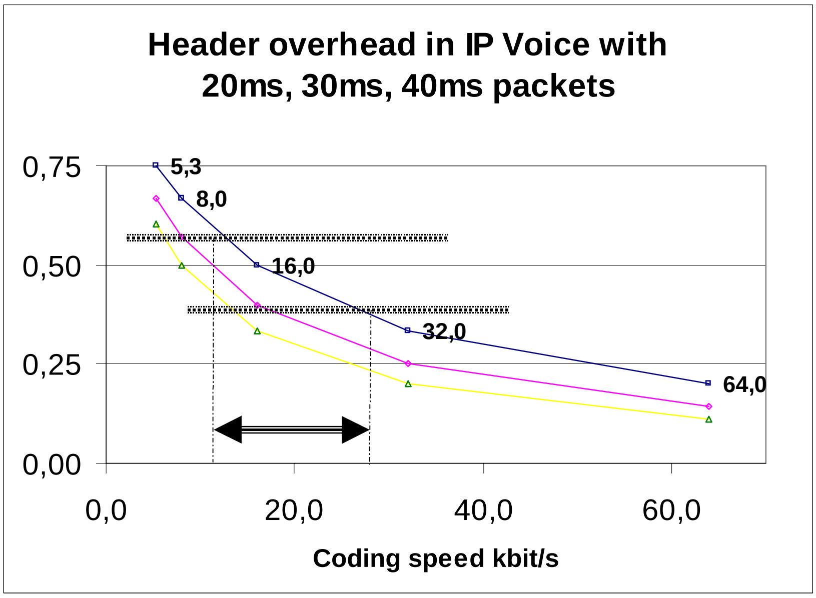
- Receivers use the estimate of Jitter to adjust the playout delay
- According to measurements the above exponential average is not always optimal

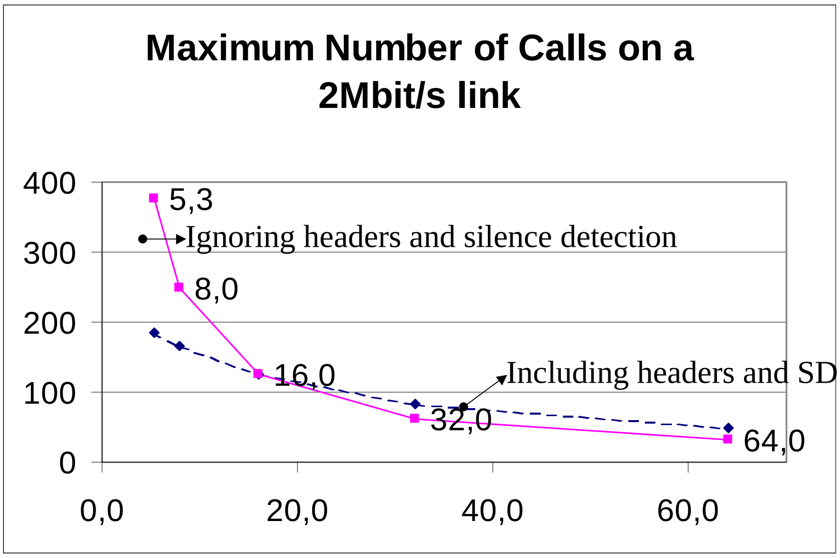
RTCP other packets

- RR are made of the fixed header + reception report blocks (see SR format lower part)
- SDES can carry
  - CNAME - Canonical Name
  - NAME - Real user name of the source
  - Email address of the source
  - Phone number of the source
  - TOOL - name of the tool used by the source

How to provide SCN-like QoS over IP?

- Integrated Services (use RSVP to make reservations in routers for each call!) changes Routers into SCN-Exchange -like systems. Does not scale well.
- DiffServ
  - mark voice packets with higher than BE priority at ingress
  - priority queuing in transit nodes
  - How to prevent voice from blocking BE traffic?
  - How to do Service Management?
  - Voice packets have high overhead - how to minimize?
- Overprovisioning




Maximum Number of Calls on a 2Mbit/s link
400
300
200
100
0
0,0
20,0
40,0
60,0
5,3
Ignoring headers and silence detection
8,0
16,0
Including headers and SD
32,0
64,0