IP Telephony

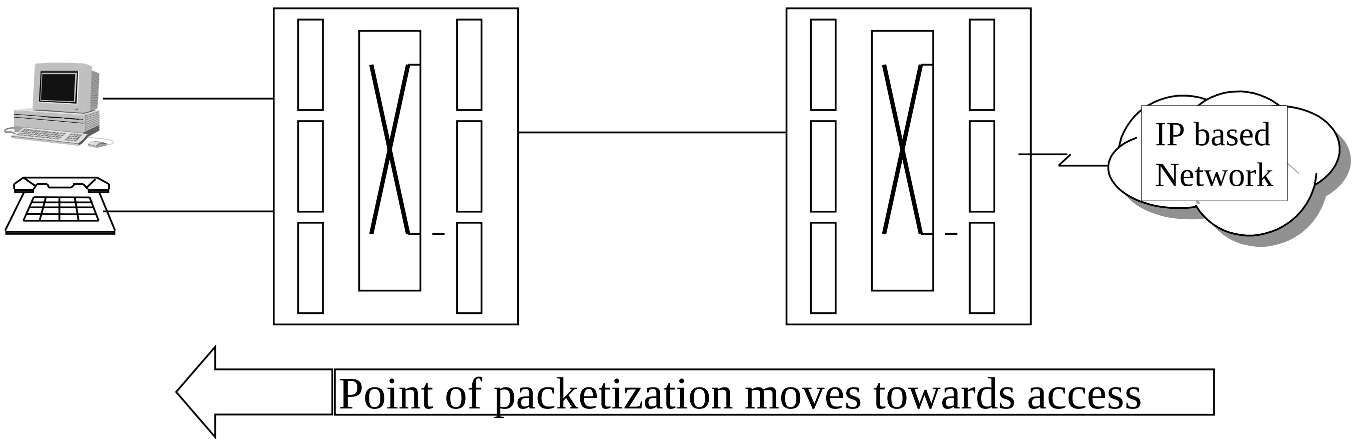
Overview of IP Telephony

Quality of Service

H.323

RTP, RTCP (not covered in class)

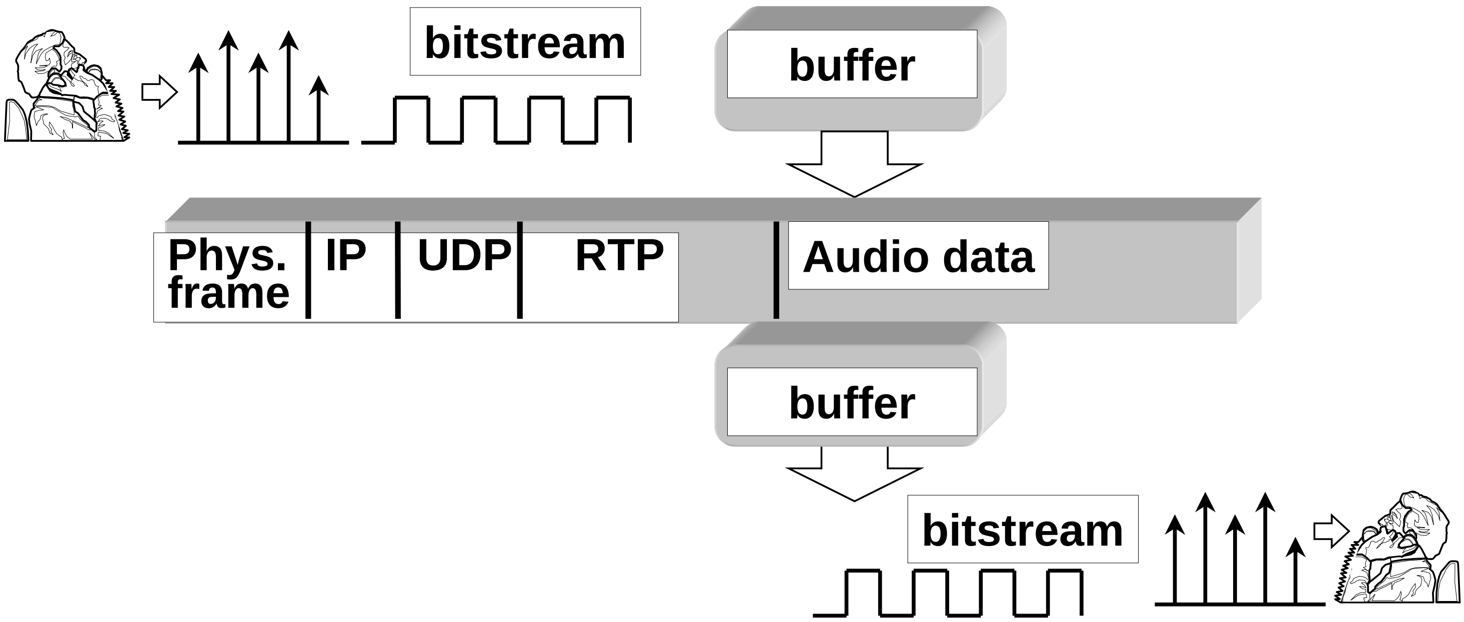
Data traffic will dominate voice in volume. Therefore Data will drive the Network Architecture.



- 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)

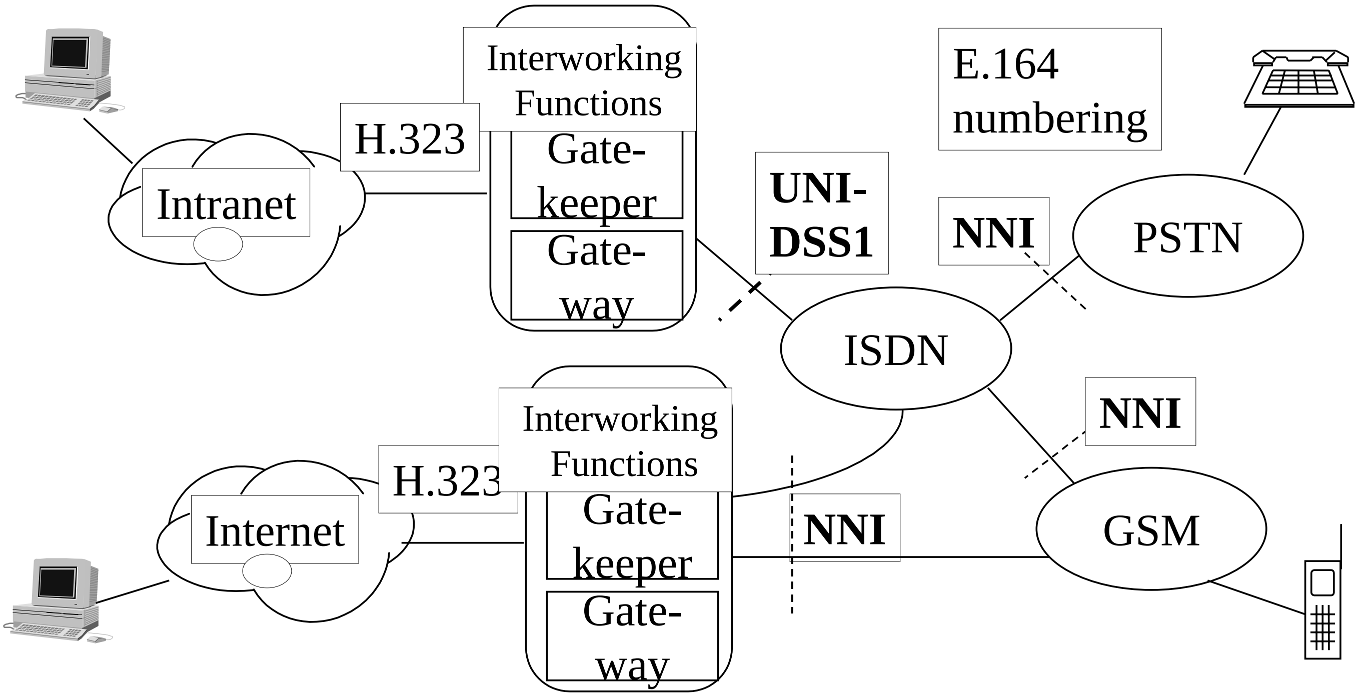


IP Telephony Standardization is active on de-jure and de-facto fora

- 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

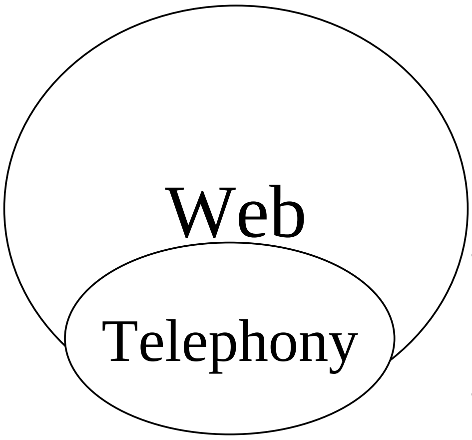
TIPHON specifies IP Voice to PSTN/ISDN/GSM Interworking



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 emerging
- Gateways and Gatekeepers/Call managers are available

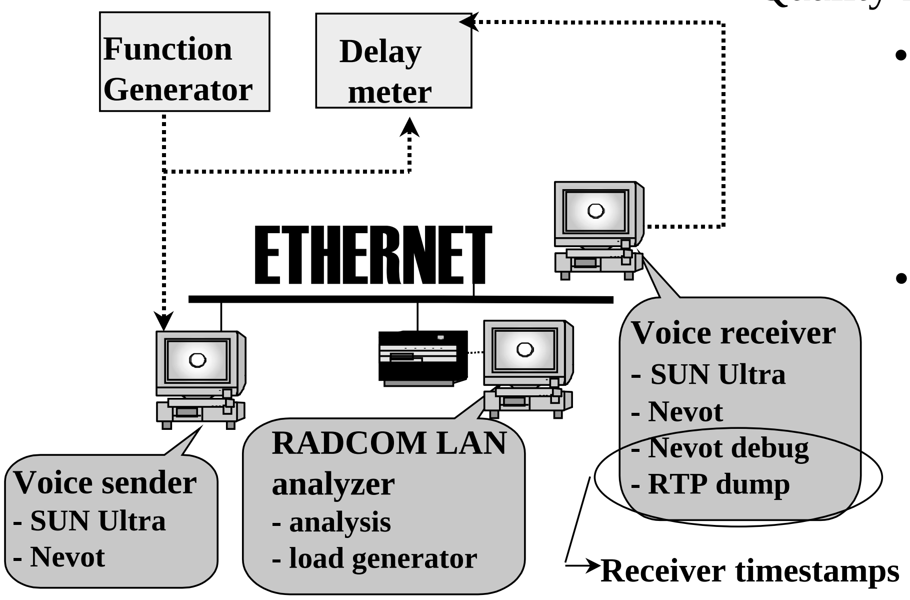
IETF alternatives to H.323 pursue Integration of Telephony to the Web



- 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.

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

Quality requires < 200 ms delay.



- **Terminal delay:**

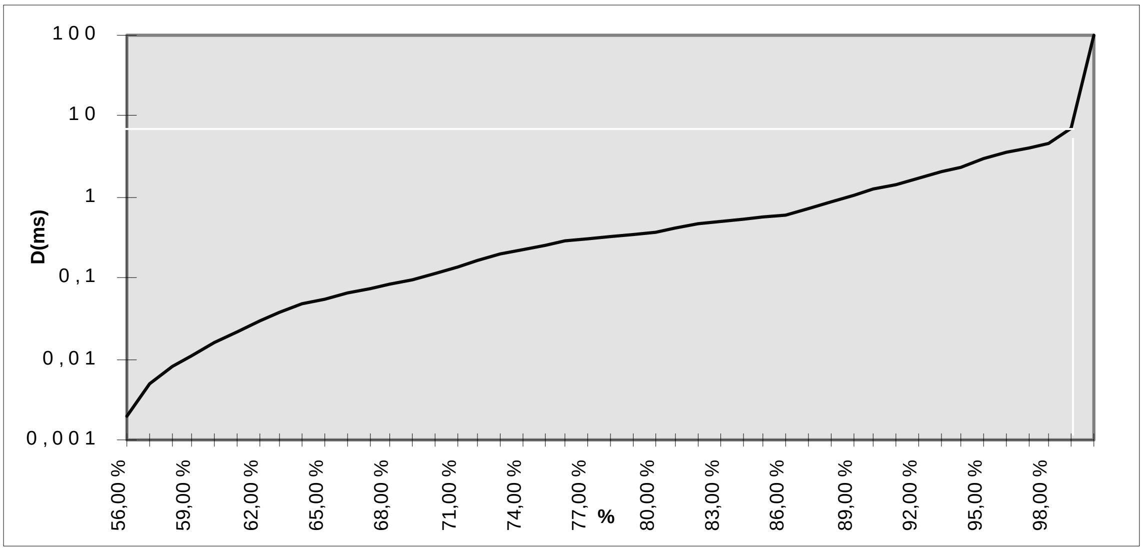
	Delay
HW	**8.9 ms**
VoIP Client	**103.9 ms**

- **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.

H.323 is the key standard for packet based multimedia communication

H.323 over: LANs, Enterprise Area Networks, MANs, Intranets, Internets

include dial-up connections and PP-connections over GSTN/ISDN with PPP packet transport.

Example networks:

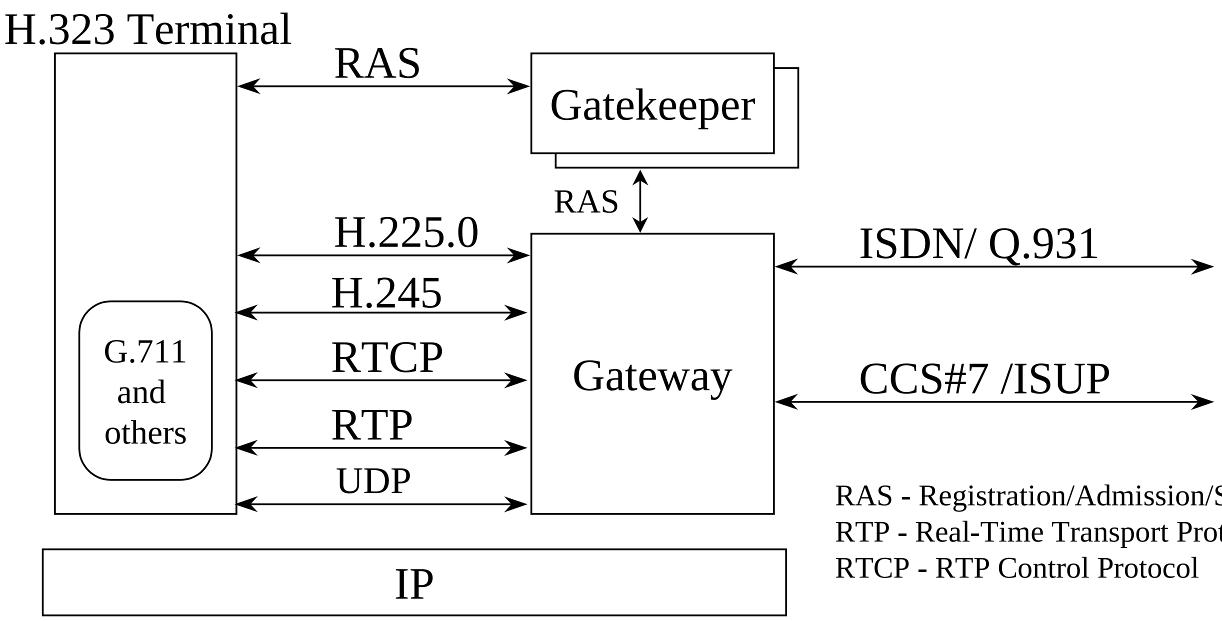
- Ethernet (IEEE 802.3)
- Fast Ethernet (IEEE 802.3u)
- FDDI
- Token ring (IEEE 802.5)
- ATM

MM includes:

- Audio (mandatory)
- Video (opt)
- Data (opt)

Communication = conference or two party call.

H.323 uses H.225.0, H.245 and RTP



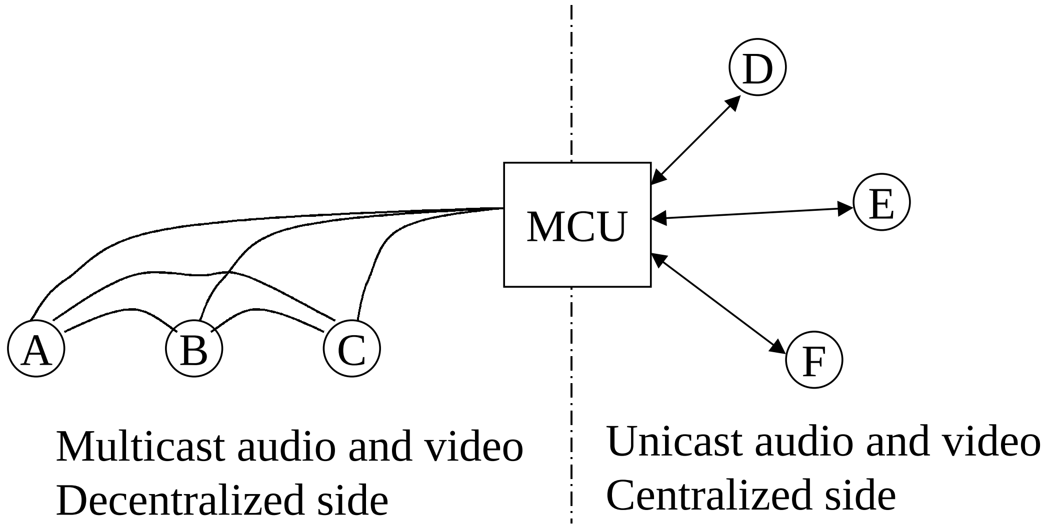
RAS - Registration/Admission/Status
RTP - Real-Time Transport Protocol
RTCP - RTP Control Protocol

Note: this is an example configuration!

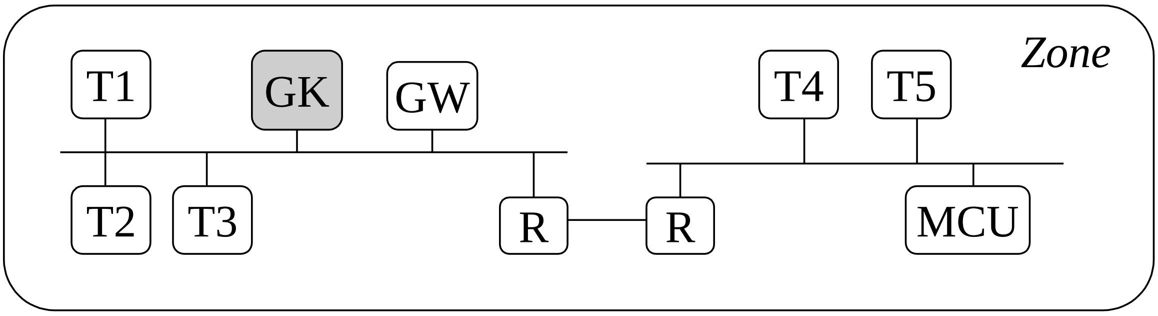
H.323 supports many call modes

- Directly between two H.323 endpoints (no GK)
- Between two H.323 endpoints using a GK
- Many conference types
  - ad hoc multipoint conference (start with 2-party call - expand to conf)
  - broadcast conference (one sender, many receivers)
  - broadcast panel conference (mp conf + bc conf)
  - centralized multipoint conference (trms pp to MCU, MP sends to trms)
  - decentralized multipoint conference (no MCU - all to all coms)
  - hybrid multipoint conference - centralized audio or video
  - mixed multipoint conference (mix of decentralized + centralized modes)

Mixed multipoint conference example

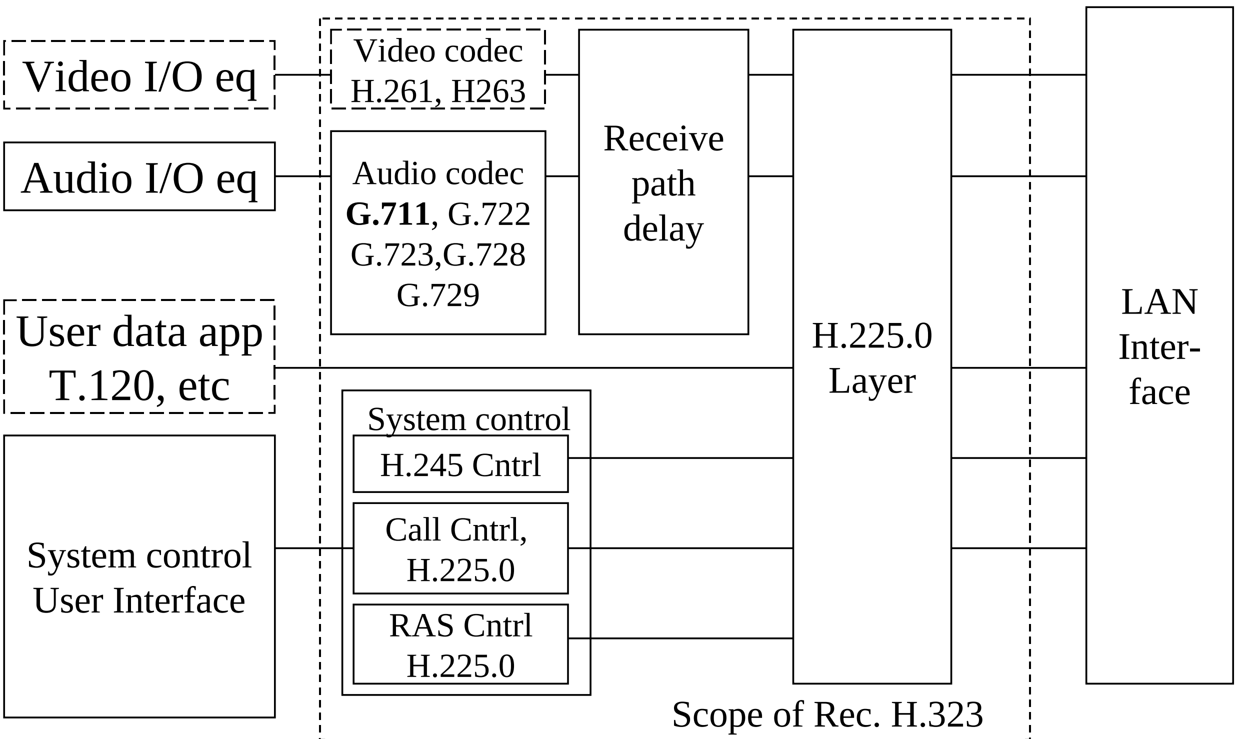


H.323 zone is controlled by a Gatekeeper



- Zone has at least one terminal, MCUs and GWs are optional.
- Zone has one and only one GK.
- Gatekeeper controls access to the network for Ts, GWs and MCUs and provides
  - address translation
  - gateway location
  - bandwidth management

H.323 terminal equipment



H.323 supports many parallel addressing methods

- H.323 entity shall have at least one Network Address (e.g. IP address)
- TSAP identifiers allow multiplexing several channels sharing one Network Address - map to port numbers
- An endpoint may have one or many Alias addresses - may represent the Ep or a Conference that the Ep is hosting. Include: E.164 numbers, H.323 IDs (e.g. John Smith), e-mail addresses. Aliases are unique in a zone.

Gateway translates between transmission formats, communication procedures and media formats

- Example: H.225.0 to and from H.221 (transm.f)
- H.245 to and from H.242 (comm procedure)
- Media format: Audio, video, data
- Represents characteristics of network endpoint to GSTN endpoint and the reverse. May also work as an MCU
- Can also do call set-up and clearing

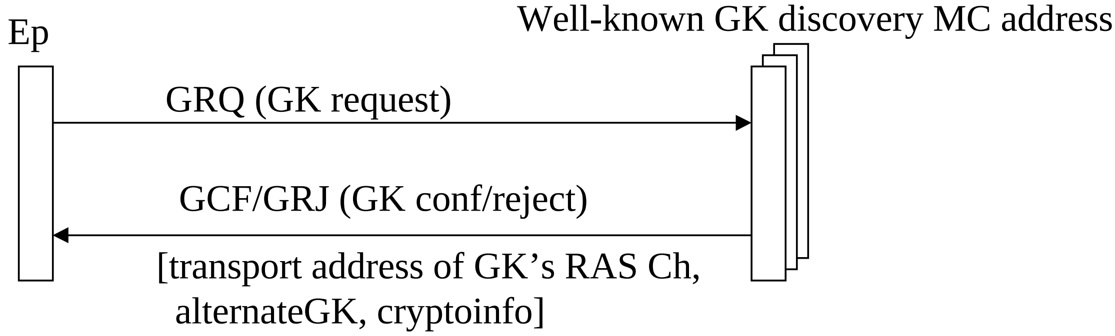
GK provides call control services, when present, shall do:

- Address translation (e.g. alias to transport address using DNS + E.164 to transport address)
  - uses the translation table produced from registration messages
- Admission control: ARQ/ACF/ARJ of H.225.0
  - based on call authorization, bandwidth, other criteria
- Zone management

GK may optionally do

- Call control signalling. May also direct the endpoints to setup call signalling channel between themselves
- Call Authorization using H.225.0 signalling
- Bwidth management controls the number of simultaneous calls in the zone
- Call management - keep list of calls -> busy conditions
- GK management, Directory service etc - FFStudy

Endpoint can discover a Gatekeeper automatically

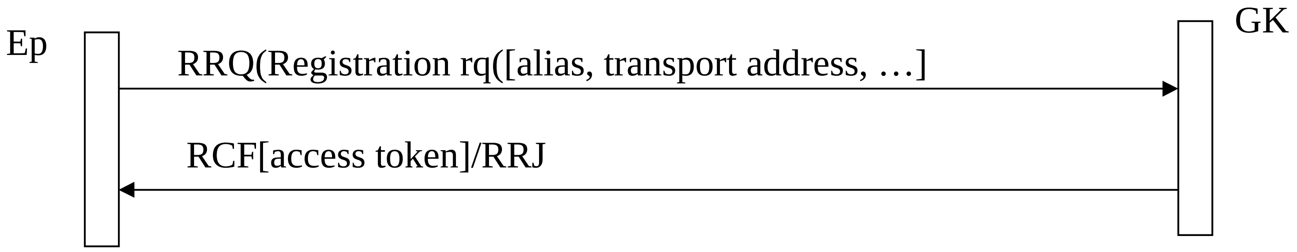


- Automatic discovery eases maintenance of individual terminals
- Terminals may also have the GK id configured

RAS signalling function

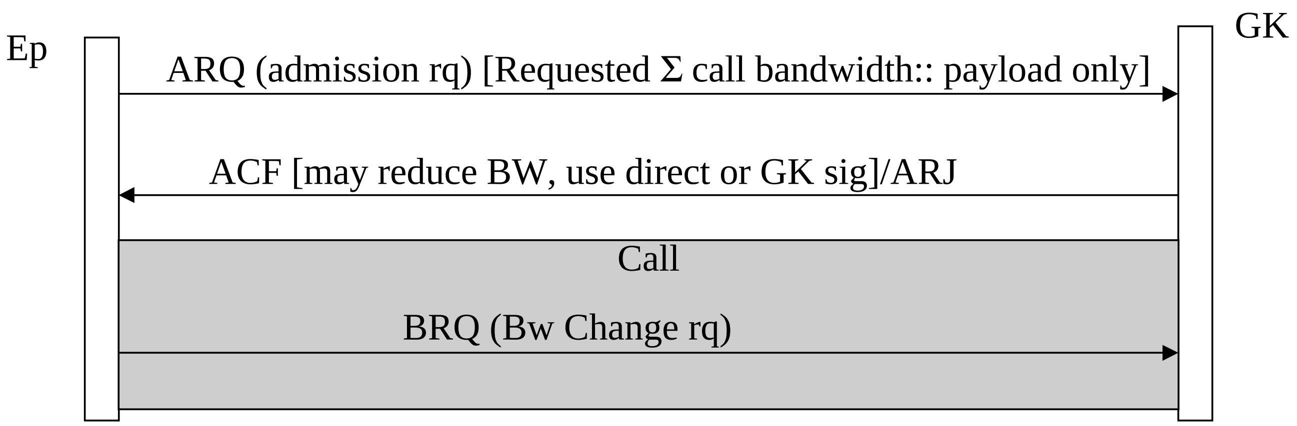
- Performs
  - Registration of endpoints, Admission of calls, Bandwidth changes for calls
  - Status
  - Disengage of endpoints.
- Uses RAS signalling channel =/= call signalling channel and H.245 control channel. GKs have a well def. TSAP id for RAS sig. channel
- Endpoint=H.323 terminal or GW or MCU (is callable)

Endpoints register using GK’s RAS Channel Transport Address prior to any calls are made



- Security policy may require that registration has time-to-live and has to be repeated from time to time
- Endpoint or GK may unregister using the URQ message
- The GK maintains an alias to Network Address translation table.
- Access token may be used later in call setup

Call Admission sets the upper limit for the aggregate bitrate of the call



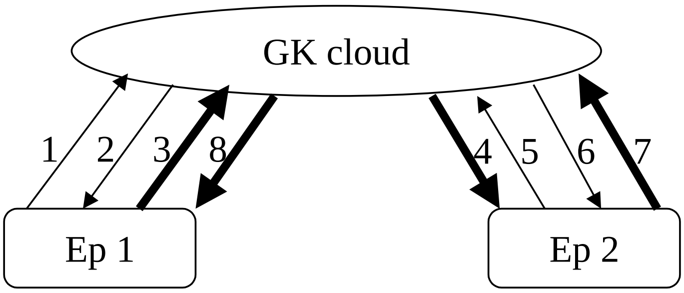
Call signalling uses H.225.0

- Call signalling= call setup, request changes in Bw of a call, get status of Ep, disconnect call
- Call signalling is largely inherited from ISDN
- Call Signalling Channel is opened prior to H.245 procedures and prior to any other logical channels between endpoints. Eps have a well known TSAP id for the Call Sig. Channel and a well-known Discovery Multicast address.

H.323 Call Signalling Channel Routing

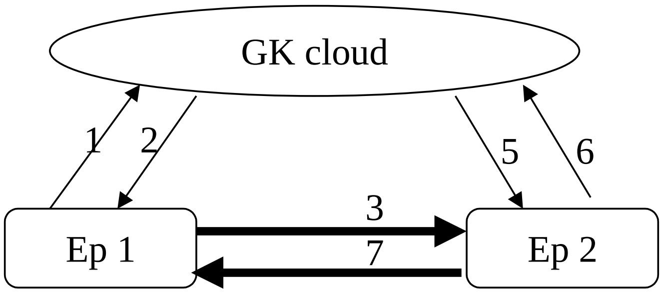
- ACF has the Transport Address of the Call Signalling Channel
- The address is either a GK address or an Endpoint address.

Gatekeeper Routed Call Signalling



RAS Signalling Channel msgs
Call Signalling Channel msgs

Direct Endpoint Call Signalling

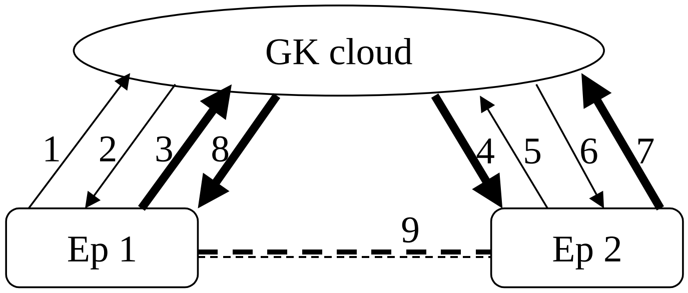


1,5 - ARQ 3,4 - Setup
2,6 - ACF 7,8 - Connect

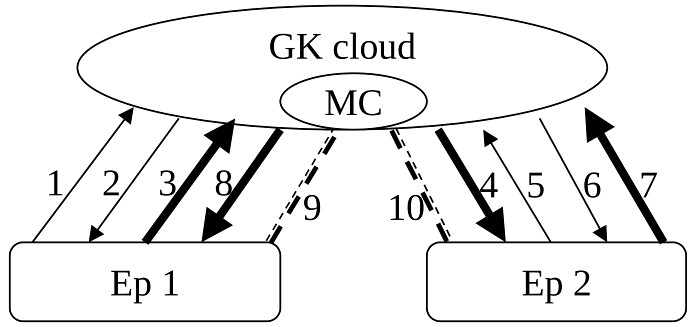
H.245 Control Channel Routing

The goal of call signalling is the setup/release of H.245 Control Channel!

Direct H.245 Control Channel



GK routed H.245 Control



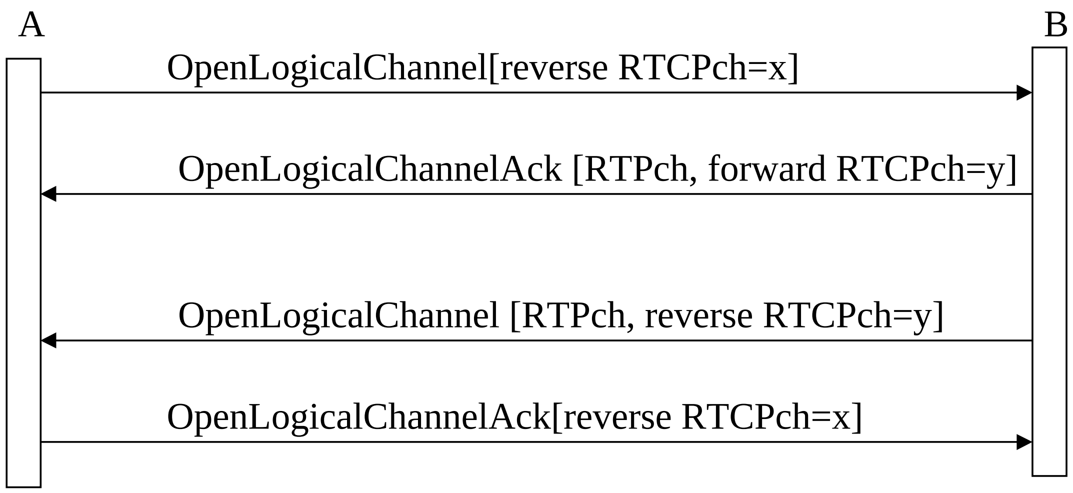
RAS Signalling Channel msgs
Call Signalling Channel msgs
H.245 Control Channel

1,5 - ARQ 3,4 - Setup
2,6 - ACF 7,8 - Connect
9,10 - H.245 Channel

H.245 carries end-to-end control messages between H.323 entities

- Master/slave determination for conflict resolution
- Capability Exchange (e.g. what codecs are supported)
- Logical Channel Signalling (bounds media type, algorithm etc. to ports)
- Bidirectional Logical Channel Signalling
- Close Logical Channel Signalling
- Mode Request (conference modes)
- Round Trip Delay Determination
- Maintenance Loop Signalling
- H.323 also uses flowControlCommand of H.245 to limit bandwidth

Sample H.245 Logical Ch Signalling for two way RTP+RTCP communications setup

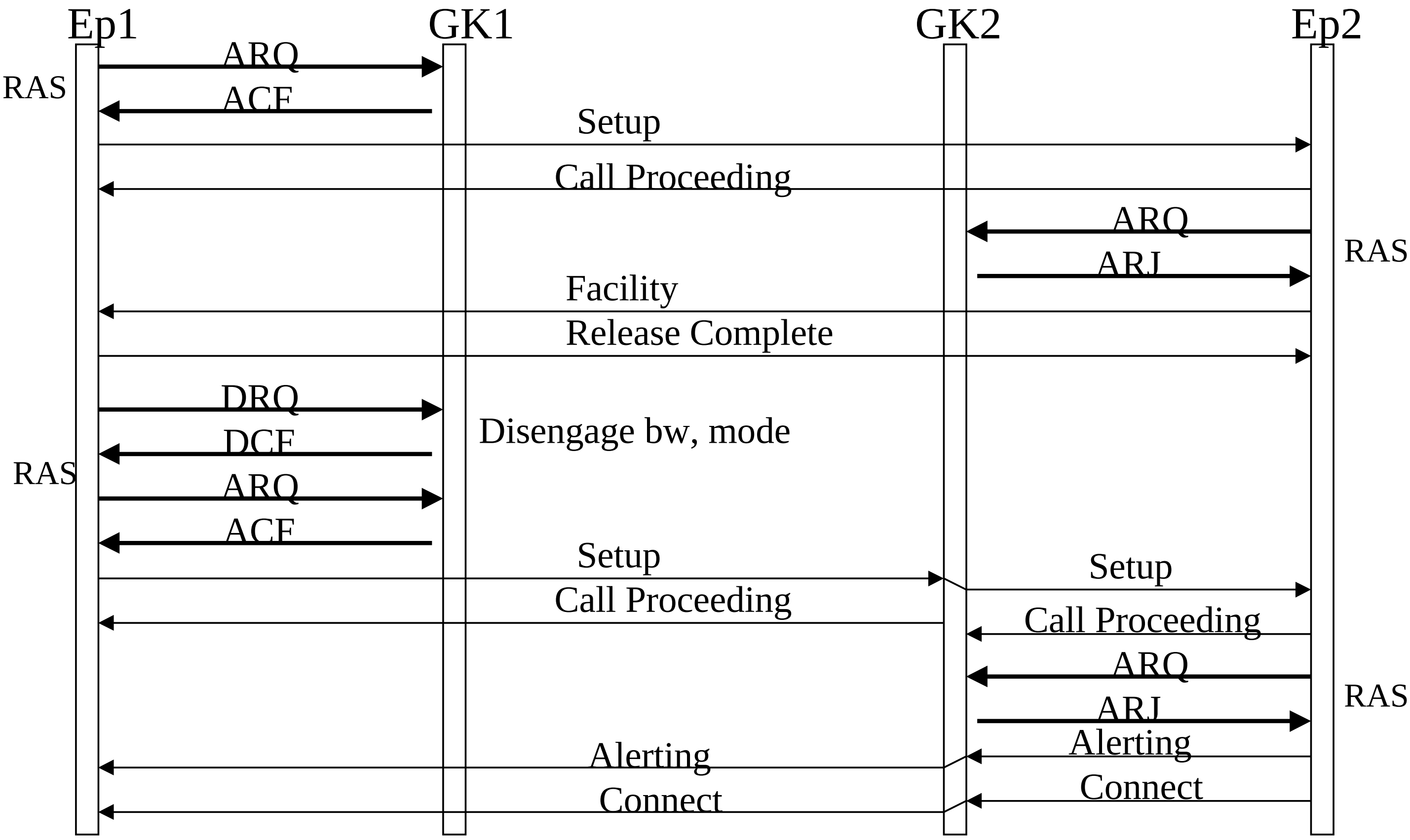


- In IP networks a logical channel corresponds to an IP port number
- Uses H.245 Control Channel

H.323 Call identification uses

- *Call reference value* - between two H.323 entities on a signalling channel (one for call signalling and another for RAS channel
- Call ID - a globally unique non-zero value created by the calling endpoint passed in all H.225 msges
- Conference ID (CID) - in all sub-calls of a conference

Both endpoints registered - Direct/GK-routed call signalling



H.323 summary

- H.323 inherits call signalling from ISDN
- H.323 has many conference modes and many signalling and call routing options
- Call setup delay is reduced by using the Fast Connect Procedure: packs all setup info from both H.225.0 and H.245 into fastStart element in *setup* and *connect* (call proceeding, alerting) messages

Real time Services in IP

RTP

RTCP

Telephony over IP

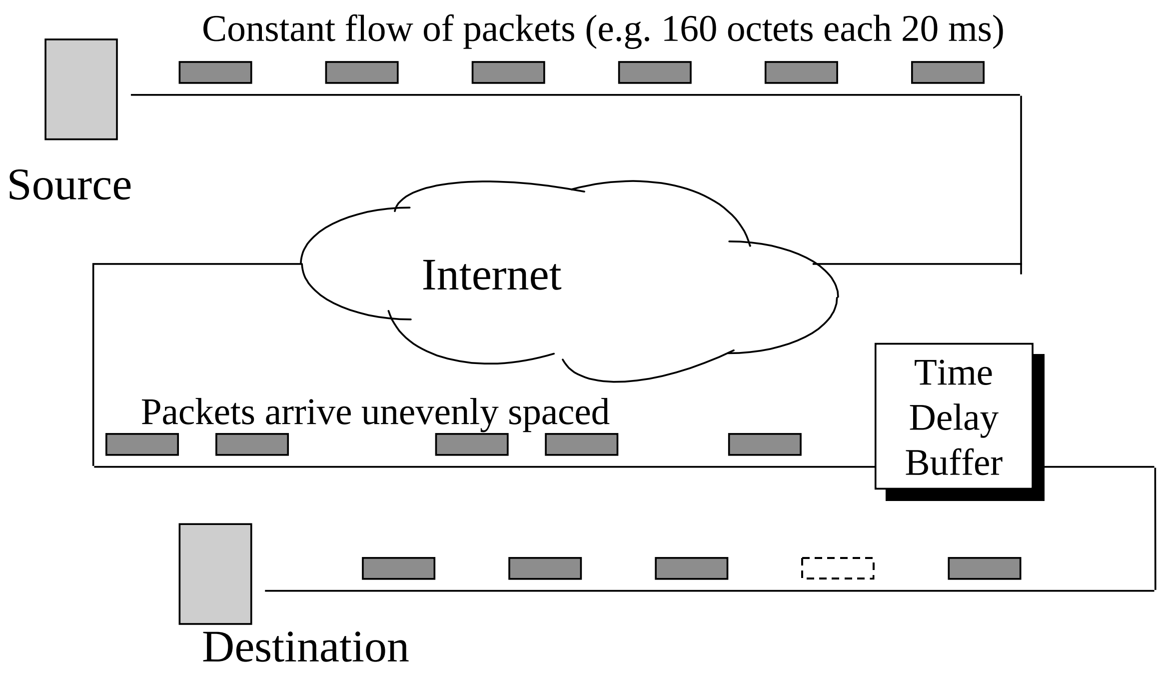
TCP is not suitable for real time services

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.

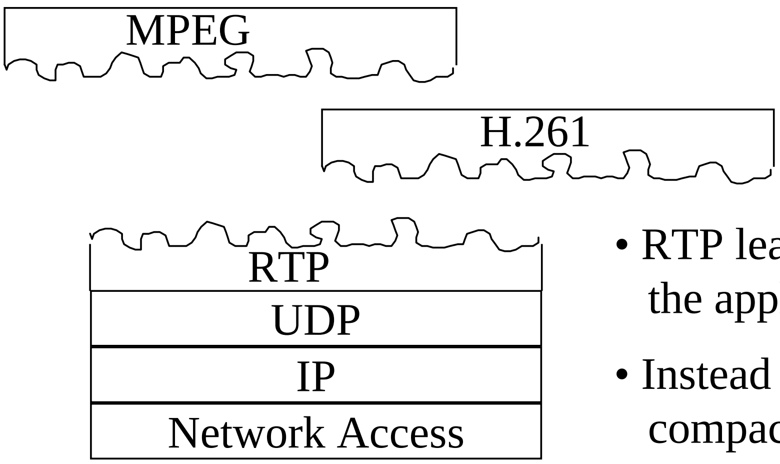
Packet spacing difference is measured based on receiver clock only:

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

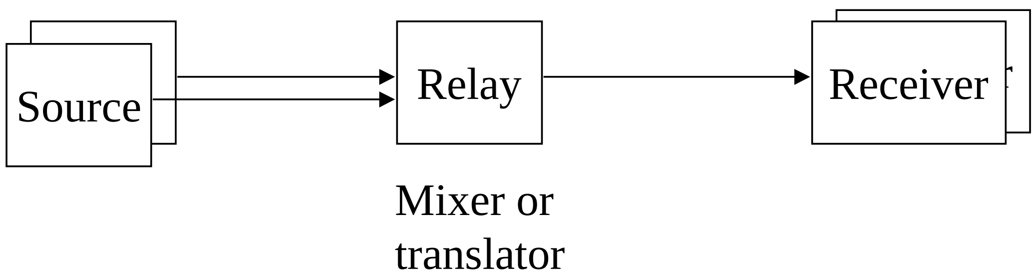


- RTP leaves recovery from loss to the application
- Instead of retransmission e.g. more compact coding may be chosen
- RTP provides sequencing

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

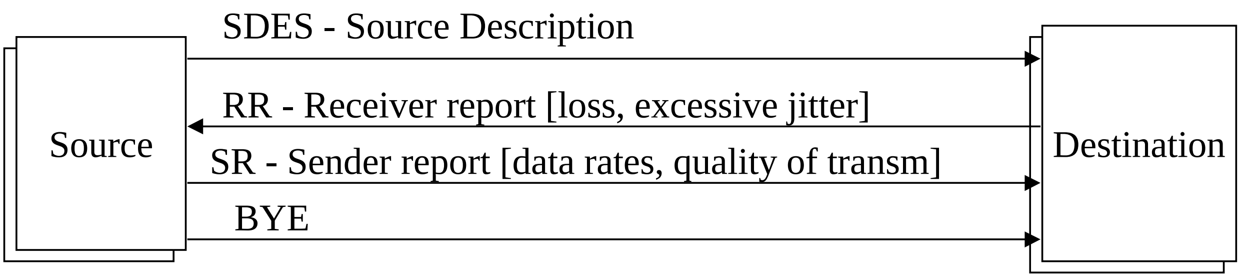
RTP header

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

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
- 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