QoS for Interactive Multimedia Over IP

Project Presentation By:-

Harsha K Rajasimha (hrajasim@vt.edu)
Fahad Koujah (fkoujah@vt.edu)
Jinggang (Gene) Wang (jiwang5@vt.edu)

Agenda

Introduction

- **In Multimedia over IP, all Information Streams are classified into 5 classes:**
- **audio, video, data, communication control and call control**
- **Audio signals contain digitized and coded speech.**
  -May use voice activation to reduce the bit rate.
  -accompanied by an audio control signal.
- **Video signals contain digitized and coded motion video.**
  -transmitted at a rate no greater than that selected as a result of the capability exchange.
  -accompanied by a video control signal.
- **Data signals include still pictures, facsimile, documents, computer files and other data streams.**

Introduction (Contd…)

- **Communications control signals pass control data between transmitter and receiver for capability exchange, opening and closing logical channels, mode control and other functions that are part of communications control.**
- **Call control signals are used for call establishment, disconnect and other call control functions.**

Why QoS?

Why do we need QoS for interactive multimedia?

- **Early years of internet were dominated by email, file transfer and chatting**
- **But with the advent of interactive real-time applications such as**
  - **voice over IP**
  - **Videoconferencing**
  - **distance learning programs, etc,**

  in the recent years, the QoS requirements are posing a challenge
- **Even though poor service can be tolerated in some cases (where low price is charged), the main focus of this work is on a high Quality of Service**

Motivations

- **Research studies have concluded that the maximum tolerable delay for natural hearing is 100 ms**
  - **This implies that we should achieve a maximum end-to-end delay (between sender and receiver) of not more than 100 ms**
- **The picture (both audio and video) is played continuously at the receiver end at the same rate at which it was produced by the sender**
  - **This requires that the end-to-end delay should be a constant**
- **Further, we should achieve all these, independent of the network instant load and the connection rate**

QoS Architecture

The QoS architecture distinguishes 3 levels – Client Level (C-QoS)
– Session Level (S-QoS)
– Transport Level (T-QoS)

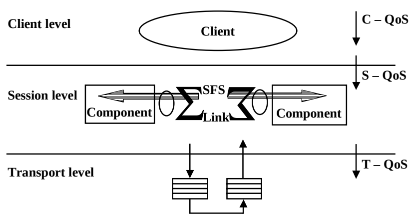


Fig. 1 QoS Architecture [1]

QoS Architecture (Contd…)

- **C – QoS specifies the QoS from the end-user perspective. E.g. A particular user group may perceive quality as good, bad or average depending on the application.**
- **S – QoS specifies a number of generic parameters such as a stream's type, delay, jitter (delay variation), loss rate , etc. These allow to express other media specific characteristics such as frame rate, picture size, color depth, etc. S – QoS is specified at the input ports of the sink components. Each sink component can request for a separate S – QoS.**
- **SFS – Session flow specification contains media specific and generic parameters which will be used during the QoS negotiation.**
- **T – QoS specified at the transport level is media independent. It specifies various parameters like packet rate, packet size, end-to-end delay, and jitter. This part of the architecture is still open for research and forms the next section of discussion. [1]**

QoS Issues

- **As discussed in the previous section, QoS can be achieved in 3 different levels. But here, we mainly concentrate on the Transport level QoS.**
- **Lot of research is going on in this area and a lot of different algorithms have been suggested to improve the quality of multimedia w.r.t. different parameters.**
- **Here we discuss end-to-end delay and jitter which we consider as the most important parameters for our study.**

Model Of Videoconferencing

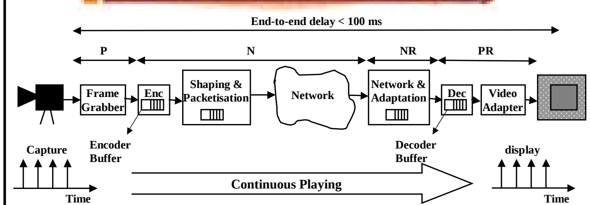


Fig. 2 Model of a Videoconferencing System [2]

P – Processing delay
PR – processing resynchronization delay
N – Network delay
NR – Network resynchronization delay

End-to-end Delay and Jitter

- **As discussed earlier, maximum tolerable end-to-end delay (including propagation delay) for natural hearing is 100 ms. Hence, the maximum delay for picture display should also be 100 ms. i.e., P + N + PR + NR + *Pr* = Constant <= 100ms [2]**
- **The picture should be displayed and the audio samples should be played by the receiver at the same rate at which they have been sent by the sender.**
- **To achieve these QoS requirements, VBR MPEG (Moving Picture Experts Group) encoding should be used with Packet switched networks as it has been shown that using raw video or CBR MPEG encoding delivers longer delays and jitter. [2]**
- **VBR encoding is impractical in the case of circuit switched networks as the network utilization is low. [2]**

VBR MPEG

- **Why VBR MPEG over packet switched networks?**
- **Only a small buffer is needed at the output of the encoder for assembling data units. The rate of the resulting compressed stream is highly variable.**
- **VBR encoding provides high compression and contributes less to end-to-end delay due to the processing resynchronization component shown in fig. 2**
- **Various studies have shown that VBR MPEG with Packet switching yield the best quality of service.**

Resource Reservation and Traffic Shaping

- Resource reservation is used in the network in order to bound the queuing delay (i.e., to reduce N and NR).
- Resource reservation is more efficient if traffic shaping is done at the network boundaries, even though this introduces a network shaping delay (i.e., increases N and NR).
- Resource reservation is based on the traffic description given in terms of burstiness and average rate.
- The network guarantees quality of service only if the actual traffic is compliant with that given at the time of resource reservation.

Internet Videoconferencing System

- Brief overview on H.261
- Error control scheme for videoconferencing
- Congestion control scheme for videoconferencing
- Summary

Videoconferencing on the Internet

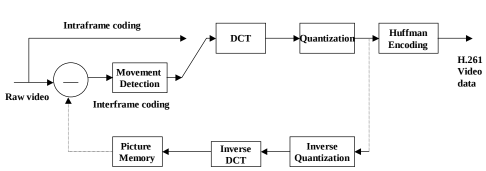
- IP multicast
- Video conferencing tools
  - IVS (INRIA video conferencing tool)
  - NV (network video)
  - VIC

Contd…

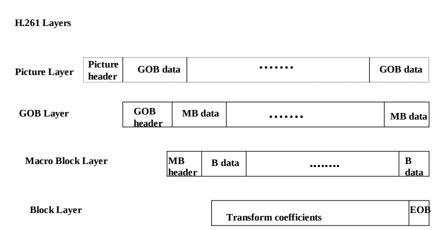
- IVS : INRIA Videoconferencing system
- Uses a modified version of H.261 compression standard for moving images.
- IVS interoperate with a large number of H.261-based commercial video codecs.
- It implements error and congestion control schemes

H.261 brief overview

Basic H.261 coding loop

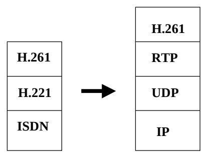


H.261 Hierarchy of groupings



Contd…

- H.261 over ISDN versus H.261 over IP



Error Control scheme

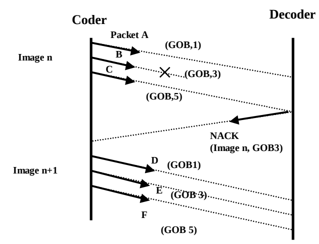
- On the Internet, most packet loss is due to congestion control rather than transmission errors.
- Real-time applications require a different form of error correction than errors in normal data stream.
- Delayed video packets are considered lost if the delay exceeds a maximum delay value.
- UDP provides no mechanisms to know if the receivers received the packet successfully.

Error control scheme (Contd…)

- Ways to reduce packet loss
  - Request a complete refreshment of the image using INTRA-frame coding.
  - Replenish in INTRA-frame mode only the MB's concerned by the loss
  - Periodically refresh the image in INTRA encoding mode
    - to control the accumulation of inverse transform mismatch errors

Error control scheme

Data and NACK packets

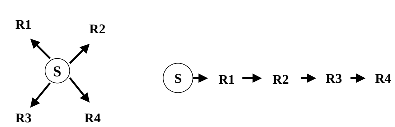


Error control scheme (Contd…)

- IVS implements a combination of :
- Replenishing in INTRA-frame mode only the MB's concerned by the loss
- Periodically refresh the image in INTRA-frame encoding mode
  - When the number of receivers is < 10 NACK's are used
  - Else, the INTRA refreshment rate is adapted to the network congestion state.
  - For an average packet loss observed by the receivers (p)
  - for R receivers, and
  - L = (avg length of NACK packet / avg length of data packet)
  - The proportion of bandwidth used by the feedback channel is within an interval [L* p, L*p*R] depending on the network topology

Error control scheme (Contd…)

- For example, for 10 receivers, with an average loss rate of 20%, average data packet 500 bytes and average NACK packet length is 12 bytes
- The proportion of bandwidth used by the feedback messages is between 0.48 % for a star topology and 4.8% for a chain topology.



Congestion control scheme

- In order to control End-to-End delay, two components are needed
  1. Network sensor
  2. Throughput controller

Congestion control

- Network sensor:
  - Uses feedback information based on measured packet losses.
    - Receivers periodically send a QoS measure – packet loss rate observed by the receivers – back to the sender
    - The source converts the different measures of QoS into a global measure – e.g. by taking the median of the loss rate.
- Throughput controller:
  - Control actions are taken by the coder at discrete points in time e.g. whenever a sequence of 100 packets has been encoded.
  - An algorithm adjusts the maximum output rate of the coder maximum rate so that the median loss rate stays below a tolerable loss rate.
  - Receivers who can not obtain sufficient resources – through reservation – will be asked to leave the conference.

Congestion control (Contd…)

- The congestion control algorithm is as follows:
  - If (median_loss_rate > tolerable_loss_rate)
    - Maximum_rate = max (maximum_rate/2, minimum_rate)
  - Else
    - Maximum_rate = gain * maximum_rate
- The value of the maximum_rate at the source decreases as losses are detected at the receiver
- The output of the source adapts its output rate according to network conditions

Congestion control (Contd…)

- Output rate control
  - By modifying the frame rate of the application.
  - By adjusting the value of the quantizer.
    - Loose quantization factor for the coefficients, reduces the precision of the image.
  - By raising the movement detection threshold, reduces the number of blocks encoded for each image.

Summary

- IVS is a video conferencing tool that implements
  - H.261 standard
  - Error correction scheme
  - Congestion control scheme.

Need a Standard !

- In Multimedia over IP market, multiple vendors develop different products. We need a standard to make them interoperable.
- One standard is ITU-T H.323 specified by ITU Study Group 16. Another is Session Initiation Protocol(SIP) from IETF Multiparty Multimedia Session Group.
- IP Telephony with H.323 truly means Multimedia over IP.

What's H.323?

- Describes terminals and other entities that provide multimedia communications services over Packet Based Networks (PBN) which may not provide a guaranteed Quality of Service.
- Four Versions: V1, 1996; V2, 1998; V3, 1999; V4, 2000.
- Many equipment manufacturers, software vendors, and service providers have already built products and services supporting H.323.
- A number of organizations, such as ETSI's Project TIPHON, IMTC, and IETF have contributed to H.323's success

Elements of an H.323 System

Elements include:

- Terminals
- Multipoint Control Units (MCUs)
- Gateways
- Gatekeeper

Referred to as "endpoints"

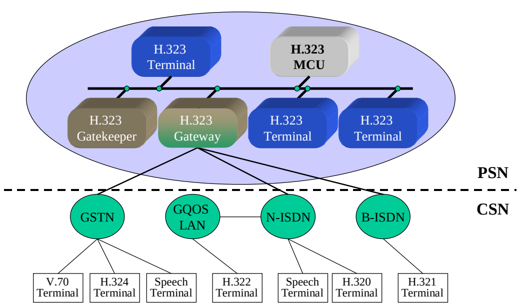
A Zone of an H.323 system

-A collection of all terminals, gateways, and MCUs managed by a single gatekeeper(no more than one).

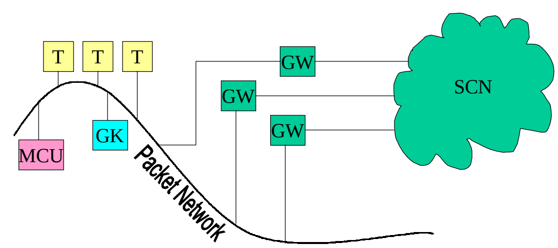
-The simplest zone just include one terminal. All other elements are optional.

-Independent of network topology.

H.323 Network Elements

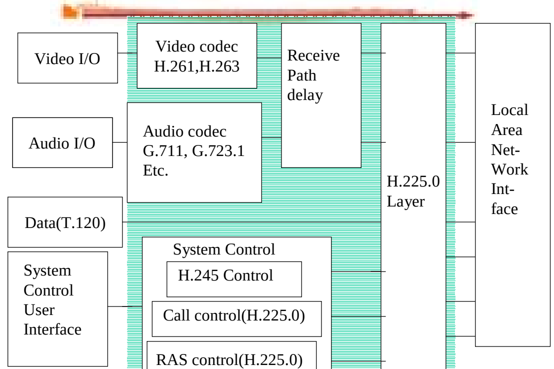


The Zone



T:Terminal; GK:Gate Keeper; SCN:Switched Circuit Network; GW:Gate Way; MCU:Multipoint Control Unit.

H.323 Terminal Protocol stack



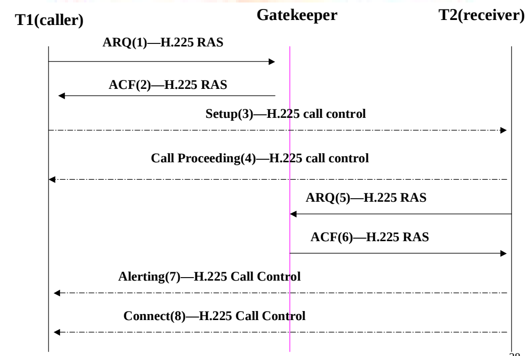
H.323 Terminal Protocol stack (Contd…)

- All H.323 terminals must have a System Control Unit, H.225.0 layer, Network Interface and an Audio Codec Unit. The Video Codec Unit and User Data Applications are optional.
- Audio Codec: must support G.711(coding at 64kbps). G.723.1(5.3kbps and 6.3kbps) predominant in low-bit-rate.
- Video Codec: must support H.261.But H.263 will be used more often due to video quality improvement at the same low-bit rate(10-20kbps).
- H.225.0 Layer (H.225.0) formats the transmitted video, audio, data and control streams into messages for output to the network interface and retrieves the received messages.
- Real-Time Transport Protocol(RTP) and RTCP deliver end to end service of real time audio and video.

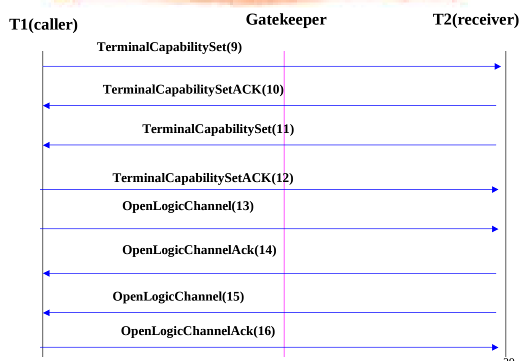
Other Elements

- The MCU contains a Multipoint Controller (MC) which is responsible for managing multipoint conferences (two or more endpoints engaged in a conference)
- Gateways interface H.323 to other networks, including the PSTN, H.320 systems, other H.323 networks (proxy), etc.
- The gatekeeper is used for admission control and address resolution.
  -may allow calls to be placed directly between endpoints or
  -may route the call signaling through itself to perform functions such as follow-me/find-me, forward on busy, etc.

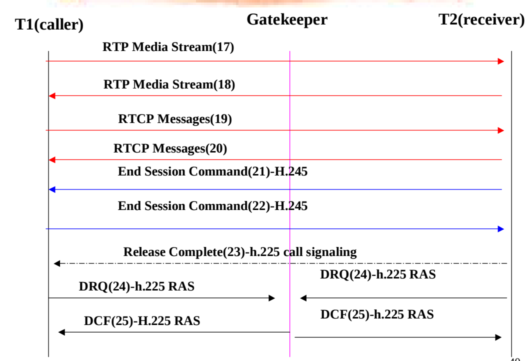
Direct Call(Call Establishment)



Direct Call (H.245 Control Signal flows)



Direct Call (Media Flow and Call Release)



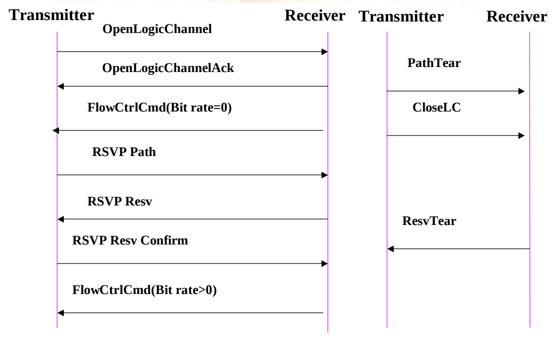
Quality of Service(QoS) Issues

- H.323 performance is limited by the quality of service offered by the network.
  - Quality of media is corrupted (delay, jitter, packet loss) on unmanaged networks.
  - 1 frame per packet (FPP) to minimize latency. But need N FPP to minimize bandwidth.
  - Intel Result: 2-3 FPP Minimizing latency at the expense of some moderate bandwidth. SIP does not negotiate FPP
  - Lack of good media quality is hindering market acceptance

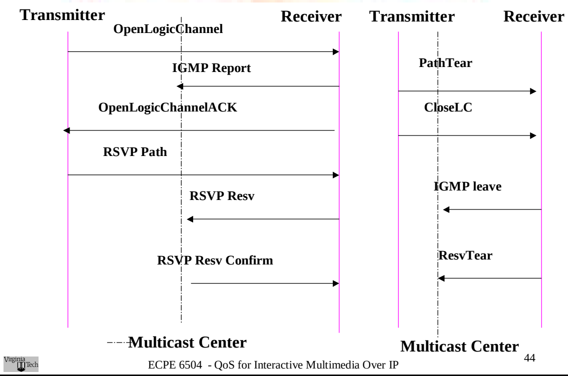
Qos Support for current H.323

- When an endpoint requests admission with a Gatekeeper, it indicates in the *transportQOS* field of ARQ message whether or not it is capable of reserving resources. It also reports bandwidth requirement in *bandwidth* field.
- The Gatekeeper should then decide either
  - to permit the endpoint to apply its own reservation mechanism for its H.323 session; or
  - to perform resource reservation on behalf of the endpoint; or
  - that no resource reservation is needed at all. Best-effort is sufficient.
- The decision is conveyed to the endpoint in the ACF message.

RSVP for a point-to-point connection



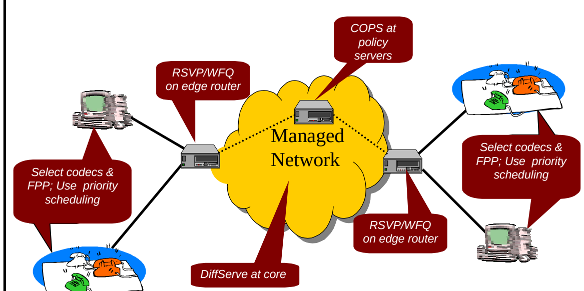
RSVP for point-to-multipoint



Future QoS Deployment

- **Define an end-to-end Framework for QoS**
  - **An architecture that addresses QoS needs at each network stage.**
    - **Endpoint parameter selection/tuning (codec, FPP,etc.)**
    - **QoS at the endpoint (e.g., priority scheduling.)**
    - **QoS at the access (e.g., RSVP/WFQ)**
    - **QoS at the core (e.g., differentiated Services)**
    - **Policy servers to distribute and police QoS deployment (e.g COPS)**

QoS Deployment



Acronyms

- **ITU: International Telecommunication Union;**
- **IETF: Internet Engineering Task Force;**
- **ETSI: European Telecommunications Standards Institute;**
- **TIPHON:Telecommunications and Internet Protocol Harmonization over Networks;**
- **IMTC:International Multimedia Teleconferencing Consortium.**
- **PSN:Packet Switched Network;**
- **CSN: Circuit-switched network;**
- **GSTN: General Switch Telephone network;**
- **GQOS:Guaranteed Quality of Service;**
- **N-ISDN,B-ISDN: Narrow- or Broadband Integrated Service Digit Network;**
- **RAS:Registration,Admission and Status;**
- **COPS:Common Open Policy Services;**
- **WFQ: Weighted Fair Queuing;**

Review Questions

- **Why is VBR MPEG used with packet switched networks?**
- **What are the various QoS parameters involved in interactive multimedia over IP?**
- **What are the characteristics of multimedia traffic?**
- **An H.323 terminal could be a phone or a PC. In a Pc-to-Phone case, a PC can call a phone just by dialing its phone number. In a phone-to-pc case, how can the phone call the PC terminal since the user cannot input the receiver's IP address on the phone dial pad?**
- **True or False: All H.323 calls are signaled by routing through a gatekeeper.**
- **True or False: An H.323 zone can span across multiple networks.**

References

[1] **QoS Negotiation and Resource Reservation for Distributed Multimedia Applications** Kurt Rothermel, Gabriel Dermler, Walter Fiederer University of Stuttgart Institute of Parallel and Distributed Systems (IPVR) Breitwiesenstraße 20-22, D-70565 Stuttgart, Germany {rotherme, dermler, fiederer}@informatik.uni-stuttgart.de

[2] **End-to-End Delay Analysis of Videoconferencing over Packet-Switched Networks** Mario Baldi, Member, IEEE, and Yoram Ofek, Member, IEEE, IEEE/ACM TRANSACTIONS ON NETWORKING, VOL. 8, NO. 4, AUGUST 2000

[3] C. Huitema and T. Turletti, **"videoconferencing on the Internet",** IEEE/ACM Transactions on Networking, Vol. 4, No. 3, June 1996.

[4] H. Kanakia, P. Mishra, and A Reibman, **"An adaptive congestion control scheme for real-time packet video transport",** Proc. ACM SIGCOMM'93, SanFrancisco, CA, Sept. 1993, pp 20-31.

[5] HTTP://www.packetizer.com/iptel/h323

[6] HTTP://www.iec.org/tutorials/h323/index.html

[7] **Jerry D.Gibson, Multimedia Communications, Academic Press, 2001**