Chapter-6

Multimedia System (Pokhara University)

Chapter 6 - Communication Systems in Multimedia

From the communication perspective, we can divide the higher layers of the Multimedia Communication System (MCS) into two architectural subsystems:

- ✓ *Application Subsystem*
- ✓ *Transport Subsystem*

Application Subsystem

This subsystem includes the software and the tools with which the end user directly interacts. E.g. of application can be email, video conferencing software etc.

Collaborative Computing

Collaborative computing is the computer supported cooperative work supported by the networks, PCs and the software that facilitates the cooperation. The examples of collaborative computing tools are *electronic mail, bulletin boards, screen sharing tools, text-based conferencing systems, telephone conference systems, conference rooms* and *video conference systems.*

Collaborative Dimensions

The collaboration dimensions are

- ✓ *Time*
- ✓ *User Scale*
- ✓ *Control*

Time:

According to time there can be two types of collaborative work and they are *asynchronous* where the cooperative works do not happen at the same time; while the other is the *synchronous* where the cooperative works happen at the same time.

User Scale:

The user of the application can be a single user interacting to the other user or to a group of users. Email between two individuals is a user to user interaction where as Email to a group is a user to group interaction. Video conferencing is also a user to group interaction application. There can be following types of groups.

- ✓ Dynamic or Static Group: When the new users can join the group for cooperative work in the real time the group is said to be dynamic where as when the number of members and their membership is predefined it is static group.
- ✓ The members of the group may simply be a *participant* of the collaborative work or he/she may be the *co-coordinator, conference initiator, conference chairman, a token holder* or just an *observer.*
- ✓ The members of the group may have *homogenous* or *heterogeneous* characteristics. For e.g. they may belong to different ethnic group or they may differ in the level of intelligence.

Control:

Control during the collaboration can be *centralized* or *distributed.* Centralized control means there is a chairman who controls the collaborative work and every group member reports to him. Distributed control means every group member has control over his/her own tasks in the collaborative works.

Group Communication Architecture

Group communication is a cooperative activity which may be synchronous or asynchronous which may be central control or a distributed control. A group communication architecture consists of:

- ✓ Support model
- ✓ System model
- ✓ Interface model

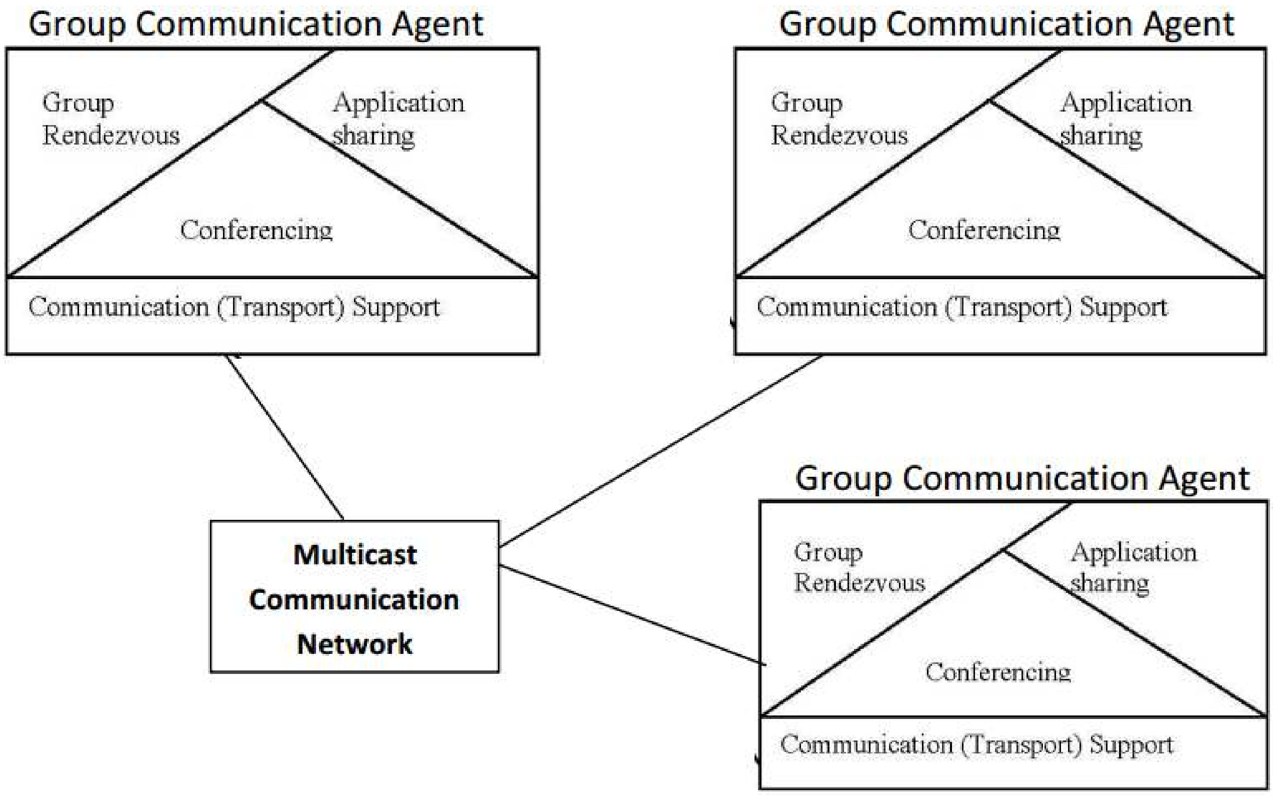

Figure: Communication Support Model

The Support Model:
It includes *group communication agents* that communicate via a multi-point multicast communication network as shown in the figure above. Group communication agents may use the following for their collaboration:

Group Rendezvous:
It represents methods which allows one to organize meetings, and to get information about the group, ongoing meetings and other static and dynamic information. The rendezvous can be Synchronous where the Directory services allows the access to information stored in a knowledge base about the conference, registered participants, authorized users and name and role of the participants. Explicit invitations method sends invitations either point-to-point or point-to-multipoint to conference participants. Asynchronous rendezvous methods may be implemented through email or bulletin boards.

Shared Applications:
It refers to the techniques which allow the replication of the information which can be delivered to all participants of the collaborative work. It may use the Centralized Architecture or Replicated Architecture. In the former a single copy of the application runs at one site say server. All participants post their information to the central application which again distributes the information to the end users. This architecture is easy to manage and is less bandwidth hungry. In the replicated architecture, a copy of the shared application runs locally at each site. Input events to each application are distributed to all sites and each copy of the shared application is executed locally at each site.

Conferencing:
It represents the service which manages the multiple users to communicate and interact with each other by the use of multimedia data. Thus conferencing is basically a management service that controls the communication among multiple users via multiple media, such as video and audio, to achieve simultaneous face-to-face communication. The conferencing services control a *conference* by the implementation of following functions

- ✓ Establishing a conference
- ✓ Closing a conference
- ✓ Adding new users and removing users who leave the conference

The conference may enforce its control centrally where the state of the conference is located on a central machine or it may enforce the control in the distributed manner.

The System Model:
It is based on a client-server model where the clients are applications that provides interface to the users who interact with the system while servers refers to function which makes it possible for the clients to communicate with each other and manage the communication.

The Interface Model:
It includes the *user presentation protocols* and *group work management protocols.* User presentation protocol is the interface available to the end users from which they can initiate, join, manage, communicate and terminate the conference. Group work management protocols specify the communication between the client and the servers for services like registration and querying the status of the conference.

Session Management

A session is the total logged in time of a user or it can be the entire conference from its commencement to its termination. The management of session is a very important task and it should consider several issues like allowing users to join and leave the conferencing, selection of the coordinator, distributing information between users.

Architecture of Session Management
It consists of following components:

- ✓ Session manager
- ✓ Media agent
- ✓ Shared workspace agent

Session Manager:
It includes local and remote functionalities. The local functionalities includes the

- ✓ Membership Control management: Authenticating the users or allowing members to join the session.
- ✓ Control Management: It may involve floor management which involves the distribution and sharing of the information and resources available to the conference.
- ✓ Media Control Management: It is required for the synchronization between the different media.
- ✓ Configuration Management: It refers to the exchange and optimization of the QoS parameters.
- ✓ Conference Control Management: It consists of functionalities for starting, changing and closing the conference.

Media Agents:
These are responsible for decisions specific to each type of media. Each agent performs its own control mechanism over the particular medium, such as mute, unmute, change video quality, start sending, stop sending etc.

Shared Workspace Agent:
The shared workspace agent transmits shared objects (e.g., telepointer coordinate, graphical or textual object) among the shared applications.

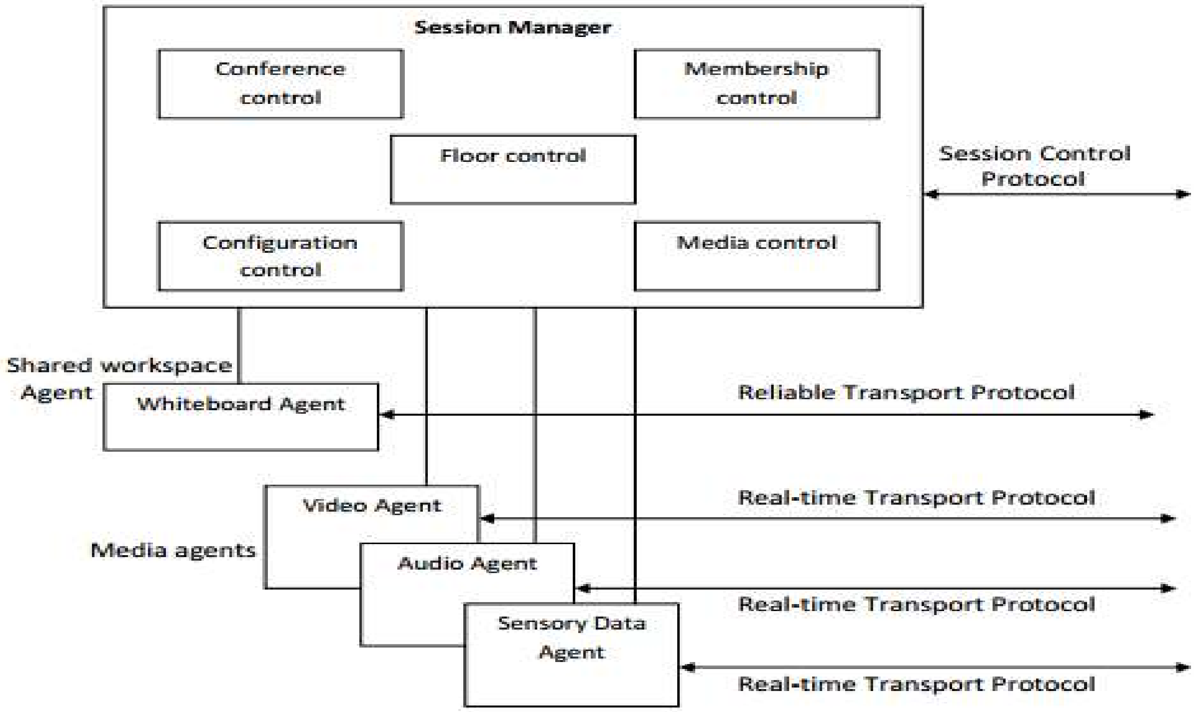

Figure: Session Control Architecture

Control
Each session is represented by its state which may the number of current users, media types, and the policies that are implemented.

Session Management includes two steps to process session state: an *establishment* and a *modification* of the session. During the establishment, the session manager negotiates, agrees, and sets the logical state of its own session. Modification means redefining the session variables. The control mechanisms implemented for the management of session includes

Floor Control:
Floor control is required to control and manage the access to the shared workspace, the resources etc. It is also required for maintaining the data consistency. In a typical *floor passing mechanism* at any time, only one participant has the floor. The floor is handed off to another participant when requested.

Conference Control:
The conferencing services control a *conference* by the implementation of following functions

- ✓ Establishing a conference
- ✓ Closing a conference
- ✓ Adding new users and removing users who leave the conference

The conference may enforce its control centrally where the state of the conference is located on a central machine or it may enforce the control in the distributed manner.

Media Control:
It is required for the synchronization of the different media streams or allocating them bandwidth.

Configuration Control:
It refers to the exchange and optimization of the QoS parameters, managing the media quality and users' requirement.

Membership Control:
It refers to the sending invitation to the members, registration of users and modification of the membership.

Transport Subsystem

Requirements

User and Application Requirements
Networked multimedia applications by themselves impose new requirements onto data handling in computing and communication because they need: *Sustainable data throughput, fast data forwarding, service guarantees, and multicasting.*

Data Throughput
This requirement wants the processing of the system to be fast and effective.

Fast Data Forwarding
The users or application wants very low end to end delay and jitter when communicating multimedia data. The holding time should be very less due to the real time requirement.

Service Guarantees
The loss of the information is undesired and the system or protocol used must ensure that the information is delivered to the intended destination.

Multicasting
It is important for sharing the bandwidth and the communication protocol processing at end-systems.

Processing and Protocol Constraints
Processing system and protocols have constraints which need to be considered while processing and transmitting multimedia information.

- ✓ Following the "shortest possible path" for quicker delivery
- ✓ Buffer management
- ✓ Segmentation and reassembly
- ✓ Retransmission on error
- ✓ Error-recovery
- ✓ Asynchronous transfer

Transport Layer
Transport protocols, to support multimedia transmission, need to have new features and provide the following functions:

- ✓ Timing information
- ✓ Semi-reliability
- ✓ Multicasting
- ✓ NAK (none-acknowledgement)-based error recovery mechanism
- ✓ Rate control

Internet Transport Protocols

TCP (Transmission Control Protocol)
It was designed to provide a reliable end-to-end byte stream over an unreliable inter-network. Each machine supporting TCP has a TCP transport entity, either a library procedure, a user process, or part of the kernel. In all cases, it manages TCP streams and interfaces to the IP layer. A TCP entity accepts user data streams from local processes, breaks them up into pieces not exceeding 64KB, and sends each piece as a separate IP datagram.

The IP layer gives no guarantee that datagrams will be delivered properly, so it is up to TCP to time out and retransmit them as needed. Datagrams that do arrive may well do so in the wrong order; it is also up to TCP to reassemble them into messages in the proper sequence. A key feature of TCP is that every byte on a TCP connection has its own 32-bit sequence number. The sending and receiving TCP entities exchange data in the form of segments. A *TCP segment* consists of a fixed 20-byte header followed by zero or more data bytes. However *TCP is not suitable for real-time video and audio transmission because its retransmission mechanism may cause a violation of deadlines which disrupt the continuity of the continuous media streams.*
UDP (User Datagram Protocol)

The Internet protocol suite supports a connectionless transport protocol, *UDP (User Datagram Protocol).* UDP provides a way for applications to send encapsulated IP datagrams and send them without having to establish a connection. UDP transmits *segments* consisting of an 8-byte header followed by the payload. UDP does not do flow control, error control or retransmission upon receipt of a bad segment. All of that is up to the user processes. Many multimedia applications use this protocol because it provides to some degree the real-time transport property, although loss of PDUs may occur.

Real-time Transport Protocol (RTP)
RTP is a UDP protocol used in the client server environment and in the real-time multimedia applications. The multimedia application consists of multiple audio, video, text, and possibly other streams. These are fed into the RTP library, which is in the user space along with the application. This library then multiplexes the streams and encodes them in RTP packets, which it then stuffs into a socket. At the other end of the socket, UDP packets are generated and embedded in IP packets.

The basic function of RTP is to multiplex several real-time data streams onto single stream of UDP packets. The UDP stream can be sent to a single destination or to multiple destinations by unicasting and multicasting respectively.

Each RTP stream has a sequence number. If a packet is missing, the best action for the destination to take is to approximate the missing value by interpolation. It has no flow control, no error control, no acknowledgements, and no mechanism to request transmissions.

Xpress Transport Protocol (XTP)
XTP integrates transport and network protocol functionalities to have more control over the environment in which it operates. XTP is intended to be useful in a wide variety of environments, from real-time control systems to remote procedure calls in distributed operating systems and distributed databases to bulk data transfer. It defines for this purpose six service types: *connection, transaction, unacknowledged data gram, acknowledged datagram, isochronous stream* and *bulk data.*

In XTP, the end-user is represented by a *context* becoming active within an XTP implementation. Two contexts are joined together to form an *association.* The path between two XTP sites is called a *route.* There are two types of XTP packets: information packets which carry user data, and control packets which are used for protocol management. It provides flow control and retransmission.

There are some features which meet the requirements for multimedia communication, such as

- ✓ XTP provides a connection-oriented transport and network transmission; hence it gives the benefit to map XTP on ATM networks and to use the possibilities of bandwidth reservation of ATM networks.
- ✓ Different transport services are provided: connection-mode, connectionless-mode and transaction-mode.
- ✓ It has error management and retransmission mechanism.
- ✓ It has rate-based flow control which allows it to provide a convenient mechanism for throughput and bandwidth reservation when QoS request is issued.

There are few problems though:

- ✓ XTP requires VLSI while most current implementations are done in software and hence are slower.
- ✓ If the round rotation time of the network fluctuates it enters into handshake stage continuously which is undesirable.
- ✓ It has large header (44 bytes) which is an overhead.
- ✓ Source identification and discrimination is missing in XTP.
- ✓ Internetworking with other protocols is not worked out to provide QoS handling and resource reservation.

Network Layer

Internet Protocol

In the TCP/IP protocol stack the network layer protocol is the *Internet Protocol (IP)* and, in order to transfer packets of information from one host to another, it is the IP in the two hosts, together with the IP in each Internet gateway and router involved that perform the routing and other harmonization functions necessary.

The IP in each host has a unique Internet-wide address assigned to it. This is known as the host's *Internet address* or, more usually, its *IP address.* Each IP address has two parts: a *network identifier* and a *host identifier.*

IP provides a connectionless best-effort service to the transport layer above it which is either the transmission control protocol (TCP) or the user datagram protocol (UDP). The source IP first adds the destination and source IP addresses to the head of the block, together with an indication of the source protocol (TCP or UDP)., to form the *IP datagram.* The IP then forwards the datagram to its local gateway. At this point the datagram is often referred to as a *packet.* An IP packet consists of header part and a text part. The header has a 20-byte fixed part and a variable length optional part.

Routers act as the guide to these packets. According to the information they store in their *routing table* they guide the datagrams to their destination. The routing is often determined according to congestion, shortest possible path etc. The recipient of the IP packets strips off the header from the packet and passes the block of information contained within it- known as the *payload*- to the peer transport layer protocol indicated in the packet header.

If the length of the packet is greater than *maximum transfer unit* (MTU) of the receiver the packet is broken into segments. There is a *Fragment offset* field in the IP header which tells where in the current datagram this fragment belongs. All fragments except the last one in a datagram must be a multiple of 8 bytes.

The *Time to live* field is a counter used to limit packet lifetimes while the *Protocol* field specifies the transport layer protocol the packet is to be handed to. All IPv4 addresses are 32 bits long and are used in the *Source address* and *Destination address* fields of IP packet.

Interconnectivity between Internet Protocol and Underlying Networks
The protocol used in Internet, data used and the services vary for e.g. there could be datagram transmission or multimedia traffic and the communication may take place through twisted pair telephone network, an Ethernet or an ATM B-ISDN. Hence, the mapping between the Internet protocol and the underlying layers is very important. More than that each and every machine is identified with its Ethernet address which is 48 bit in length so mapping of IP address to its corresponding Ethernet address is very important and which is done through ARP (Address Resolution Protocol). A related protocol, reverse ARP, can be used to map the Ethernet address to IP address.

Routing
Routing is used for guiding the packets from its source to the destination. Routers are dedicated for this purpose. Routing is based upon the congestion information, shortest path method. These routers may be administered by a common authority and are called *Autonomous Systems (AS)* for which protocol- *Interior Gateway Protocol (IGP).* ASs of gateways exchange reachability information by means of *Exterior Gateway Protocol (EGP).*

Internet Group Management Protocol (IGMP)
Multicasting in Internet is done with the help of multicast routers. About once a minute, each multicast router sends a hardware multicast to the hosts on its LAN asking them to report back on the groups their processes currently belongs to. Each host sends back responses for all the class D addresses it is interested in. These query and response packets use a protocol called *IGMP (Internet Group Management Protocol).* It has only two kinds of packets: query and response, each with a simple, fixed format containing some control information in the first word of the payload field and a class D address in the second word.

Resource Reservation Protocol (RSVP)
In order to ensure that the real-time traffic flows does not exceed that which is allocated for it, the resources required for each flow are reserved in advance of each packet flow starting. The resources can be bandwidth and buffer capacity. The protocol used to do this is called *Resource Reservation Protocol.* Because many of the new real-time applications involve multiple participants, RSVP is used to reserve resources in each router along either a unicast or a multicast path. When making a reservation, a receiver can specify one or more sources that it wants to receive from. It can also specify whether these choices are fixed for the duration of the reservation or whether the receiver wants to keep open the option of changing sources later. The routers use this information to optimize bandwidth planning. Once a receiver has reserved bandwidth, it can switch to another source and keep that portion of the existing path that is valid for the new source.

Quality of Service and Resource Management
During a multimedia communication, the services in the multimedia systems need to be parameterized. Parameterization of the services is defined in ISO standards through the notion

of Quality of Service (QoS). Each service can be characterized by a quality of service. As a simple example some services are reliable i.e. they do not loose data while some are unreliable as they may loose data. The parameters can be bandwidth, maximum and minimum end to end delay, jitter, buffer allocation etc. There are several issues that need to be addressed and they are:

QoS Layering

The QoS requirement is associated with each layer of the OSI model, or the TCP/IP model. However the QoS for multimedia communication system (MCS) consists of three layers: application, system and devices. Application means the software and the program parameters, where as system refers to the overall system of communication and then in the network.

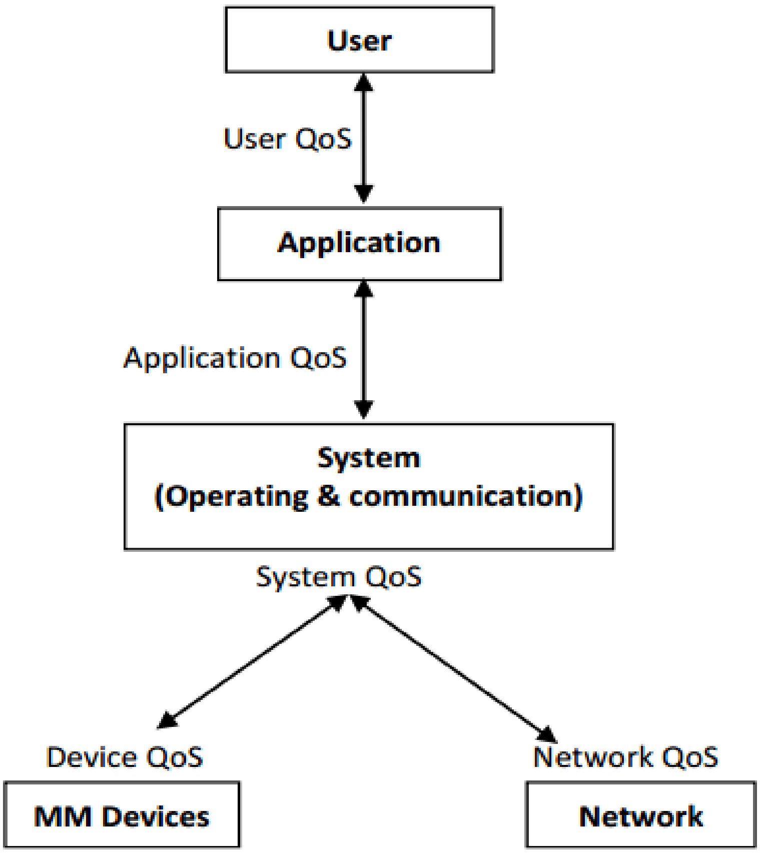

Figure: QoS-layered model for the MCS

Service Objects

Services are performed on different objects, for example, media sources, media sinks, connections and Virtual Circuits (VCs), hence QoS parameterization species these *service objects.*

QoS Description

The QoS is described in terms of the required parameters by the end systems.
The *application QoS* parameters may include media quality, transmission delay, jitter, synchronization.

The system QoS parameters describe requirements on the communication services and OS services resulting from the application QoS. They may be specified in terms of both *quantitative* (bandwidth, PDU size, buffer size) and *qualitative criteria* (level of synchronization, order of data delivery, recovery).

The network QoS parameters describe requirements on network services. They may be specified in terms of network load (packet size, service time) and network performance (congestion, delay).

The device QoS parameters typically specify timing and throughput demands for media data units.

QoS Parameter Values and Types of Service
There are three major type of service and they are:
- ✓ *Guaranteed Services*
- ✓ *Predictive Services*
- ✓ *Best-effort services*

In the Guaranteed services the QoS parameter values are deterministic or statistical in nature. It may ensure the lossless transmission of data i.e. reliable transmission.

In the Predictive Services the QoS parameter are predictable with the help of the past parameters. Though the exact value of the parameters may not be known a rough estimate can be made.

In the Best-effort services, the QoS parameters depend on the load of the network. It ensures that the best possible service is provided to the multimedia data.

Resource
Resource is a system used for processing, storing, manipulating data.
- ✓ The resource can be *active* and *passive.* The active resource can be a CPU which processes data or manipulates data where as the passive resource is a bandwidth which only serves a particular purpose.
- ✓ A resource can be *exclusive* i.e. used by a single process or it may be *shared* where it is shared between various processes.
- ✓ The resource may be *single* or it may be *multiple.*

Resource Management Architecture
The main goal of resource management is to offer guaranteed quality of service. It addresses three main actions
- ✓ Reserve and allocate resource
- ✓ Provide resources according to QoS specification
- ✓ Adapt to resource changes during on-going multimedia data processing.

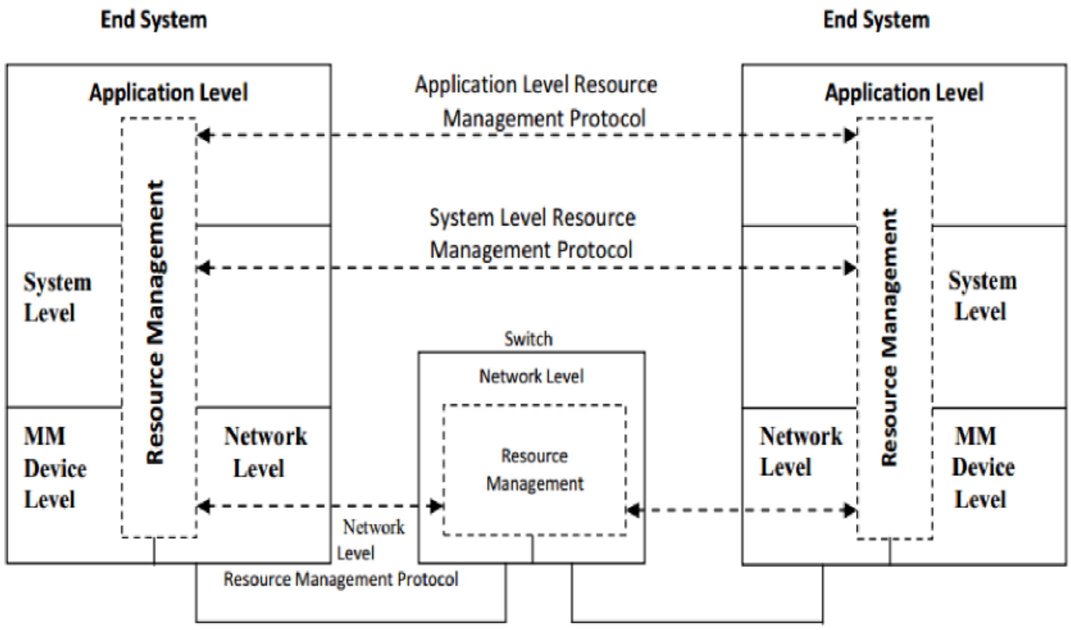

Figure: Resource Management in MCSs

Relation between QoS and Resources

The QoS parameters and their corresponding resources are mapped for the management of resources. For e.g. the end to end delay QoS parameter determines the behavior of transmission services along the path between source and sink with respect to packet scheduling, queuing and task scheduling. Description of a possible realization of resource allocation and management shows the QoS and resource relation.

Establishment and Closing of the Multimedia Call

QoS Negotiation

It can be:

Bilateral Peer-to-Peer Negotiation

Negotiation occurs between the two service users and the service provider is not involved.

Bilateral Layer-to-Layer Negotiation

It occurs between the service user and the service provider.

Unilateral Negotiation

The user and the provider cannot modify the QoS parameter. It is based on “take it or leave it” model.

Hybrid Negotiation

The negotiation between host and sender is bilateral layer-to-layer negotiation and negotiation between network and host-receiver is unilateral.

Triangular Negotiation for Information Exchange
The user specifies its required QoS parameters while the provider may change it according to the possibility and availability before confirming that the caller agrees upon it.

Triangular Negotiation for a Bounded Target
It is similar to the above method but in this type of negotiation the caller specifies both the target value and the minimum required value. If the parameter that the provider does not meet the minimum required value the caller rejects the provider.

Triangular Negotiation for a Contractual Value
In this case, the QoS parameters are specified through a minimal requested value and bound of strengthening. The goal of this negotiation is to agree on a contractual value, which in this case is the minimal request QoS parameter value.

Translation
The QoS parameters for the users and application, system and network are different. However it must always be possible to derive all QoS values from the user and application QoS values. This derivation is known as *translation* which may require "additional knowledge" stored together with the specific component.

The translation functions used for the purpose are as follows:

- ✓ *Human Interface – Application QoS*
  The user must be able to tune the QoS parameters that are provided by the application. For e.g. in case of video he/she must be able to specify the resolution, and the compression ratio.
- ✓ *Application QoS – System QoS*
  The application level parameters are mapped to the System parameters for e.g. frame size specification at application level can be mapped as segmentation and reassembly.
- ✓ *System QoS – Network QoS*
  It maps the System QoS parameters like transport packet, end-to-end delay into the underlying network QoS parameters like in ATM, the end-to-end delay of cells and vice versa.

Scaling
Scaling is a process of sub-sampling the data stream and only present a fraction of its original contents which can be done either at the source or at the receiver. Scaling can be

- ✓ *Transparent Scaling*
  In the transparent scaling the scaling done by the lower levels is not visible to the application level or the higher level.
- ✓ *Non-Transparent Scaling*
  Here the higher level notices that the lower levels have performed scaling and it may compensate the loss of data stream accordingly.

For *audio,* transparent scaling is a difficult task because even a small drop in data stream would be noticeable to the listeners.

For video the scaling can be Temporal Scaling which reduces the resolution of the video stream in the time domain i.e. the number of video frames transmitted within a time interval decreases. In Spatial Scaling the number of pixels of each image in a video stream. Frequency Scaling reduces the number of DCT coefficients applied to the compression of an image. Amplitude Scaling reduces the color depth of the image pixel. Color space scaling reduces the number of entries in the color space.

Resource Admission
It is a procedure in which the source specifies the resources and their parameters that can be available to the user. The user tests for the availability of resource are called *admission tests.* Based on the results of the admission tests, the reservation protocol creates either a “reserve” message with admitted QoS values or a “reject” message when the minimal bound of QoS values cannot be satisfied.

The tests can be
- ✓ *Schedulability test* : e.g. CPU schedulability or packet schedulability
- ✓ *Spatial Test* : for buffer allocation for delay and reliability guarantees
- ✓ *Link bandwidth Test:* for throughput guarantees

Resource Reservation/Allocation
It can be *pessimistic* or *optimistic.* In the former the user system doubts that the resources will be available when communication will begin so, it reserves the required resources in advance. In the later, the user system reserves the required system only when required.
The different data structures and functions that are used are: *Resource Table, Reservation Table, and Reservation function.*

Managing Resources during Multimedia Transmission

Rate Control
A rate-based service discipline is one that provides a client with a minimum service rate independent of the traffic characteristics of other clients. Several rate-based scheduling disciplines have been developed:

Fair Queuing:
If N channels share an output trunk, then each one should get 1/Nth of the bandwidth. The users are usually determined by first come first serve approach combined with round robin.

Virtual Clock
A virtual transmission time is allocated to each packet. It is the time at which the packet would have been transmitted, if the server would actually be doing Time Division Multiplexing.

Delay Earliest-Due-Date
If the source obeys a peak and average sending rate, then the server provides bounded delay. The server sets a packet’s deadline to the time at which it should be sent, if it had been received according to the agreed contract.

Jitter Earliest-Due-Date
Here the server sets the jitter boundary and at the entrance of router and switch it is ensured the maximum value of allowable jitter is not exceeded.

Stop – and – Go
Here if a packet arrives at a frame n, the packet is sent only at frame n+1. It ensures that the packets that would arrive quickly are instead being delayed.

Hierarchical Round Robin
The server sets a priority and discriminates between the packets according to the priority. The packets at the same level get equal share of bandwidth. However packets at lower level get lesser bandwidth than the higher level.

End – to – End Error Control

Error Detection
Error Detection can be done by parity check, cyclic redundancy check or by including the checksum values.

Error Correction
Go-back-N Retransmission
If a packet arrives with an error, all the subsequent packets are also rejected unless the packet with error is retransmitted.

Selective Retransmission
If a packet arrives with error, it is discarded but the subsequent packets are not rejected but are buffered. The error packet is then asked to be retransmitted and subsequent processing occurs.

Forward Error Correction (FEC)
FEC uses Huffman codes, Checksum mechanism so that the packets can be corrected at the receiver and there is no need of retransmission.

Slack Automatic Repeat Request (S-ARQ)
For voice when there is jitter the receiver observes gaps, which result in interruptions of continuous playback of the voice stream. With this approach the first packet is artificially delayed at the receiver so that the buffer becomes full and the playback would exhibit no gaps. The *slack time* of a packet is defined as the difference between its arrival time at the receiver and its playback time. S-ARQ allows the control time to extend so that this time can be used to lengthen the slack time of arriving packets.

Resource monitoring
Resource monitoring function is responsible for checking the proper utilization of the resource and the shared resource is fairly used by the servers. For this the different parameter related to resource is exchanged with the help of agents.

Monitoring can add overhead during multimedia transmission, which should not cause violation of QoS guarantees. Monitoring at end-systems includes a *supervisor function* to continuously observe that the processed QoS parameters do not exceed their negotiated values. Different protocols like *Resource Administration Protocols, Simple Network Management Protocol* are used for the purpose.

Resource Adaptation
In the network it should be possible to dynamically change the QoS parameters and hence the use of Resources. This should be supported by the protocols for reporting the QoS changes in the existing connections and adaptive resource schemes to respond to and accommodate the changes either in the network, the hosts or both.

User Request for Renegotiation
If the user-receiver required change of QoS for the receiving media, first the resource manager checks the local resource and reserves it. Then, the sender is notified via a resource administration protocol and the same admission procedure follows as in the case of a user-sender requiring QoS changes.

Host System Request for Renegotiation/Change

If host QoS changes result in degradation of the application performance, the host resource manager may invoke the resource administration protocol to lower the QoS parameters in the network between the sender and receiver.

Network Request for Renegotiation/Change
The load in the network can be adapted or it may not be adapted. In the latter case the QoS parameters have to be changed and the request is made to the resource manager.

Network Adaptation
In order to properly balance the network load and to maintain the negotiated QoS it is necessary to

- ✓ increase network availability
- ✓ allow network administrators to reclaim resources
- ✓ reduce the impact of unscheduled, run-time maintenance on clients with guaranteed service

Network adaptation refers to balancing the load, maintaining the network performance, efficient routing and solving congestion.

Source Adaptation
Another alternative reaction to changes in the network load is to adapt the source rate according to the currently available network resources. This approach requires *feedback* information from the network to the source which results in graceful degradation in the media quality during periods of congestion.

Trends in Collaborative Computing
Simplification of Groupware Development
Groupware development can be simplified by dividing it into components which can later be combined to serve its purpose. Groupware should have a plug-and-play architecture.

Shared Abstractions and Standard Interfaces
In order to adapt with the heterogeneity the abstractions needs to be shared and the interfaces should be standardized. This is possible by the involved parties to meet a common specifications and abstractions.

Standard Protocols
For the inter-working and simplify the collaborative computing protocols needs to be standardized so that users working with systems from different vendors can communicate.

Self-describing Media Agents
The media agents should be descriptive i.e. they should be able to describe their capabilities and requirements.

Translations
Translations among different Computer Supported Collaborative Work sites are needed to overcome hardware and software heterogeneity, for example, to bridge different media encoding schemes.

Trends in Transport Systems
The trends in transport systems go in two directions:

Special-purpose protocol approach
It deals with designing various special-purpose protocols on top of IP for different classes of applications. E.g. Development of RTP for the delivery of data and services that have real time requirement.

General-purpose protocol approach
It aims at providing a general set of services that the user can pick and use. An example is XTP, where the user can select one-way, two-way or three-way handshaking for connection setup and release, etc.

Multimedia Database Management System

The main task of the Multimedia Database Management System (MDBMS) is to abstract from the details of the storage access and its management. It deals with efficient storage of multimedia data so that retrieval, modification and other operations are possible.

It provides several properties like:

Persistence of Data
The data should persist for a large span of time so that even the old data are available whenever they are required.

Consistent View of Data
In multi-user systems it is important to provide a consistent view of data during processing database requests at certain points. This property is achieved using *time synchronization protocols.*

Security of Data
The data in the database system should be accessible only to the authenticated users and no others should be able to access data.

Query and Retrieval of Data
The database should be able to execute the query of the users and retrieve the required information.

References:

- ✓ Multimedia: Computing, Communications and Applications", Ralf Steinmetz and Klara Nahrstedt, Pearson Education Asia
- ✓ "Multimedia Communications, Applications, Networks, protocols ad Standards", Fred Halsall, Pearson Education Asia
- ✓ "Multimedia Systems", John F. Koegel Buford, Pearson Education Asia

Assignments:

(1) Describe briefly about the Application and Transport Subsystem in Multimedia Communication System.
(2) Describe Group Communication Architecture in brief.
(3) Discuss the requirements of Transport Subsystem for Multimedia Communication.