Optimization Of Multimedia Data Over Wireless Network

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***Abstract*—Optimizing multimedia data over wireless network by utilizing efficient quality of service factors to improve the effectiveness of data with providing compression technique and data security. In this project, we propose a QoS-aware joint working compression algorithm and dynamic buffering algorithm to support stable multimedia data over wireless network. The proposed dynamic buffering algorithm estimates the packet loss by giving their threshold value and performs its own functions based on the information. Main aim of our project is to replace the static buffer with dynamic buffer and allocating the queue to the buffer. That minimizes the overloading of buffer and PLR ratio. Further, Compression technique is applied over the data to reduce the size of data manually and to provide the security to data over wireless network cryptography is used. The proposed scheduling algorithm continuously updates the control parameter to pursue an effective tradeoff between the quality-of-service of multimedia data and the network throughput. The main goal of this project is to provide result as performance of QoS that will help in the design process of buffering for wireless networks, aiming to get better QoS. Finally, simulation results are provided to show the performance of the proposed video streaming system.**

I. Introduction

Mobile communication technologies are continuously evolving and hence it becomes essential to revise QoS parameters for next generation of wireless communication. An important objective of our software is to provide enhanced video quality with an improved quality of service that meets the need of customer. This can be achieved with continuous mobility for reliable wireless communication by using intelligent dynamic buffering techniques, GZIP compression and RSA algorithm. With the progress of high data rate for multiple data formats it is necessary to support cognitive compression techniques.

In high speed mobile networks, data transmission process should be completed quickly. Hence, available amount of data for scheduler is limited and also PLR should be minimized. So in current research work we attempt to further optimize QoS parameters for WN. Also current work includes dynamic memory allocation techniques to avoid packet loss over WN. An effective data security is provided through RSA algorithm .These algorithms offers high throughput with less amount of delay.

II. Background

QoS is especially important for Internet applications such as video-on-demand, voice over IP (VoIP) systems, and other consumer services where high-performance and high-quality streaming is involved. QoS (Quality of Service) refers to a broad set of networking technologies and techniques designed to guarantee predicable levels of network performance. Elements of network performance within the scope of QoS include availability (uptime), bandwidth (throughput), latency (delay),and error rate (packet loss). Many home broadband routers implement QoS in some form. QoS involves prioritization of network traffic. QoS detects different kinds of network traffic (video, audio, gaming) according to its data types and makes dynamic routing decisions based on predefined priorities. Allocation of Buffer Dynamically for data Packets to be transmitted. One of the main requirement of video streaming is, to delivered the data packet within a certain deadline. To Overcome the problem of Buffer overloading,transmission impairments dynamic buffering and cognitive data security techniques are used.

III. Literature review

In order to support the rich demand of real-time multimedia services like video streaming, VoIP, and etc., it is necessary to ensure the quality-of-service (QoS) requirements are met and packet loss ratio (PLR) is minimised by keeping it below the applications required threshold. In a video streaming service environment, it is important to maintain the PLR threshold below 1that the QoS requirements of video streaming service users are satisfied.

Compressed video data is generally of variable bit rates due to the generic characteristics of the entropy coder and scene changes/inconsistent motion changes of the underlying video. Furthermore, video streaming services are time constrained and wireless channel is inherently time-varying. These facts make the problem more challenging over wireless network.

As far as static buffering is concerned, buffer gets overloaded whenever the data of size greater than buffer threshold value is sent over the network.In order to avoid this data must be broken down in packets and each time buffer must get flushed to avoid packet overloading.This concept mainly emphasizes on allocating buffer dynamically.Hence buffer is to be allocated dynamically whenever data is sent over wireless network.

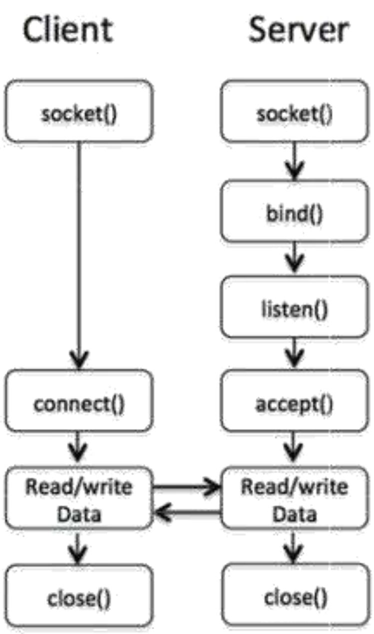

Fig. 1. FUNCTIONS OF SOCKET PROGRAMMING.

A. *PROPOSED METHOD*

B. *Socket Programming*

Socket programming basically deals with connection establishment in between client and server.It is controlled by various functions like socket,bind,listen,accept,read,connect,read or write,close which runs implicitly on programming platforms like java,c,cpp.Once the connection gets established we can send data on the network which in turn returns the respond from server side.Multiple clients can communicate with one server via getting access to servers ip address and local host whenever its not necessary.Normally, a server runs on a particular pc and has a socket that is bound to a specific port number. The server is in listening mode to the socket for a client to make a connection request. On the client-side: The client knows the host name of the machine on which the server is running and the port number on which the server is listening. To make a connection request, the client tries to rendezvous with the server on the server's machine and port. The client also needs to identify itself to the server so that it can binds to a local port number so that, it will be used during this connection. This is usually assigned by the system.If everything goes well, the server accepts the connection,otherwise it won't. Upon acceptance, the server gets a new socket bound to the same local port and also has its remote endpoint set to the particular address and port of the client machine. It needs a new socket, so that it can continue to listen to the original socket for connection requests while tending to the needs of the connected client.On the client side,if the connection is accepted, a socket is successfully created and the client can use the socket to communicate with the server.The client and server can now communicate by writing to or reading from their sockets.

Working: Sockets provide the communication mechanism between two computers using TCP. A client program creates a socket on its end of the communication and attempts to connect that socket to a server.

When the connection is made, the server creates a socket object on its end of the communication. The client and the server can now communicate by writing to and reading from the socket. The java.net.Socket class represents a socket, and the java.net.ServerSocket class provides a mechanism for the server program to listen for clients and establish connections with them. The following steps occur when establishing a TCP connection between two computers using sockets

The server instantiates a ServerSocket object, denoting which port number communication is to occur on. The server invokes the accept() method of the ServerSocket class. This method waits until a client connects to the server on the given port.After the server is waiting, a client instantiates a Socket object, specifying the server name and the port number to connect to.The constructor of the Socket class attempts to connect the client to the specified server and the port number. If communication is established, the client now has a Socket object capable of communicating with the server.On the server side, the accept() method returns a reference to a new socket on the server that is connected to the client's socket.After the connections are established, communication can occur using I/O streams. Each socket has both an OutputStream and an InputStream. The client's OutputStream is connected to the server's InputStream, and the client's InputStream is connected to the server's OutputStream.TCP is a two-way communication protocol, hence data can be sent across both streams at the same time. Following are the useful classes providing complete set of methods to implement sockets.

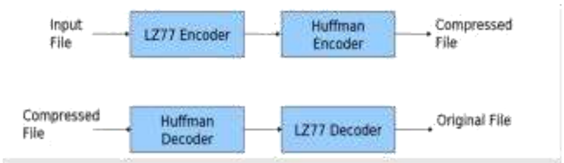

Fig. 2. GZIP Compression.

1) COMPRESSION: Compressing data can be a lossless or lossy process. Lossless compression enables the restoration of a file to its original state, without the loss of a single bit of data, when the file is uncompressed. GZIP Lossless compression is the typical approach with executables, as well as text and spreadsheet files, where the loss of packets or frames would change the information. GZIP Lossless compression permanently eliminates bits of data that are redundant, unimportant or imperceptible. GZIP Lossless compression is useful with graphics, audio, video and images, where the removal of some data bits has little or no discernible effect on the representation of the content.

Algorithm: Gzip compression

1) Choose any multimedia file as an input
2) pass the input file to LZ77 encoder
3) After getting an encoded file compute the hash index of bytes.
4) Filter out repeated strings with the help of hash chaining.
5) Compressed file will be int the form of .gzip ,oass it to LZ77 decoder and huffman decoder to decompress it.
6) Extract the original file .
7) Examine the hash index each time to filter replicated strings

2) DATA SECURITY: RSA derives its security from the difficulty of factoring large integers that are the product of two large prime numbers. Multiplying these two numbers is easy, but determining the original prime numbers from the total – factoring – is considered infeasible due to the time it would take even using today's super computers. The public and the private key-generation algorithm is the most complex part of RSA cryptography. Two large prime numbers, p and q, are generated using the Rabin-Miller priliminary test algorithm. A modulus n is calculated by multiplying p and q. This number is used by both the public and private keys and provides the link between them. Its length, usually expressed in bits, is called the key length. The public key consists of the modulus n, and a public exponent, e, which is normally set at 65537, as it's a prime number that is not too large. The e figure doesn't have to be a secretly selected prime number as the public key is shared with everyone. The private key consists of the modulus n and the private exponent d, which is calculated using the Extended Euclidean algorithm to find the multiplicative inverse with respect to the totient of n. Generally,key size of 2048 bits is used which is difficult to break.

RSA Algorithm:

1) Choose two distinct prime numbers, p and q.
2) Let $n = pq$.
3) Let $(pq) = (p1)(q1)$. (is totient function).
4) Pick an integer e such that $[1 < e < (pq)]$, and e and (pq) share no divisors other than 1 (e and $[(pq)]$ are coprime).
5) Find d which satisfies the given geometrical condition .
6) Encryption Sender A does the following:-
7) Obtains the recipient B's public key (n, e).
8) Computes the cipher text $c = m^e$ mod n.
9) Decryption: $1.m{=}c^d$ mod n

3) RSVP PROTOCOL: The RSVP resource-reservation process initiation begins when an RSVP daemon consults the local routing protocol(s) to obtain routes. A host sends IGMP messages to join a multicast group and RSVP messages to reserve resources along the delivery path(s) from that group. Each router that is capable of participating in resource reservation which passes incoming data packets to a packet classifier and then queues them as necessary in a packet scheduler.

4) TCP/IP: TCP/IP specifies how data is exchanged over the internet by providing end-to-end communications that identify how it should be broken into packets, addressed, transmitted, routed and received at the destination. TCP/IP requires little central management, and it is designed to make networks reliable, with the ability to recover automatically from the failure of any device on the network. The two main protocols in the internet protocol suite serve specific functions. TCP defines how applications can create channels of communication across a network. It also manages how a message is assembled into smaller packets before they are then transmitted over the internet and reassembled in the right order at the destination address.IP defines how to address and route each packet to make sure it reaches the right destination. Each gateway computer on the network checks this IP address to determine where to forward the message.

Socket is a class name used at client side where as TCP-socket used at server side.Whenever client sends a request to server each time a new thread is created.Server has authority to either reject or accept at request.If server accepts that request, a connection gets established. This simply tends to create multiple threads,whenever client tries to communicate to server TCP/IP protocol divides data into packets which are sent over the network.These packets are nothing but sequence of bytes,which are send one by one over the network.For successful transmission of packets from client to server or vice versa, TCP/IP protocol is used.

5) DYNAMIC BUFFER: Dynamic buffer is a remedy to avoid buffer overloading by allocating buffer dynamically based on size of incoming multimedia data. Working: The data which is send over the network is in the form of packets.So by extracting 9022386 number of bytes every time the data of amount mentioned is stored in the buffer. Now if size of file is greater than threshold value of buffer,original data will get divided into fragments of size 9022386 and every time we are going to flush that buffer.It means we are reading and writing

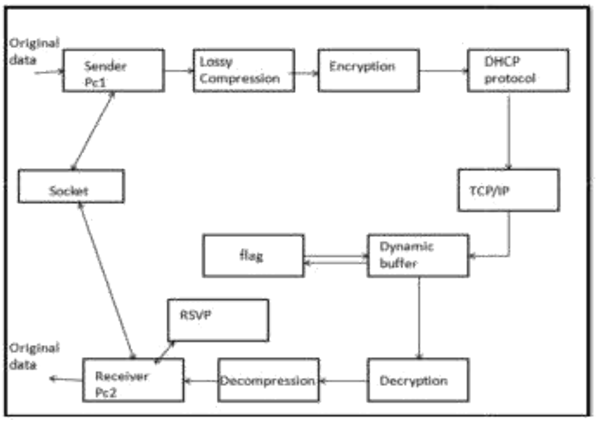

Fig. 3. ARCITECTURE.

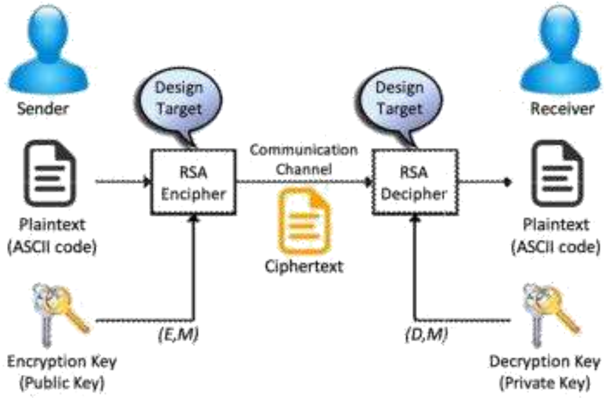

Fig. 4. RSA.

the data with to different threads. This makes buffer empty whenever new data fragments are sent towards buffer.

6) ARCHITECTURE: Working: Socket programming works on the basis of implicit argument passing through arguments like port number,IP address,localhost.Socket programming is a basic platform to establish connection between client and server.If Single PC is taken in consideration,the port number and local host are sufficient to communicate between client and server. As far as multiple clients are concerned, the system is not bother about port number,but servers IP address must be encountered so as to connect with server.Whenever exception occurs in try block controlled automatically get shifted in catch block which executes a current conditions.Multimedia data includes various data formats such as .MP3,.MP4,.png,.jpg which is taken as an input. This multimedia data undergoes gzip encoding which results into encoded data. This encoded data is allowed to encrypt, by using RSA algorithm which converts encoded data file into cipher file.After compression tar.gz file get created.Whenever hacker or attacker tries to get access to original file instead getting original file,he gets compressed file which in the form .gz extension and due to cryptographic security attacker will not get access to that file.This compressed file is sent for further decryption process which actually converts cipher file to original file. Same file is decompressed by using gzip algorithm.All the data packets which are sent over the network are handled TCP/IP protocol.Data is stored in such a buffer which is allocated dynamically,so as to avoid buffer overloading.Multiple clients can be configured by using DHCP protocol(one server and seven clients).

7) ANALYSIS AND RESULT: 1.Compression Multimedia file is compressed by using gzip algorithm.Whole scenario of compression can be analysed with the help of graph shown in figure.Compression is applied to the original files twice in a process. Initially, less amount of compression is obtained before sending the file over network.The percentage of compression increases furthermore when it is sent over the network.Deviation in percentage of compression is observed for different multimedia files.

2.Quality of service

Packet transfer rate: In order to improve the quality of service time required to transmit packets and number of packets must be in inverse proportion.Hence,number of packets varied in particular time bound. As far as image is concerned, 4-5 packets are transferred per second and number of packets are varied as time increases.In case of audio file,3-4 packets are transferred per second as shown in figure.It varies for different file formats with respect to time.

Delay: Delay is given by $D = N/R$, where D = transmission delay N = Number of bytes R = Packet ratio

Here, N = 1MB ,R = 5, D = 1/5=0.2,which is negligible. Hence, quality of service is maintained.

Latency: Variation in delay is stated as latency.Delay is directly proportional to latency.As Delay decreases, latency also decreases.

3.Data Security:

As far as data security is concerned ,it depends on key factors like key size of algorithm,key management,encryption methodologies and no of possible iterations that is to be performed to get access to the original data file.RSA along with Huffman encoding makes it more tedious to break the security of the currently transmitted data because the time complexity of the algorithms is much more than expected.In case of RSA algorithm,it becomes more complex,as the key size of algorithm increases and this makes attackers ,so difficult to get access to the original file.2048 bits is the most appropriate key size as far as RSA is concerned.Also,creation of tar.gz file is transferred over the network and it is not possible to have access to that file.This is due to complexities of RSA and HUFFMAN algorithms.

8) About fig.6,7: As shown in figure 6,no of seconds and no of packets deviates dynamically.The graph of no of packets is and no of seconds is plotted.It shows us that 3 to 4 packets are transferred per unit,then 5 and so onn.. It typically specifies about audio file packet ratio and shows that how this project helps to improve quality of service.In case of audio file it is

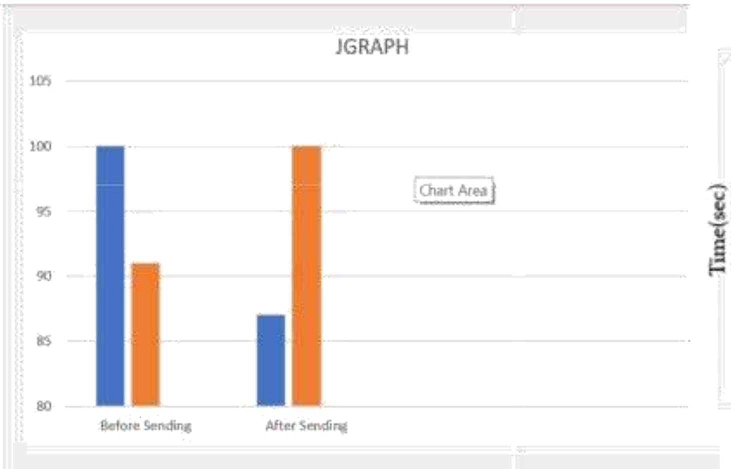

Fig. 5. ANALYSIS OF DATA COMPRESSION.

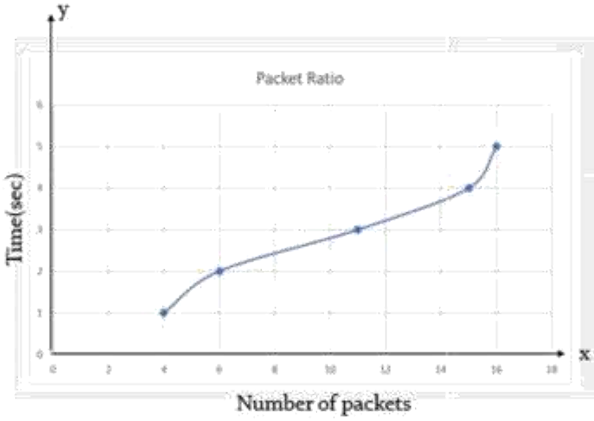

Fig. 7. PACKET TRANSFER RATE FOR IMAGE FILE.

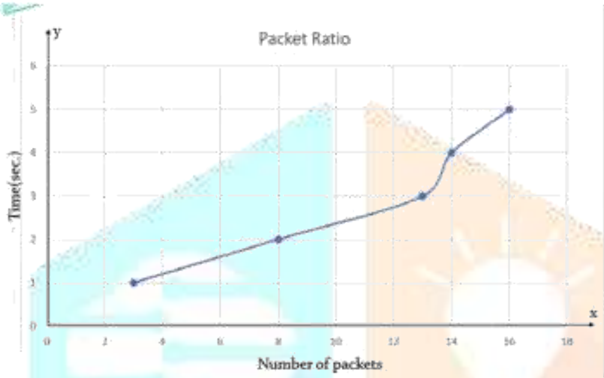

Fig. 6. PACKET TRANSFER RATE FOR AUDIO FILE.

important to maintain the quality of audio ,even after applying compression.The same thing can be verified with the help of graph.

In next fig(7) 4 to 5 packets are transferred per minute the 6 and so on.In case of image file,compression can be applied for png,jpg files and different files results in different compression ratios.Also,with the help of Gzip compression quality of image is maintained.Nature of graph is steady in between 3rd and 4th second.

IV. Conclusion

In this paper we have proposed a methodology to overcome the problem of buffer overloading.With proper implementation of RSA,Huffman,Gzip algorithm it maintains the quality of service factors like delay.jitter.latency,packet transfer rate,etc.We are trying to embed such a system which can work successfully in between 1 server and multiple clients for data transmission without any data loss even after compression of file.Our main motive is to avoid buffer overloading whenever large sized data is to be transmitted and it must take very less amount of time to get transmitted.Also,we are providing large key size for RSA(2048 bits) which is much more complex to split it down into numbers.This helps us to avoid different security attacks and data security is enhanced.If we take transmission time into consideration,the we will come to know that delay is very negligible as compared to the general scenario of normal file transfer.That is why quality of service is improved.This system empirically determines packet ratio and amount of data compressed based on multimedia data file format.

V. Tabular Analysis for Multimedia File Formats

The table shows different compression rates and packet ratio for various file formats. The major multimedia data formats like image,video,audio and text are taken in consideration.Table shows 4 various file format analysis grades which are done on the basis of compression rates and their respective packet ratio.First column is about compression ratio and second is about packet ratio.First row values stands for image formats like .png,.jpg,etc.Remaining 3 row values are for video,audio and text files respectively.

SR no	compression	packet ratio
1	0.25	2.5
2	0.5	4.33
3	0.36	2.0
4	0.47	3.5

VI. Future Work

We have proposed the system to minimize the packet loss.This minimization improves the live streaming.For this smooth live streaming,Even if the transmission gets faster,frames should not get overlapped.If we try to implement this it will help us to obtain enhanced data which can be resulted into better data quality.In future software will work more efficiently even outside of the network domain that can be accessed through default gateway(Access point).Data sharing and streaming will become more smoother than the current one.

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