Performance Modeling of HTTP and RTP Streaming for QoS Support over Next Generation Networks

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ABSTRACT
Attempts to display media on computers date back to the earliest days of computing in the mid-20th century. However, little progress was made for several decades, primarily due to the high cost and limited capabilities of computer hardware. From the late 1980s through the 1990s, consumer-grade personal computers became powerful enough to display various media. The primary technical issues related to streaming was having enough CPU power and bus bandwidth to support the required data rates and creating low-latency interrupt paths in the operating system (OS) to prevent buffer under run. However, computer networks were still limited, and media was usually delivered over non-streaming channels, such as by downloading a digital file from a remote server and then saving it to a local drive on the end user's computer or storing it as a digital file and playing it back from CD-ROMs. The challenges of new communication architecture are to offer better quality of service (QoS) in internet Network. A large diversity of services based on packet switching in 3G network and beyond 3G leads dramatic changes in the characteristics and parameter of data traffic. Through this paper we propose a streaming solution to offer better QoS over 3G and 4G networks. A comparative analysis of HTTP streaming and RTP streaming has been done and packet loss, average bandwidth utilization and total streaming time has been measured. Result shows RTP provides better quality of service over HTTP for multimedia based content transfer over various network. Simulation is done using Java Media Framework.

General Terms
Performance, Design, Verification.

Keywords
QoS-Quality of service, NGN-Next generation network, RTP-Real-time Transport Protocol, HTTP - Hyper Text Transfer Protocol Heterogeneous networks.

1. INTRODUCTION
Internet is rapidly evolving from a text/images medium into a multimedia platform. Fuelled by broadband adoption, various kinds of multimedia content such as audio and video streams are quickly becoming the preferred content types. Advances in technology have steadily increased the features and computing power across the board in all computing devices and cell phones and handhelds have benefited from these advances as well. Many efforts have been made in past to discuss multimedia based content transfer over the network [8].Initially the cell phone were used for voice based transmission but due to advancement with the technology and network it is being used for various other applications such as a music player, a gaming device and multimedia based content transfer. Some issues have been discuss related with QoS support over 3G and 4G network [5].

However, currently there is no standard way of delivering the video to the cell phone, due to several problems such as limited device memory/storage size, incompatible video protocols, and Video processing CPU requirements. Also, due to different codec supported by the mobile and mobile screen size, mobile devices are not able to process the large content and could not support the format of the requested video file.

As per the current existing multimedia transmission system, which mostly works on the concept of http streaming, a research has shown that such concepts are unnecessary introducing delay in the network and condition got worse during peak hour of internet usage. Hence, there is a need of such methods that can ensure the effective transmission of multimedia data during peak hours of internet usage and which can efficiently make a use of available bandwidth resource.

Streaming media is multimedia that is constantly received by and presented to an end-user while being delivered by a streaming provider. The name refers to the delivery method of the medium rather than to the medium itself. The distinction is usually applied to media that are distributed over telecommunications networks, as most other delivery systems are either inherently streaming such as radio, television or inherently non-streaming such as books, video cassettes, audio CDs. Live streaming, more specifically, means taking the media and broadcasting it live over the Internet. The process involves a camera for the media, an encoder to digitize the content, a media publisher where the streams are made available to potential end-users and a content delivery network to distribute and deliver the content. The media can then be viewed by end-users live.

2. HTTP STREAMING
Conceptually, HTTP Live Streaming consists of three parts: the server component, the distribution component, and the client software. The server component is responsible for taking input streams of media and encoding them digitally, encapsulating them in a format suitable for delivery, and preparing the encapsulated media for distribution. The distribution component consists of standard web servers. They are responsible for accepting client requests and delivering prepared media and associated resources to the client. For large-scale distribution, edge networks or other content delivery networks can also be used .The client software is responsible

for determining the appropriate media to request, downloading those resources, and then reassembling them so that the media can be presented to the user in a continuous stream.

HTTP server push (also known as HTTP streaming) is a mechanism for sending data from a web server to application. HTTP server push can be achieved through several mechanisms. Generally the web server does not terminate a connection after response data has been served to a client. The web server leaves the connection open such that if an event is received, it can immediately be sent to one or multiple clients. Otherwise the data would have to be queued until the client's next request is received. Most web servers and application servers offer this functionality. HTTP streaming and quality of experience over 3G and 4G network have been discussed in many latest research paper in past [4][9].

In this technique, the server takes advantage of persistent HTTP connections and leaves the response perpetually "open" (i.e. it never terminates the response), effectively fooling the browser into continuing in "loading" mode after the initial page load would normally be complete. The server then periodically sends snippets of java script to update the content of the page, thereby achieving push capability. By using this technique the client doesn't need Java applets or other plug-ins to keep an open connection to the server. The clients will be automatically notified by new events, pushed by the server. One serious drawback to this method, however, is the lack of control the server has over the browser timing out. A page refresh is always necessary if a timeout occurs on the browser end. Some efforts have been taken in past to discuss HTTP streaming over live content and maintaining quality of service [6]. HTTP streaming is a good option for websites with modest traffic, i.e. less than about a dozen people viewing at the same time. For heavier traffic a more serious streaming solution should be considered.

3. RTP STREAMING

The Real-time Transport Protocol (RTP) defines a standardized packet format for delivering audio and video over IP networks. Using RTP in context with 3G and 4G networks is the latest research area and it has been discussed in latest research forum [2][3]. RTP is used extensively in communication and entertainment systems that involve streaming media, such as telephony, video teleconference applications and web-based push-to-talk features.

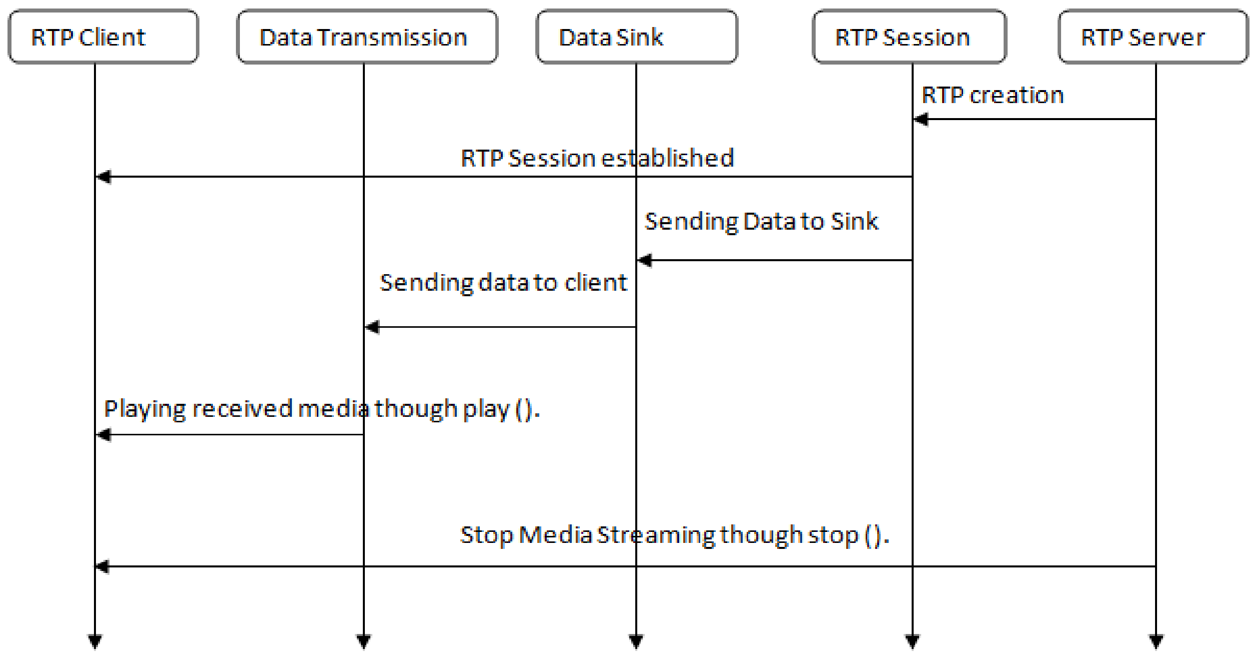

Fig 1. Sequence diagram of RTP streaming

RTP is used in conjunction with the RTP Control Protocol (RTCP). While RTP carries the media streams (e.g., audio and video), RTCP is used to monitor transmission statistics and quality of service (QoS) and aids synchronization of multiple streams. When protocols are used in conjunction, RTP is originated and received on even port numbers and the associated RTCP communication uses the next higher odd port number.

4. MAINTAINING QUALITY OF SERVICE

Quality of Service refers to the broad collection of networking technologies and techniques. The goal of the QoS to provide guarantees on the ability of a network to deliver predictable. Elements of Network performance within the scope of QoS often include availability bandwidth latency and error rate. It can also refer as a resource reservation control mechanism to achieve the service quality. QoS involves prioritization of network traffic. QoS can be targeted at a network interface, toward a given server or router's performance, or in terms of specific applications. A network monitoring system must

typically be deployed as part of QoS, to insure that networks are performing at the desired level.
QoS is especially important for the new generation of Internet applications such as VoIP, video-on-demand and other consumer services. Some core networking technologies like Ethernet were not designed to support prioritized traffic or guaranteed performance levels, making it much more difficult to implement QoS solutions across the Internet.
In multimedia based services QOS is essentially having two phases. Initial setup phase and real-time multimedia exchange phases. In the audio based application, end to end transmission delay should be small enough so that interference should not affect the normal conversation. A number of studies based on the integration of call signaling with resource negotiation and maintaining quality of service over next generation network can be found in the literature [10][11].
In order to provide a requested QOS, the nodes of a network must perform session initiation phase, reservation setup, admission control, policy control, packet scheduling, and packet classification functions. Quality of Service refers to the broad collection of networking technologies and techniques. The goal of the QoS to provide guarantees on the ability of a network to deliver predictable. Elements of Network performance within the scope of QoS often include availability bandwidth latency and error rate. It can also refer as a resource reservation control mechanism to achieve the service quality. QoS involves prioritization of network traffic. QoS can be targeted at a network interface, toward a given server or router's performance, or in terms of specific applications. A network monitoring system must typically be deployed as part of QoS, to insure that networks are performing at the desired level. QoS is especially important for the new generation of Internet applications such as VOIP, video-on-demand and other consumer services. Some core networking technologies like Ethernet were not designed to support prioritized traffic or guaranteed performance levels, making it much more difficult to implement QoS solutions across the Internet. High QoS is often confused with a high level of performance or achieved service quality, for example high bit rate, low latency and low bit error probability. This project has tried to achieve good quality of Service, for example video communication with minimum delay in the wireless environment. Bit Error Rate Analysis provides the best configuration to achieve QoS with minimum Bit Error Rate. QoS issues have been discussed in past in context with 3G and 4G networks [12].

5. DESIGN AND CONFIGURATION

Simulation has been done using Java Media Framework. Though there are many simulation platform available and discussed in past to do the simulation for multimedia content over the network [1] [7]. Figure 1 shows the architecture for media streaming. Java Media Framework (JMF) is a Java library that enables audio, video and other time-based media to be added to Java applications and applets. Initial configuration has been done by bit streaming technique with the proper bandwidth analysis concept that reduces the packet loss and increase the efficient utilization of bandwidth. This module is done in NetBeans application builder environment with proper GUI and all the statistics.

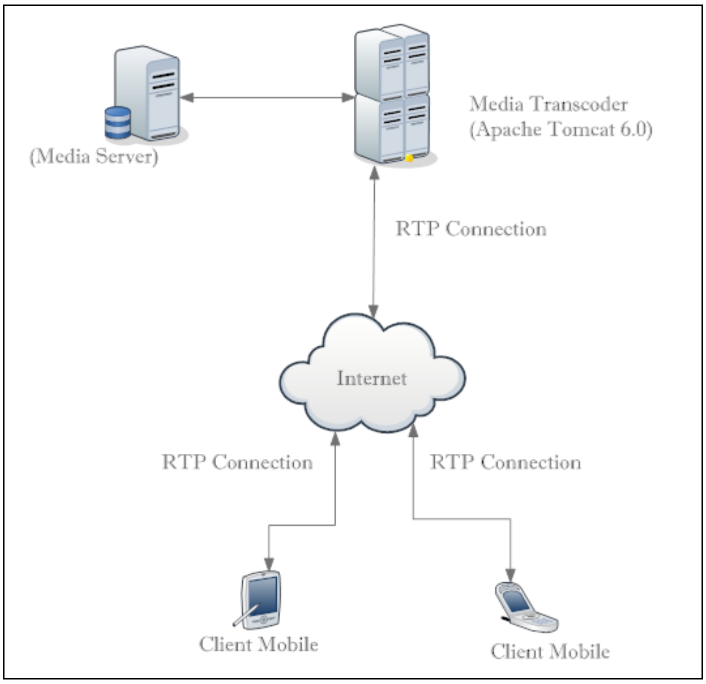

Fig 2. Architecture for media streaming

This optional package, which can capture, play, stream, and trans code multiple media formats, extends the Java Platform, Standard Edition (Java SE) and allows development of cross-platform multimedia applications. MF binaries are available under a custom license, and the source is available under the SCSL. The current version ships with four JAR (file format) files, and shell scripts to launch four JMF-based applications. Axence NetTools is used to monitor various network parameters. It is a comprehensive set of administration tools. NetWatch is a network monitoring with e-mail alerting, real-time and historical charts of response time and packet lost. WinTools lists exhaustive system information from Windows computers: running processes and services, registry, event log, disk, memory and CPU information.

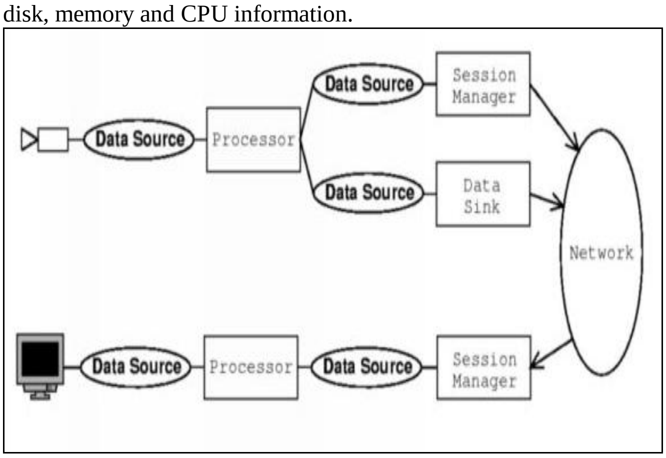

Fig 3. JMF Framework

6. SIMULATIONS RESULTS AND ANALYSIS

It is observed from the testing that the speed of the communication is very less affected if more clients are added because project makes a separate session between the communication parties. Because the client mobiles use the internet bandwidth and it is shared among all the clients so traffic is there if huge numbers of client are active at the same time. In Streaming, packet loss is less even if traffic is increased because bit streaming takes care of this.

TABLE 1. COMPARATIVE ANALYSIS REPORT OF RTP AND HTTP

	RTP streaming	**HTTP streaming**
Total time to play a file	44 seconds(23 seconds)	53 seconds(23 seconds)
Average Bandwidth utilization	3782 b/s	4334 b/s
Packet loss	49 sent 48 received	49 sent 43 received

7. CONCLUSION

One of the unexpected results of the latest research effort has been the realization that the uses of RTP protocol without severe complexity. Here the network architecture has been tested on variable network connection for RTP and HTTP for multimedia based application. Results have been evaluated for both RTP and HTTP protocol. It has been observed that packet end to end delay and bandwidth utilization is less by using RTP protocol. Packet loss has been minimizing to great extent by using RTP. It shows fair results from various network connections. General objective of achieving high degree of quality of service over 3G and 4G networks can be met by using RTP.

8. ACKNOWLEDGMENT

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