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Assessing Network Parameters by Web Real-time Communications

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***Abstract:** In the context of networks where assurance of information delivery is a prime user requirement, it becomes essential to estimate the key performance indicators and carry out a proactive analysis to ascertain if the current network conditions would meet the Quality of Service requirement of particular service. In this project the key is to carry out a QoS aware transmission of Voice, Video and Data over an IP network for ensuring delivery assurance with requisite service specific QoS. An integrate GUI to be deployed at both the sender and the receiver will be developed and this will act as first front end for the transmission and the measurements. An 'active and collaborative tool based or a passive tool based approach' will be used for measurement of network KPI whereas 'COTS (Commercial off the shelf)/FOSS (Free and Open source)/freely downloadable or a custom developed utility/tools' would be used for generation of traffic.*

***Keywords** : Web-RTC, Network, Parameters, QoS and Protocol.*

I. INTRODUCTION

Today's internet usages are migrating to triple way functionalities and it is necessary to provide good Quality Of Service for the networks in real-time like Video Conferences, Telephony using IP, e-commerce and e-business etc., The current internet model is well suited for the traditional applications like transferring a file, sending emails, browsing and chatting etc., But these model doesn't provide the quality, guaranteed and timely carriage of the actual packets. Today's Multimedia based functionalities in the real-time applications need very good guarantee, timely delivery without delay and loss of the actual packets.

The delays in the real time applications are highly sensitive. It leads to reproduction of the continuous events like images and speech. The data packets reaching the destination should not be delayed to enable playing at the exact time. If the packet was not arrived to the destination in time or lost in the transmission media, then it leads to generation of a gap in the information availability to the user. It reduces the quality of the audio or video production at the destination and the performance will degrade. The performance reduction is directly proportional to the amount of delay or loss of the packets in the transmission media. In real time applications, the destination device will not wait for the whole data to be available at the end. It begins to play the received stream of data immediately once the packets of the video or audio has been received at the end. The stream pattern of the media is not defined earlier by the duration. It is not required to wait for the long time duration and need to be downloaded for playing the audio or video.

The real time applications expect the packets to be available in the correct timing. The existing protocol doesn't have a mechanism to request for the lost packets to resend and will not wait for the packet to be received from source system again. The round trip delay between the source and the destination will be more in the synchronization process. TCP doesn't have any mechanism to handle this and so User Datagram Protocol (UDP) will be used for carrying this type of packets. But, UDP doesn't have the facility of recovering the lost packets. It doesn't give the guarantee of the packet delivery and not bother about the order of packet delivery. But the voice or video conferencing applications need to be guaranteed for timely delivery without any loss and delay of the packets in the transmission media. . This is the major constraint for real-time applications. The Brief history of the proposed WebRTc is shown in Figure 1.

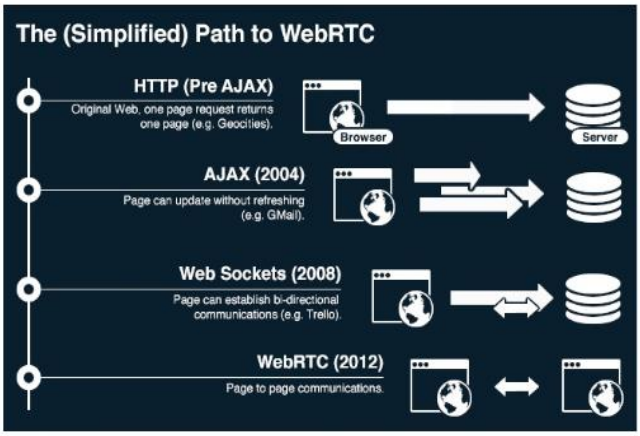

Fig .1 Brief history of WebRTC

II. LITERATURE SURVEY

The real-time applications rely on absolute differentiated services in order to have guarantee on the end-to-end delay. Major resources referred throughout this paper are IETF-RTCWEB standards draft-ietf-rtcweboverview-15, RFC 5766, draft-ietf-rtcweb-rtp-usage-25, draft-ietf-rtcweb-data-channel-13,and draft-ietfrtcweb-security-08.

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To progress digital universities eLearning atmosphere this article propose a two-way system based on WebRTC technology. Using the WebRTC APIs: Media Stream, Peer Connection, RTCDataChannel the author implement a WebRTC signalization server for its design and realization to manage real-time applications.[1]

The WebRTC technology and WebRTC implementation (include client and server), waving are described in this paper. The WebRTC API main parts are described and clarified. By the WebRTC standards Waving methods and protocols are not stated. Therefore a novel signaling mechanism design and implementation has been done in this study. As a WebSocket server The server application is implemented. The use of the WebRTC API is demonstrated by the client application for achieving real-time communication. The WebRTC technology Benefits and future development are mentioned. [2]

A WebRTC design and implementation is introduced based on online video teaching system on android OS in this paper. An user approachable education platform brings out by the teaching system which allows aural and videotape teaching cooperation at any time with members at anywhere. In this system only one-to-one is limited for Online video teaching nowadays because of upper necessities for hardware exploitation, memory employment of Web Real Time Communication. It will be extended to one-to-many in the next stage[3]. The introduction of P2P video conferencing system based on Web-RTC is aim of an article. Within network webRTC provides P2P connections establishment without added plugins and software. The design of scalable live video conferencing structural design is presented by this article based on WebRTC.[4] Analyzing the deficiencies and challenges faced by WebRTC and offering a Multipoint Control Unit or traditional communications entity based architecture as a solution is discussed in this paper. To support the WebRTC by using MCU, for the video conferencing the author proposed a best centralized architecture. how this structure offers resolutions to certain contexts is discussed thoroughly by the author [5]. For the development of the masses communication between people is an important factor. As people can now talk through web using video conferencing which is much more preferred and convenient, Sending letters for communication is a passé.To perform the operation the traditional system of VoIP needs to install plugin or application is drawback that can be overcome with the help of new technology called web-RTC. [6] Without having any issue how to do Advance communication between two or more browsers is presented in this paper. WebRTC build a complete call center solution Using Asterisk Call center is introduced in this paper. By providing an Android Application with inbuilt soft phone, system hope to hair Agents As part time workers in future work [7]. Getting an idea about webRTC features is able by a reader and how the users gain a communication familiarity with webRTC is known by this survey. The Web Real-Time Communications (WebRTC) protocols and the Voice over IP (VoIP) technologies are focused in this survey. Further different insights related to this topic are enable in this survey.From this finally this survey concluded that to make the communication process easy as well as to provide it in a simple way WebRTC helps in enabling the network server.[8] Educates students by the idea of actual peer-to-peer communication for networking and interaction between students is tha aim of this paper.The minimal functionalities are performed by current system to include real time data transfer and also to safe and sound the message among two or more parties to encrypt the multimedia communication network this project can be extended [9]. The Constrained Application Protocol called jCoAP which implemented by a a lightweight Java ispresented by the author in this paper. The author showed that CoAP based message for devices with comparably lesser latencies. To enable immediate communication with jCoAP for multiple devices, different TDMA and time synchronization approaches will be evaluated in future work. [10] To ensure that interoperability is effective among any WebRTC client and web browsers, it is significance for the upcoming success of WebRTC. Before first release candidate of the WebRTC specifications, a comprehensive outlook of different testing challenges researchers have encountered are presented by this survey [11]. Challenges are faced every day by People having hearing disability. Communication is the major challenge faced by them. For deaf people an progressive communication system improvement is the key objective of this paper. The author using internet of things to solve this problem. An assistanting actual communication which offers slight interval is main advantage of this system. For short and long distances this system can be implemented [12]. Using AWS based on a mobile application, weather monitoring system real-time plan is proposed by this paper. Using the WeatherLink software weather conditions sensor data is taken from the AWS Device. The data is passed through the data logger using serial communication, uploaded via FTP and stowed on webserver. Compared to other solution the real-time weather monitoring through the mobile application with a litheness in the limits and the need of UI design successfully shown by this system. [13] Another major resource referred for writing this paper is a published book "High Performance Browser Networking" written by Ilya Grigorik. He works as web performance engineer at Google. In this book author discusses about the network of things behind browser, starting from fundamental limitations to powerful innovations across browser applications such as HTTP 2.0, WebSocket and Peer to peer communication with WebRTC. In this book, author explains about networking protocols (TCP, TLS, UDP, HTTP and many more) and their performance characteristics for building powerful web applications. This book also answers a lot of questions about networking protocols such as why TCP isn't good for transporting media when compared to UDP, why latency is the major problem for better performance and how bandwidth management can be achieved by making reuse of network connections etc. After explaining about browser networking foundations, author has discussed the latest advancements in protocols and browser such as benefits of HTTP 2.0 standard, Use of Websocket for building data channels, and building low latency video conference applications using real-time WebRTC transports.

The next major reference used for writing this technology case study is “WebRTC APIs and RTCWEB Protocols of the HTML5 Real-Time Web”, Edition 3.0. Alan B. Johnston and Daniel C. Burnett are the authors of this book which provides information about the architecture, protocols, application program interfaces (APIs) and technical goals of WebRTC. Dr. Alan B. Johnson works as a distinguished engineer at Avaya, Inc. and he is also working as adjust professor at Washington university in St Louis. Daniel C. Burnett works as a chief scientist at Tropo and he is also performing duties as Director of Standards at Voxeo. Sam Dutton who works as a developer advocate for Google Chrome referred this book as a bible for learning about WebRTC. This book provides great details about various network topologies and signaling pathways involved for WebRTC development.

III. PROTOCOL STACK FOR MULTIMEDIA SERVICES

The Protocol Stack for Multimedia Services is shown in Figure 2.

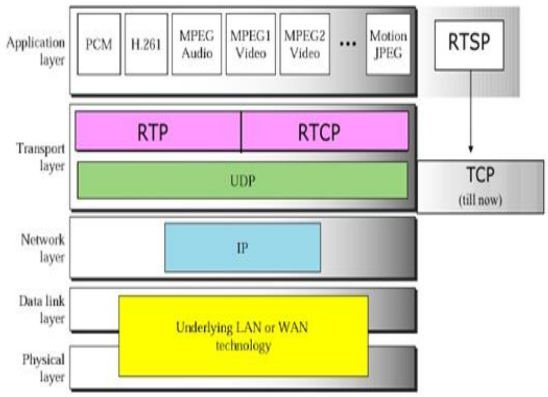

Fig. 2. Protocol Stack For Multimedia Services

IV. REAL-TIME NETWORK TRANSPORTS

Real-time communication is time-sensitive because it is more important that the information has sent on time to the receiver rather than guaranteeing its delivery. While looking at an existing audio video streaming apps one can observe that these have been designed tolerant to packet loss and output quality. If needed applications has to implement their own logic to overcome packet loss and delay in packet transport. Therefore, low latency and timeliness are significantly more critical than reliability for implementing successful real time communication.

In order to meet the above-mentioned requirements, UDP has been preferred over TCP for data transport in real time communications. TCP provides reliable data transport where if packet loss occurs then TCP doesn't continue sending remaining packets instead it buffers all the packets after the lost packet and waits for retransmission until it delivers them in an orderly manner to the application. By comparing UDP with TCP, it differs from the following services. UDP doesn't guarantee message delivery which means no acknowledgment, no retransmission. UDP doesn't guarantee packets being delivered orderly which means no packet sequencing, no reordering. UDP doesn't track for connection status.

UDP doesn't provide congestion control which means there's no built-in network feedback mechanism. The transport layer of the WebRTC uses UDP which delivers the packets the moment they arrive with no sequencing or ordering. This UDP alone in the transport layer won't be enough for implementing successful real time communication. Several other mechanisms along with protocols should be implemented for many other activities like traversing many layers of NAT's and firewalls, negotiating each stream parameter, implementing flow control, providing data encryption and many more.

UDP is the basis for implementing real-time communication in web browser, but will also need a large supporting cast of protocols on top of UDP to meet requirements as shown below in Figure 3.

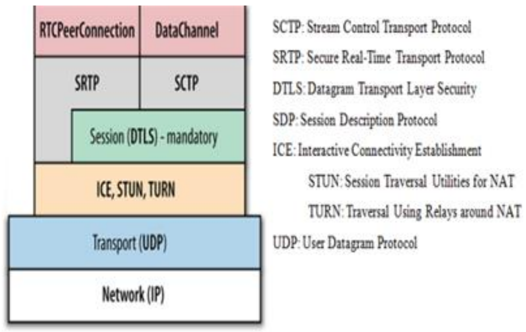

Fig. 3 WebRTC Protocol Stack

V. REAL-TIME TRANSMISSION AND SUPPORTING PROTOCOL

Below Figure 4 is a graph of a typical data communication application, showing periods of low and high network utilization.

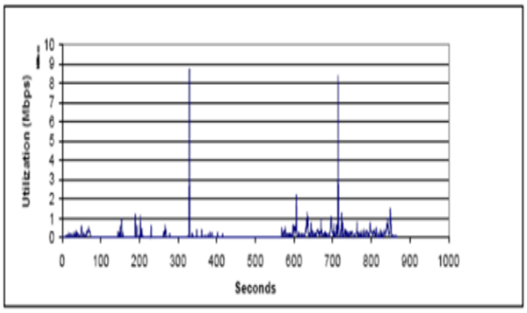

Fig:4 Typical Data Bandwidth Utilization

Real time media streams implies the sending or receiving of stored or live media (voice or video) broadcast or videoconferencing over the internet. Real-time traffic is very 7 different in its characteristics. It results from the output of a codec which is sampling a continuous real-world environment (speech or images) and transmitting constant updates of this information to reproduce the image or speech. So the bandwidth utilization of voice and video is sustained during the time the application is running.

Figure 5 is a graph of a 384K video conference, showing both the audio and video streams and their relatively constant use of bandwidth during operation.

The real-time streams are delay sensitive. It samples and reproduces a continuous event, such as speech or image. Individual data samples must arrive at the destination end to be played at the right time. If a packet is late, or is lost in transit, then there will be a gap in the information available to the player, and the quality of the audio or video reproduction will degrade. This degradation is significant with increasing delay or loss.

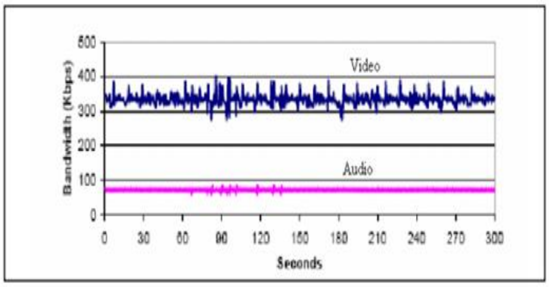

Fig. 5 Bandwidth Utilization For Typical Video Conferencing.

Before start to begin any network check or session transaction, peer has to check whether the remote peer is reachable and willing to set up the connection or not? This mechanism is known as offer and answer where an initiating peer will send an offer to the remote peer. Now the remote peer should send an answer back for a successful peer connection establishment. In order to achieve this offer and answer mechanism, a shared signaling channel is needed between peers to notify each other about the connection as shown in Figure 6.

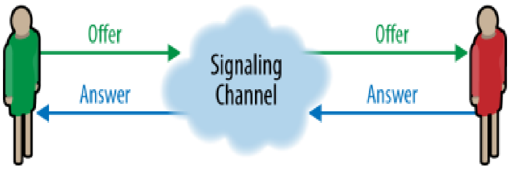

Fig. 6 WebRTC Based Signalling Channel.

The choice of protocol being used for signaling transport has been left to application developers making it easy to choose from a variety of available signaling protocols. Also, it enables the interoperability making it possible for WebRTC to communicate with an existing communication infrastructure like SIP, Jingle, ISDN and many more. In such cases signaling server acts like a gateway to communicate with an existing communication network. The responsibility will be on the configured network to notify the target peer for the connection offered and send the answer back to the WebRTC client that initiated the exchange. If required, WebRTC can also choose to have its own particular signaling service with a custom protocol built-in to communicate the messages. For example, a telephone call can be initiated using PSTN client.

VI. CONCLUSION

Performing dynamic measurements, up on receiving a service request and admitting the service on availability of good enough KPI raises the chance of a successful reception at the receiver with satisfactory QoS. Here we are using iperf3 tool to check the network parameters. Iperf is used when need to run the server and client, respectively, in the test version of the software is the best guarantee is consistent both ends, so will eliminating some of the unnecessary trouble. By default, a TCP connection is used, which is bound to port 5001.

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