QoS for Virtual Reality Software Based on RTCP over the Protocols of IP/UDP/RTP

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Abstract— **Current virtual reality environments are mainly visual experiences, displayed either on a computer screen or through special displays. Providing efficient and reliable network communication solution of virtual reality software has become an increase challenge for the industry experts and researchers, especially regards the aspects of the communication quality over the network. Thus, this paper intends to explore the possibility of using RTCP protocol with existing Panoweaver virtual reality software that uses IP/UDP/RTP communication protocol, in order to provide reliability, using RTCP could improve the existing software and Quality of Service (QoS) in terms of packet loss on increasing bandwidth. Therefore the amount of packet loss during the data transmission will be reduced respectively. Then, network simulation is used in this paper to compare the performance of the Panoweaver software with the Panoweaver-RTCP. This paper has the potentials to be much similar in the real time and is highly scalable and reliable for existing virtual reality software.**

Keywords: Virtual Reality, Panoweaver, RTP, IP, UDP, RTCP, QoS, NS2.

I. INTRODUCTION

Nowadays, there has been a growth in concern in the prospective social impact of new technologies. In general, virtual world use widely for reflecting the self experience towards describing the variety of applications that commonly associated with immersive, highly visual, 3D [1]. Virtual world environments are designed based on applying a definite architecture for allowing network communication among parties, which usually divides the system into small components to process orders/requests from components [2]. This greatly helps to provide an advance services via the distributed processing mechanism. In addition, this distributed design can easily be adapted to work with any advanced network infrastructure for better transmission and delivery of data. An example of virtual reality software is the Panoweaver that use for creating the tour building by allowing stitching of 360 panoramas and publishing the panoramas to a single virtual reality tour showing the 360 degree view of the place. There are three main components that assist Panoweaver virtual reality software to be functioned and distributed among the network such as:

A. Server

The server component which is regarded as the main component of Panoweaver virtual reality software, acts as the controller. Using predefined rules upon conditions and statuses help to manage the communication among the network parties. However, it‘s invisible to users, as the system runs transparently. The system administrator is able to allow proper coordination of numerous sessions of queries and control between clients requests.

B. Client

The client components consist of individual PCs configured with suitable audio and video capturing or playback hardware. The client component comprises of six subcomponents:

i. Interface module (Virtual representation)
ii. Communication module
iii. Communication Control module
iv. Compression/Decompression module
v. Video module
vi. Audio module

C. Data Decompression

Data compression helps Panoweaver virtual reality software to keep continuous stream, which requires to be implemented on a separate machine.

Unfortunately, the current Panoweaver virtual reality software does not provide Quality of Service (QoS) for definite transmission and it relies on Real-time Transport Protocol (RTP) for delivering video and audio content through the Internet during the communication between the Panoweaver virtual reality users.

II. RTP CONTROL PROTOCOL (RTCP)

Some researchers as in [5] explained the RTP Control Protocol (RTCP) as a companion control protocol for RTP which used to provide out-of-band control information for an RTP flow as shown in the next figure and the reception quality feedback, participant identification, and the synchronization between media streams. RTCP runs alongside RTP and provides periodic reporting of this information [3]. Although data packets are typically sent every few milliseconds, the control protocol operates on the scale of seconds. The information sent in RTCP is necessary for synchronization between media stream; i.e.,

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for lip synchronization between (audio and video). The RTCP's two main functions are:

i. It provides feedback on the quality of the media distribution. This function is performed by RTCP receiver and sender reports.

ii. For each sender, RTCP maps RTP time stamps for each RTP stream to a common sender clock [4], which allows audio and video synchronization on the receivers.

The RTCP packets contain direct information for quality-of-service monitoring. The RTCP consumes about 5% of the total bandwidth [5]. The Sender Reports (SR) and Receiver Reports (RR) exchange information on packet losses, delay and delay jitter [6]. This information may be used to implement a TCP like flow control mechanism upon UDP at the application level using adaptive encodings. A network management tool may monitor the network load based on the RTCP packets without receiving the actual data or detect the faulty parts of the network. The RTCP packets carry also a transport-level identifier (called a canonical name) for RTP source, which is used to keep track of each participant. Source description packets may also contain other textual information (user's name, email address) about the source. Albeit the source of the RTP packets is already identified by the SSRC identifier, an application may use multiple RTP streams, which can be easily associated with this textual information.

Novotny and Komosny (2007) introduced an optimization of hierarchical structure in SSM in order to achieve the lowest feedback transmission intervals, because the main issue in which the bandwidth is dedicated for the RTCP protocol. As defined in the RTP specification, it is limited to 5 % of the total allowed bandwidth and hence this creates a limiting factor for large-scale media streaming services based on Source-Specific Multicast-SSM (Figure 1) since the RTCP bandwidth is shared among all the receivers. As a result, noticeable larger delays 30 in sending feedback data from each receiver are encountered. However these noticeable larger delays have been curtailed by a hierarchical structure of receivers with summarization nodes.

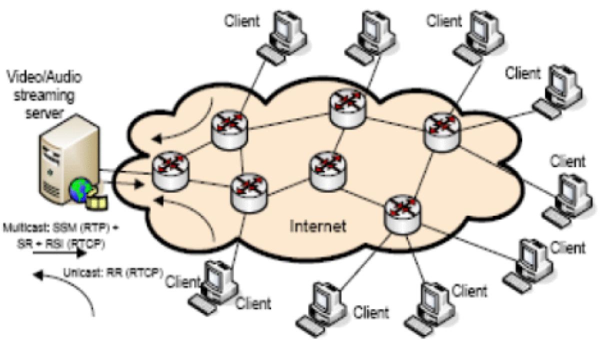

Figure 1: Example for SSM media streaming on internet [7]

While, Randa and Enugnla (2006) established their study according to different communication issues concerning the scalability of RTCP, such as; the delay of feedback and Bandwidth usage problem. They proposed a new scheme to tackle and reduce these problems. This scheme presents a hierarchical architecture, to organize the members dynamically in a hierarchy of local regions, and each region has (AG) aggregator, the members in every region will send their receiver reports RRs with limited scope to reach their aggregator AG. The AG aggregates statistics from the received receiver reports RRs and sends them to a Manager (AG-0) and the main function of the manager is to compute other statistics and evaluate the performance of network and eventually figuring out the regions with network congestion as shown in Figure 2. Even though, some issues were addressed such as congestion and overload due to the number of AGs for the manager (AG-0), and result is the burst of AG that are transmitted to the Manager (AG-0).

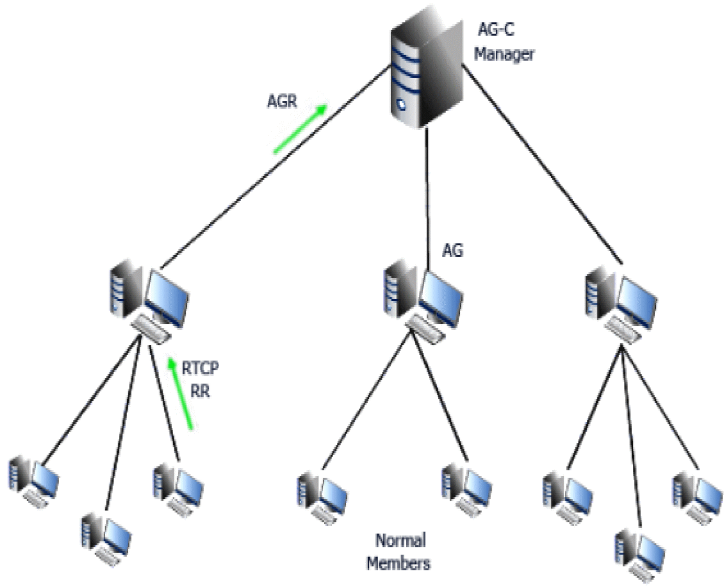

Figure 2: The scalable RTCP Architecture [8]

III. Simulation Environment

This paper test-bed was based on the Wide Area Network (WAN) with one Panoweaver virtual reality software (Demo) that links the clients with the server as shown in Figure 3. In our test scenario generated in the NS-2 simulators, Panoweaver server generates the client data and receives the processed data, presented in Figure 4.

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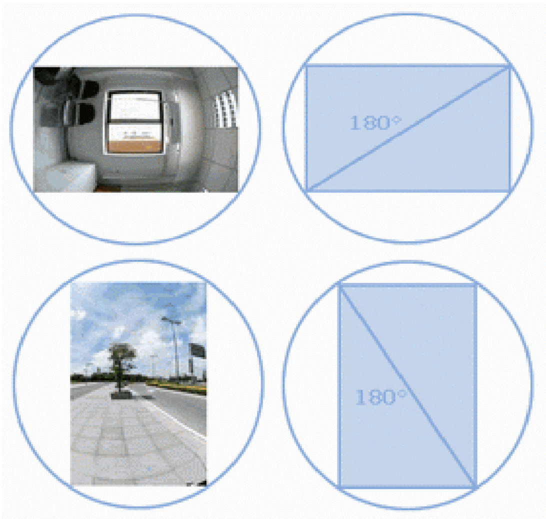

Figure 3: Panoweaver Dimensions

The definition of the sample time interval is the time elapsed between two consecutive receiver reports (RR). This can be expressed as the equation below:

$$\boxed{Network\_rate\_Loss = Network\_lost\ per\ Packet / Network\_recv * 100}$$

In RTCP for the Panoweaver software, RTCP was capable to adjust the frame rates of the generated packet that results in improvised the communication quality but at the same time results large bandwidth over the network. The graph below shows the following scenario for the network topology of Panoweaver.

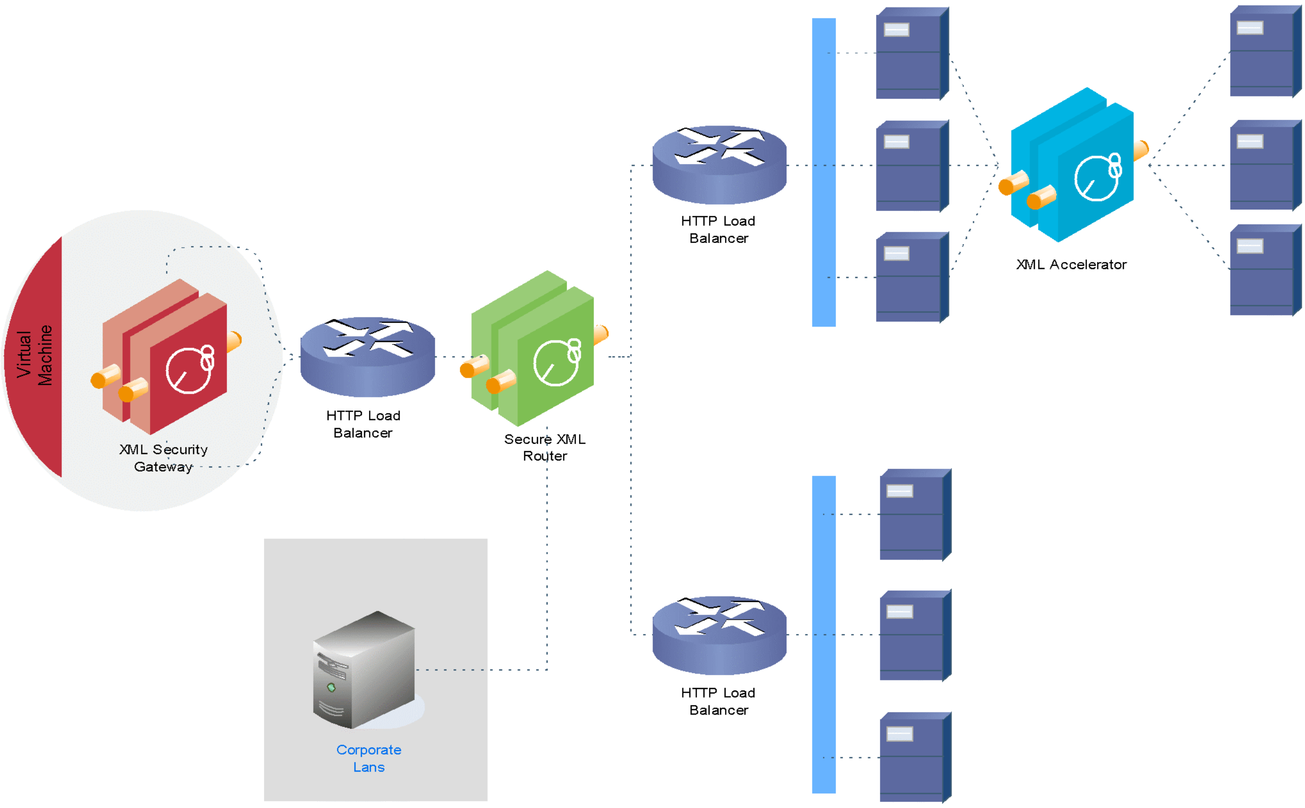

Figure 4: Simulated Network Topology of Panoweaver

IV. Evaluation: Network packet loss

Panoweaver evaluation was carried on one server that considered to be as a multicast to multiple numbers of clients. Normally in the packet transmission over network, the receiver has the capability to know the packet loss by calculating the difference or sequence of RTP sequence number that is present in the RTP communication message format. However, this paper simulated the Panoweaver by indicating the overall rates of the packet loss as a ratio of packet lost during the transmission over the packet

received by the Panoweaver client. The Figure 5 illustrates the comparison of the packet loss in the Panoweaver and Panoweaver -RTCP communication protocol. But it is worth mentioning that although packet has been lost even with the utilization of RTCP feedback report, yet the

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Panoweaver -RTCP will be able to adjust the frame rates of packet again whereas the normal Panoweaver do not.

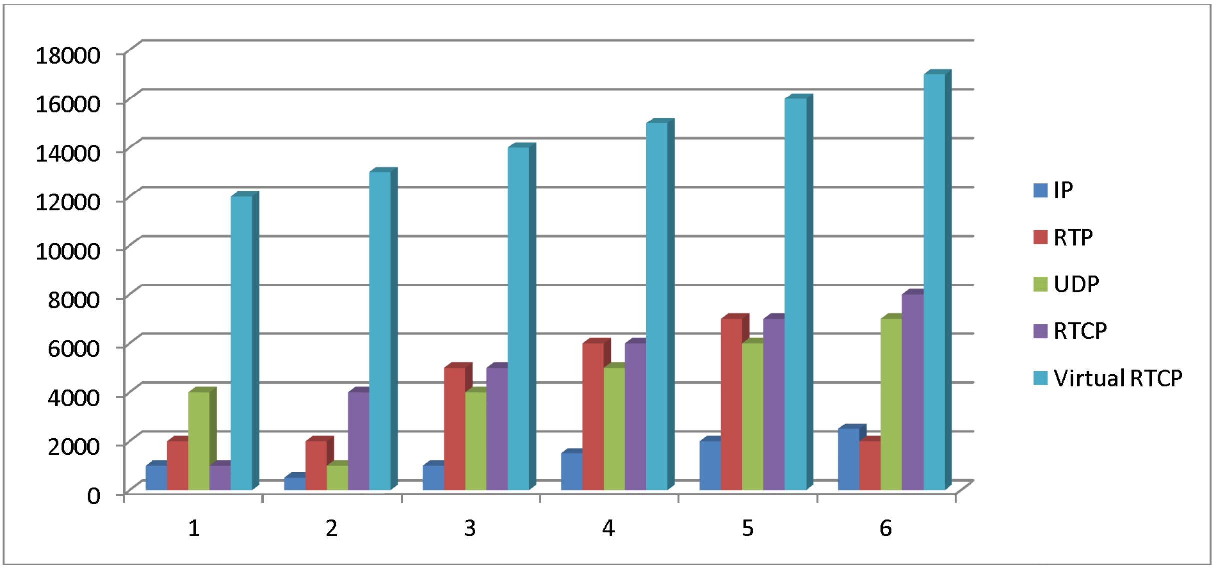

Figure 5: Packet Loss on Increasing Bandwidth

V. Conclusion

This paper was carried out to measure the performance of Panoweaver-RTCP virtual reality software for providing an efficient communication among user network over IP, UDP, and RTP. Decision feedback scheme, which is based on a RTCP, was designed by simulating the Panoweaver based on the packet loss rate. Implemented RTCP on top of UDP, where UDP assists the transmission of real time data, while RTCP provides feedback to sender and receiver about the transmission and reception of the media quality.

The approach of this paper has been justified by presenting a new architectural model on Panoweaver simulation using well suited network simulator NS-2. The finding indicated that Panoweaver gives better performance than the existing Panoweaver where the packet loss is expected.

VI. Acknowledgement

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References

[1] G. Burdea and P. Coiffet, "Virtual reality technology," *Presence: Teleoperators & Virtual Environments*, vol. 12, pp. 663-664, 2003.

[2] T. Grantcharov, *et al.*, "Randomized clinical trial of virtual reality simulation for laparoscopic skills training," *British Journal of Surgery*, vol. 91, pp. 146-150, 2004.

[3] H. Gharavi, *et al.*, "MM06-8 RTCP-based Frame-Synchronized Feedback Control for IP-Video Communications over Multipath Fading Channels," in *International Conference on Communications*, USA 2004, pp. 1512-1516.

[4] Y. J. Liang, *et al.*, "Analysis of packet loss for compressed video: Effect of burst losses and correlation between error frames," *Circuits and Systems for Video Technology, IEEE Transactions on*, vol. 18, pp. 861-874, 2008.

[5] J. Ott and J. Chesterfield, "E. Schooler," RTP Control Protocol (RTCP) Extensions for Single-Source Multicast Sessions with Unicast Feedback," RFC 5760, February2010.

[6] A. R. Reibman, *et al.*, "Quality monitoring of video over a packet network," *Multimedia, IEEE Transactions on*, vol. 6, pp. 327-334, 2004.

[7] V. Novotny and D. Komosny, "Optimization of large-scale RTCP feedback reporting in fixed and mobile networks," in *Third International Conference on Wireless and Mobile Communications*, UK, pp. 85 - 85., 2007, pp. 85-85.

[8] E. I. M. Randa and M. Enugnla, "Enhanced Qos for Real-time Multimedia Delivery over the Wireless Link using RFID Technology," in *IEEE International Symposium* US, pp. 728 – 734., 2006.

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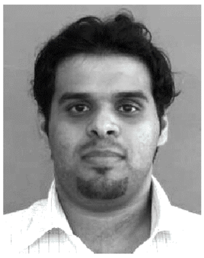

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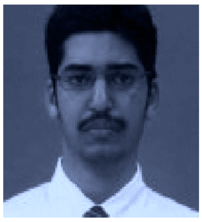

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