

5. Conclusion and Recommendation

In this study supports statistically the superiority of UDP and RTP over TCP in all the scenarios. The good performance of UDP in VoIP applications makes it a preferred transport layer protocol to carry voice packets from source to destination. However, it is likely that RTP may perform better with some modifications/extensions in the protocol as, we observe, its performance is comparable to UDP in most of the cases. We are working towards this goal as such modified RTP will be more promising because it overcomes the shortcomings of both UDP, e.g., traffic control mechanism, and TCP, e.g., head of line blocking.

Depending on the design scenario implementing in NS-2, results are achieved. The QoS factors are captured graphically, and regarding the criteria of low packet loss to kept voice quality high, in general note that TCP beats UDP/RTP for data.

Packet losses weren't obtained for TCP unless the background traffic was at maximum load. Low delay and jitter are higher priority in contrast with loss for UDP/RTP's quality.

TCP's stagnation delay and large jitter is undesirable for VOIP applications. On the contrary, the simple best effort characteristics for RTP allows for very small delay and jitter, with an acceptable voice quality during low/moderate background traffic which makes RTP or UDP the most preferred protocols for VOIP application.

Generally, RTP is always used over UDP because of its identical performance and additional features.

Finally, this implementation concludes that RTP is the best among the transmission protocols.

As a future work, we recommend to compare our topology in real environment instead Simulation programs to analysis fact data on network to get new results.