**ABSTRACT**

Wireless LAN (WLAN) provides flexible data communication systems with the features and benefits of traditional LAN technologies, such as Ethernet and Token Ring without the limitations of wires or cables. The infrastructure of WLANs is dynamic and mobile and it combines data connectivity with user mobility. Practically WLANs provide the final few meters of connectivity between the wired network and the mobile user. IEEE 802.11 standard permits devices to establish either peer –to-peer networks or networks based on fixed Access Points with which, the mobile nodes can communicate. Hence, the standard defines two basic network architectures: the Infrastructure network and the Ad-hoc network. In this thesis, we focus on Infrastructure network, due to its predominant use. IEEE 802.11 has been expanded considerably to include a family of WLAN standards like IEEE 802.11a, b, e, g, s and IEEE 802.11n etc.., These standards operate at different frequency bands, data rates and physical layers to enhance the throughput, Quality of Service (QoS), security and similar kind of parameters. In this thesis, IEEE 802.11b, IEEE 802.11e and IEEE 802.11g standards are used.

Providing QoS guarantees for real time traffic in wireless LAN is the objective of this dissertation. Without QoS, all packets on the network contend for the same pool of resources resulting congestion in the network. Since, the network resources are insufficient, QoS need to be guaranteed, especially for real time multimedia applications. Hence, in this dissertation, quality provisioning issues of intra networking, internetworking and the quality assurance issues related with real time data transfer are addressed to a limited extent. With respect to these issues, the relative literature survey on wireless channel models, channel access schemes, congestion control, routing mechanisms, Voice over IP (VoIP) and video streaming have been conducted and presented.

The approaches generally used in practice to support QoS guarantees in the network are (i) network centric approach and (ii) end to end approach. But the best approach devised is network centric based, in which the routers, switches and base stations of the network are required to provide the demanded QoS support. Hence, this

thesis proposes and analyzes the following the techniques for the quality assurance of real time traffic.

 Analysis of appropriate channel access mechanisms used in the MAC layer and finding the suitable technique for real time data transfer

 proposed a cross layer routing algorithm TLAODV for secured routing in the network layer

 proposed a SNACK-TCP-ECN protocol for congestion control in the transport layer

Moreover, to optimize the application layer quality for real time multimedia traffic, we have

 proposed a deterministic approach for MPEG-4 video streaming

 proposed an integrated model for VoIP transmission under wireless distribution system.

Some of the proposed schemes are also implemented in the hardware by using the network processors, as application modules with constraints to validate the results.

It is well known that, the channel access schemes play an important role in aiding the demand of real time flows to guarantee the QoS requirements. Hence, the support and limitations of basic and adaptive channel access schemes for real time traffic are analyzed and compared for a typically chosen network. From the analysis, it is proved that, the adaptive channel access mechanisms perform better than the basic channel access schemes. This observation is validated, by efficient video transmission on an existing cross layer architecture, by modifying its MAC layer with an adaptive channel access technique. The obtained results ensure that the adaptive techniques yield better results than the basic techniques.

For the upcoming Gbps high-speed network, it is expected to support a wide range of communication – intensive, real time multimedia applications. The requirement for timely delivery of digitized real time information raises new challenges for the next generation integrated- service networks. Among the others, one of the key issues involved is quality enhanced routing. Hence, a wireless routing protocol named Trust established Link Aware Adhoc On demand Distance Vector (TLAODV) protocol based on a cross layer route selection process for wireless multi hop network is proposed. It

eliminates the routes with bad links. The route selection process is based on a cross layer Route Metric, which considers the Frame Transmission Efficiency (FTE) of the MAC layer and the Signal to Noise Ratio (SNR) of the PHY layer. The proposed protocol also incorporates security by establishing trust relationships among the wireless nodes. The performance of the protocol is tested under the delay sensitive voice traffic. Moreover, the routing performance of core and edge routers has also been tested by incorporating the popular security algorithms RSA, AES and firewall type of screening / filtering techniques in the routers by using network processor hardware. The routing performance is observed under varying traffic load conditions along with the incorporated security check up on the router input data. The results clearly indicate that, the routing performance along with the added security will not be affected, whenever the dedicated hardware with parallel process engines is used for routing.

In the transport layer of any reliable network, though the performance of TCP is appreciable in wired environment, it requires adequate improvement in wireless scenario. The performance degradation is mainly because of its inability to distinguish the congestion losses and other types of link losses. Hence to address the issue of loss differentiation, a SNACK – TCP with ECN algorithm is proposed. It is based on the TCP variant TCP-NJ+. The SNACK (Selective Negative Acknowledgement) is incorporated to indicate the multiple packet losses at one time and the Explicit Congestion Control (ECN) is incorporated to forecast the congestion status in the network. Moreover, another issue of using TCP as the transport layer protocol in video streaming applications is also addressed with an Extended TFRC Veno (ETFRCV) algorithm. It is an enhancement of TFRC veno protocol, proposed to meet the special needs of video streaming. It decouples the wireless loss from that of the congestion loss based on the queuing delay of the routing buffer.

Quality of Service support has a profound role in ensuring better audio-visual experience to the end users. The stringent QoS requirements of encoded video make video streaming over wireless links, a challenging problem. The existing approach to provide QoS guarantees in Wireless Local Area Network is through resource reservation mechanisms. A key component in the reservation scheme is to characterize the traffic. Traffic characterization overcomes the difficulty in resource allocation by accurately

specifying the traffic arrivals on a video connection and verifying the resource availability to support traffic at the desired QoS. Moreover, real-time video transmission demands the delivery of video content within a period. The bounds on initial delay and buffer size dictate the performance of the system. Frequent buffer underflows and overflows affect the QoS parameters like delay and throughput. Increasing the initial delay would reduce the occurrence of an outage at the expense of increase in buffer size. Hence, determining the minimum initial delay and the minimum required buffer size is the key to provide a certain guaranteed QoS in wireless environment. In this dissertation, a deterministic approach towards QoS provisioning for MPEG-4 video streaming is proposed. Since the bounds on initial delay and playout buffer size dictate the performance of the video streaming system, the computation of minimum initial delay and optimal playout buffer size becomes necessary. Frame loss, Peak Signal-to-Noise Ratio (PSNR) and Mean Opinion Score (MOS) are computed for different playout buffer sizes to prove that the minimum playout buffer size is sufficient to provide QoS guarantee. This part of the thesis also incorporates an application model for an adaptive buffer management technique that reduces the packet loss at the video player. Here, the maximum buffer size that can be offered at the maximum transmission rate of the video packets is obtained. This application model is implemented using IXP 2400 Network Processor.

Voice over IP (VoIP) is defined as the routing of voice signals over any IP-based network. It is one of the emerging trends feasible for carrying voice and call signaling messages over the Internet by adopting certain standards. In the process of offering QoS to voice traffic, various schemes for improving the voice capacity are investigated previously, and they all require modification in the MAC protocol used by the VoIP stations. But in this dissertation, an integrated model for VoIP transmission over WLAN under Wireless Distribution System (WDS) is proposed. Wireless Distribution System is a wireless multi-hop network that provides quick and easy network setup in open areas, disaster areas and the battle fields. WLAN can be connected to the wired back bone network through WDS, to extend its wireless service coverage by incorporating a number of intermediate Access Points in the network. However, the quality of voice in WDS environment is deteriorated due to the larger transmission delay incurred in the network.

Hence, schemes must be devised / incorporated to reduce this delay. The conventional VoIP transmission in WLAN suffers from the problems of header overhead being larger than the VoIP payload and the quality of VoIP with the back ground traffic is not assured. The proposed model addresses these issues through a novel compression / decompression algorithm and the improved Codec system. Moreover, the thesis also includes an efficient Call Admission Control (CAC) technique with EDCF based MAC layer protocol. It is an application model implemented using network processor, to regulate the traffic at the access points.

Most of the proposed algorithms are simulated using the Network Simulator – NS2. The Integrated Development Environment (IDE) of the Intel network processor along with its hardware has been used for the implementation of certain algorithms. The results achieved in this thesis are encouraging and are applicable to future wireless networks. Future work could be to design and develop the network hardware for the lossless transfer of real time multimedia data in wireless networks.