Abstract

This paper refers to the transmission of speech across data-style networks. This form of transmission is conceptually superior to conventional circuit switched communication in many ways. Quality of service (QoS) is fundamental to the operation of a VoIP network that meets users’ quality expectations. However, the implementation of various security measures can cause a marked deterioration in QoS. These complications range from firewalls delaying or blocking call setups to encryption-produced latency and delay variation (jitter). Because of the time-critical nature of VoIP, and its low tolerance for disruption and packet loss, many security measures implemented in traditional data networks are simply not applicable to VoIP in their current form; firewalls, intrusion detection systems, and other components must be specialized for VoIP. To address the QoS issue of packet loss, this research work develops a new model making use of adaptive buffering technique. Our model implements dual buffers, strategically placed to manage and ameliorate frame erasure problem amongst other QoS factors. With this implementation, the new model achieved a better reduction in the amount of packet loss.

References

Index Terms

Computer Science          Communications

Keywords

Delay  Buffer  Packet Loss  Qos  Voip  Cs/gr  Frame Erasure