Abstract

The ability to transmit voice traffic over conventional data networks has revolutionized the way we communicate. The question is how to provide voice quality comparable to PSTN networks. For this reason, researchers have looked to understand the entire communication components and path between the source and destination. One way is to introduce QoS mechanisms that prioritize voice or latency sensitive data transmissions but most of these techniques also introduce some amount of delay whereas voice traffic is delay sensitive and a second delay can cause degradation in quality. This study describes a means for transmitting voice traffic over conventional data networks by making use of a variable bit rate encoder while anticipating network congestions in real time in order to actively mitigate delays in voice transmissions thereby achieving improved QoS.

References

Global Internet. IEEE Journal on Selected Areas in Communication, Vol 13, no. 8, pg. 1465-1480.

Index Terms

Computer Science  Security

Keywords

VoIP, Quality of Service (QoS), Congestion Control, Link Adaptation, TCP Vegas, AMR Encoder