Comparative Study and Analysis of various VoIP coding Algorithms

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Abstract

Voice over internet protocol (VoIP) is a method of providing phone services over dedicated public IP networks. It allows significant cost savings over traditional Public Switched Telephone Networks (PSTN). Speech quality, as perceived by the users of VoIP telephony, is critically important. Signal quality of the VoIP system is degraded by various network layer problems, which include delay, packet loss and jitter. The implementation of signal through various coding algorithms and digital signal processor can improve the quality of degraded VoIP signal. The present work deals with comparative study and analysis of VoIP Codecs (G.711, G.729, AMR, AMR-WB etc.). The VoIP simulations are conducted for G.711, G.729 and AMR-WB speech coders for different network conditions. The coding algorithms are implemented on VoIP speech signal. The results are validated through the measurement of enhancement signal using perceptual evaluation of speech quality (PESQ)/MOS measurement.

References
9. 3GPP TS 26.171,” Speech codec speech processing functions; Adaptive Multi-Rate - Wideband (AMR-WB) speech codec; General description”, 2008-12.

**Index Terms**

Computer Science  
Algorithms

**Keywords**

VoIP, QoS, G.711, G.729, AMR, AMR-WB.