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Performance Analysis on the Effect of G.729, Speex and GSM Speech Codec on 802.11g Wireless Local Area Network over VoIP using Packet Jitter

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Abstract

In this paper, three speech codecs; G.729 (8 kbps), Speex (8kbps) and GSM (13kbps) were tested together with several predetermined SNR value ranging from 10dB to 45dB with a sample of 8 second speech. VoIP QoS such as packet jitter is analyzed in order to make a comparison of speech quality of those three speech codecs in wireless LAN 802.11g environment. Result shows that at lower SNR, GSM achieve higher packet jitter than that G.729 and Speex. At higher SNR, GSM achieves lower packet jitter as compared than that G.729 and Speex.

Keywords: *Speech codec;G.729; Speex; GSM; VoIP; 802.11g; WLAN, Packet Jitter*

1. Introduction

Voice over IP or VoIP is a term used in IP telephony for a set of facilities that use the Internet Protocol (IP) to deliver voice information. It all started when in February of 1995 by a small company called Vocaltec Inc. [1]. One of the main reason why VoIP became so popular and slowly but surely replacing the traditional public switch telephone network (PSTN) is when PSTN line is being used, we typically pay for time used to a PSTN line company, in other words the more time we stay at phone and more we need to pay. During the "carry over" session, codec is being used. This is the method of how the "audio" data is placed within the UDP datagram. Information about what codec to use is between the systems and is negotiated during the call setup. Some codecs use compression, while others do not. Some standard voice codecs available are ADPCM (Adaptive Differential Pulse Code Modulation), G.711 (A-Law and μ-Law), G.723.1 (pass through), G.729, GSM, iLBC, Linear, LPC-10, and Speex. Today, the installed-based of Wi-Fi client devices exceeds 200 million worldwide in 2007 [2]. IEEE 802.11g offer adequate capacity for supporting Voice over WLAN (VoWLAN) applications. Research have shown that various speech codec gives different speech quality [4-8]. Speech codec which require higher bandwidth have a better quality voice compare to speech codec which require lower bandwidth. Hence, by increasing SNR and better data rate, it gives a better voice quality with respect to any speech codecs available. A key determinant in voice quality is the speech codec, but wireless network performance will have a substantial impact as well.