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Performance Evaluation of The Quality of VoIP Over WLAN Codecs

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***Abstract*— The adoption of Voice over Wireless Local Area Network is on tremendous increase due its ease, non-intrusive, inexpensive deployment, low maintenance cost, universal coverage and basic roaming capabilities. However, deploying Voice over Internet Protocol (VoIP) over Wireless Local Area Network (WLAN) is a challenging task for many network managers, architects, planners, designers and engineers. Voice codec is one of the most critical components of a VoIP system. This work evaluates the effects of various codecs such as G.711, G.723.1, G.729A, G.728, G.726, Adaptive MultiRate (AMR) and Global System for Mobile communication (GSM) codecs on a VoIP over WLAN. Result from simulated network shows that the GSM codec offers the best quality of service for VoIP over WLAN.**

***Index Terms*—Voice over Internet Protocol, Wireless Local Area Network (WLAN), Codecs, throughput, delay**

I. Introduction

Voice over Internet Protocol (VoIP) refers to the transmission of voice over data network [1]. Data networks such as the Internet, Local Area Network (LAN) and Wide Area Network (WAN) are packet-switched technology using internet Protocol (IP) [2]. This does not imply that VoIP is transmitted via the Internet; but that the same Internet Protocol is applied [3]. VoIP often referred to as IP Telephony converges multiple forms of communication such as voice, video and data into a single network. This enables both data and voice to be managed on same network. These multiple forms of IP packetized communication are transmitted over privately managed IP-based network [4].

VoIP has eliminated barriers in international communication, providing an alternative to telephone calls and universal access to cheap calls through the computer and the internet. VoIP networks are very cheap to deploy compared to PSTN which is highly capital intensive. Most calls through the VoIP services are free no matter the distance. VoIP is inexpensive and easy to use, with the ease of upgrading; it is not distance or location dependent; it does not require any extra cablings, a virtual number enables you to make calls from anywhere in the world with an available broadband connection.

A Wireless Local Area Network (WLAN) connects two or more devices over a distance using wireless communication networks such as radio or infrared signal, providing a connection through an access point to the wider Internet. WLANs have become very popular allowing users to move around in a confined area while they are still connected to the network. It provides high speed data communication in small areas such as offices, homes and in commercial buildings. The deployment of Voice over WLAN is easy, inexpensive and non-intrusive, universal coverage; low maintenance cost and has the basic roaming capabilities [6]. The challenges of Wireless LAN however, is that the network is not well equipped to meet the quality of service (QoS) requirements of VoIP.

Data transmitted over the network without being compressed uses a lot bandwidth [7] hence, codecs is required to compress speech prior to being transmitted. The tuning of a codec for a particular type of network is very important [8]. Voice codec is one of the most critical component of a VoIP system. It converts the input speech signal into digital form, transmit the signal to the receiver and reconstruct the original speech signal. This paper evaluates various VoIP codecs such as G.711, G.723.1, G.729A, G.728, G.726, AMR and GSM codecs.

A. Related Works

Codecs are used to compress and encode voice data to enable optimization of bandwidth utilization [7] [8] [9]. Softphones were used to communicate between two parties in [10] using several codecs such as G.711, G.726, G.S.M, G.722 and SPEEX to define which codec selection is suitable to provide better VoIP performance over wireless network.

Simulation method was used in [11] to investigate the performance of VoIP over WLAN using different coding schemes. Voice codecs were employed to investigate VoIP traffic with silence suppression technique where no packets are generated in silence period in [12].

Constant Bit Rate (CBR) traffic were considered in [13][14] with G.711, G.729, G.723.1 codecs; while in [15] G.729 codec with Voice Activity Detection (VAD) enabled was used to produce the variable Bit Rate (VBR) characteristics. The authors in [15] suggest that G.729 codec generates smaller packets and is more error resilient than G.711 [14], hence it is more suitable for use in wireless network where there are higher channel errors. Two codecs G.711 and G.729 were compared over 802.11 Distributed Coordination Function (DCF) protocol in infrastructure mode, resulting in

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the output bit rate of the G.729 encoder being eight times less than that of the G.711 encoder, however, the corresponding increase in capacity is less than 50% when G.729 is used [16].

The header overhead for voice traffic was reduced in [17] to improve quality of service capabilities. Packet headers can also be compressed during multiplexing to increase the bandwidth efficiency [18]. The analytical results of voice capacity with different codecs and packetization intervals compared in [19] matched well with the simulation and measurement results due to the accurate modeling of the CSMA/CA mechanism and collision events.

II VoIP CODECS

Voice codec is one of the most critical components of a VoIP system. It converts the input speech signal into digital form, transmit the signal to the receiver and reconstruct the original speech signal.

The codec samples the waveform at regular intervals and generates a value for the samples. Samples are taken 8000 times/s (8 kHz sampling rate) or 16 000 times/s (16 kHz sampling rate). The algebraic relationships are given below:

$$f_s = 8f\,bps \tag{1}$$

$$T_{SI} = \frac{N_S}{f_I}\,ms \tag{2}$$

$$f_{SI} = \frac{1}{T_{SI}} = \frac{f}{N_S}\,Slps \tag{3}$$

Where,

f is the sampling rate (Bps),

f_S is the (codec) bit rate (bps),

is the (codec) sample size (Bytes),

T_{SI} is the (codec) sample interval (ms),

f_{SI} is the sample intervals per second (Slps).

These values are then quantized in order to map values into discrete-finite value which can be represented using bits, which forms the voice data frame consequently transmitted over the network [7]. Codecs provides compression capabilities to save network bandwidth [20]. High-quality voice codecs requiring very low bandwidths for transmission were designed for the PSTN network. These low bitrates voice codecs standardized by the International Telecommunication Union (ITU-T) as G.711 and G.72X series fall in the class of narrowband codecs.

The major issue with this design is in the ability of the narrowband codec to adequately handle delay, jitter and the packet loss which is associated with IP networks, hence, the need for progressive design of VoIP codecs to address the issue. Increase deployment of VoIP has driven the design and use of new class of codecs such as wideband codecs and adaptive multi-rate codecs. These advancements in the design of VoIP codecs will provide better QoS management capabilities in the IP networks, hence, the future of VoIP promises to provide better services to users than the PSTN. The designed codecs can be categorized into four broad classes depending on the speech coding and transmission techniques used:

- Waveform Codecs
- Source Codecs
- Hybrid Codecs
- Adaptive Multi-rate Codecs

Waveform Codecs

The input speech signal is converted into digital signal and then packetized. Waveform codec produce a reconstructed signal at the receiver as close as possible to the original one thereby reducing the bandwidth requirement. The simplest form of waveform coding is *Pulse Code Modulation (PCM)* based on 8 kHz, 8 bit sampling codec which results in 64 kbps audio codec. Differential Pulse Code Modulation (DPCM) based on predicting the values of the next sample from the previous samples, was further developed to increase the efficiency of PCM.

Source Codecs

Source coding employs a model-based representation of the speech signal. The codec estimates speech signal based on digital model, hence instead of sending the actual waveform, only the parameters of such models are encoded in the bit-stream [7, 21]. Codecs used for source coding are also called *Vocoders*, example of such codec is Linear Predictive Coding (LPC). This can operate at very low bit rates of about 2.4 Kb/s producing a comprehensible speech quality but does not sound normal.

Hybrid Coding

Hybrid codecs are a combination of both waveform and source codecs. The simplest type of codec based on hybrid coding principles is the Analysis-by-Synthesis (AbS) codec which functions by splitting the input speech to be coded into frames about 20ms long. Successive developments resulted in other types of hybrid codecs such as Multi-Pulse Excited (MPE) and Regular Pulse Excited (RPE). They both provide good quality speech at rates of around 10 Kb/s and higher, however, both MPE and RPE cannot operate adequately below 10 Kb/s due to the large amount of information required to transmit to each pulse in a given voice frame. The Code Excited Linear Prediction (CELP) codecs were also developed in contrast to MPE and RPE. It provides better quality than other low bit-rate codecs and is presently the most widely used [wiki]. Further improvements on CELP – based codecs include the low delay CELP, Department of Defence (DoD) and Conjugate-Structure Algebraic –CELP (CS-ACELP). They provide a wide range in bit rate selection between 4.8 Kb/s to 16 K b/s.

Adaptive Multi-rate

The Adaptive Multi-Rate (AMR) audio codec is an audio data compression algorithm developed to work with inaccurate transport channels. The codec is also referred to as a Multi-Rate ACELP (MR-ACELP) codec and is based on the Algebraic CELP (ACELP) technology [22]. It is flexible on bandwidth requirements and has tolerance for bit errors which makes it suitable for wireless links and VoIP services. The codec operates at 8 different bit-rates with a frame size of 20ms and 4 subframes of 5ms. The codec is made up multi-rate narrowband speech codec, which encodes narrowband signals (200 – 3400 Hz) at bit rates ranging from 4.75 to 12.2 kbit/s (equivalent to the GSM EFR codec) and toll quality speech from 7.4 kbit/s (equivalent to the EFR codec) [22, 23].

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Narrowband CODECS

Narrowband codecs are high quality voice codecs which require very low bandwidths for transmission were originally designed for the PSTN network. The codecs which fall in this class of codec use different speech coding techniques offering a wide range of bit rates, coding complexity and quality [7]. Some of these codecs commonly used are G.711, G.72 series such as G.721, G.728, G.723.1, G.729, RPE-based GSM codecs and internet Low Bit-rate Codec (iLBC).

PCM – Based G.711

The most common codec defined by the ITU-T Recommendation is the G.711. It uses the Pulse Code Modulation (PCM) of voice frequencies at a standard bit rate of 64 Kb/s and non- standard bit rates of 56 Kb/s and 48Kb/s [7]. It is analyzed by two algorithms: *µ-law* and *A-law*. The µ-law algorithm is used where the input variable x is captured with 14 bits of uniform quantification, and transformed with a memoryless function *f(x)* that reduces the distortion error for speech as shown below [21][24];

$$f(x) = A\frac{ln\left(1+\mu\frac{|x|}{A}\right)}{ln(1+\mu)}sgn(x), |x| <= A \tag{4}$$

$$f^{-1}(y) = \frac{A}{\mu}\left[exp\left(\frac{ln(1+\mu).|y|}{A}\right) - 1\right]sgn(y), |y| <= A \tag{5}$$

where A is the input magnitude's peak and µ is a compression control degree. This codec is still widely used because of its simplicity, excellent voice quality and low delay [7, 24].

G.72x Series

The most widely used of the G.72series codecs are the G.723 and G.729 codecs. The Adaptive Differential PCM (ADPCM) based G.721 operating at 32k b/s was standardized in the mid – 1980s to enable reconstruction of speech. Furthermore, G.726 and G.727 codecs were recommended to convert the 64kb/s µ-law or A-law PCM channel to and from 40, 32, 24 and 16 kb/s channel [7, 25]. The low-delay CELP codec G.728 operating at 16 kb/s with a delay of less than 2ms standardized with speech quality as good as or even better than G.721 and has a good robustness to channel errors.

The G.723.1 codec based on the DoD CELP was standardized by the ITU-T in 1991 to operate at two bit rate either 5.2Kb/s or 6.3Kb/s and can switch between the two rates at the frames boundary. The 6.3 Kb/s bit –rate version uses a 24 byte frame, while the 5.2 Kb/s version uses 20-byte frames. The codec compresses voice audio in 30ms frames with look-ahead time of 7.5ms for frame construction which results in a total delay of 37.5ms in generating voice sample. The G.729 codec based on Conjugate-Structure Algebraic-Code-Excited Linear Prediction (CS-ACELP) is referred to as G.729 annex A (G.729a) has lower complexity and consequently lower algorithmic delay and operates on a 10ms frames with 5ms look-ahead delay which result in total algorithmic delay of 15ms.

RPE-Based GSM Codec

The digital mobile radio system Global System for Mobile communications' (GSM) used all over the world is a full rate speech codec operating at 13 kbit/s and uses the RPE codec. The GSM codec provides good-quality speech. The speech input is a 16 bit word sampled at 8 Khz is analyzed by the LP

The Internet Low Bit-rate Codec (iLBC)

The internet Low Bit-rate Codec specifically designed for VoIP application is a royalty free narrow band codec developed by Global IP Sounds (GIPS). The codec based on a block-based Linear Predictive Coding algorithm enables graceful speech quality degradation in the case of lost frames, which occurs in a connection with lost or delayed IP packets [7]. It operates at 13.33 Kb/s with an encoding frame length of 30ms and at 15.20 kbps with an encoding frame length of 20ms. The iLBC codec is popularly used by software such as Skype, Google talk and Gizmo Project.

Wideband And Multi-rate Codecs

Wideband codecs such as Adaptive Multi-rate (AMR-WB), G.722.1 and Speex are popularly used in the deployment of VoIP over current broadband access networks where bandwidth is high. Some wideband codecs used in VoIP software are extensions of the narrowband speech coding techniques with a higher sampling rate. The wideband codecs have a higher sampling rate of 16 kHz, hence provides a better sound quality. The most popular class of wideband codecs also comes with multi-rate adaptation by providing both low bit rate and high bit rate transmission thereby, ensuring applicability to any underlying network condition [7]. These types of codec also eliminate the need for transcoding when the voice is routed from a high bandwidth network to a low bandwidth network.

Adaptive Multi-rate Wideband (AMR-WB)

Adaptive Multi-rate Wideband (AMR-WB) standardized as ITU-T G.722.2 codec was jointly developed by VoiceAge and Nokia for the next generation packet-switched wireless network. The codec based on the ACELP coding technique is designed specifically for packet-switched networks by providing robustness to packet loss. The AMR-WB codec can operate in 8 different modes with bit rates of 4.75, 5.15, 5.90, 6.70, 7.40, 7.95, 10.2 or 12.2 kbps [26] and results in 20ms frame size with 5ms look-head resulting in a total of 25ms packetization delay.

G.722.1

G.722.1 a digital wideband codec algorithm operating at a bit rate of 24 kbps or 32 kbps, provides an audio bandwidth of 50 Hz to 7 kHz. It operates on 20-ms frames (320 samples) of audio with look-ahead time of 20 ms and frame size of 480 bits for 24 Kbps and 640 bits for 32 Kbps. In G.722.1 Annex C the sampling rate doubles from 16 to 32 kHz, and it can generate three different packets sizes; 480 bit, 640 bit, and 960 bit over the same frame duration. Hence, the data rate also changes based on the packet size [27].

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Speex

Speex based on CELP was designed specifically for VoIP over broadband connections. The codec was designed to be robust to packet loss, would allow both very good-quality speech and support multiple bit rates. It supports ultra-wideband (32 kHz), wideband (16 kHz sampling rate) in addition to narrowband (8 kHz sampling rate). Speex also supports variable bit rate (VBR) encoding and Voice Activity Detection (VAD) and can provide a wide range of bit rates starting from 2 kb/s to 44 kb/s hence, able to adapt to available bandwidth.

III SIMULATIONS AND DISCUSSIONS

This work used Riverbed Modeler Academic Edition (OPNET Modeler) simulation tool to design, model, simulate and analyze the VoIP over WLAN [8]. Six scenarios of the same node models were simulated using six different voice codecs. Figure 1 represents the simulated wireless network scenario configured with TCP and UDP protocols. The scenario of the same node models were simulated using seven different voice codecs. The effect of various encoder schemes with Speech Activity Factor enabled or disabled on the load and throughput were investigated.

Encoder related parameters such as encoder name, frame size, lookahead size, DSP Processing Ratio, coding rate, Speech Activity Detection were set. All outgoing and incoming call use these encoder schemes attributes mentioned above. The nodes maintain Application Layer related parameters that can be used by all the nodes in the network. This node avoids duplication of parameters in multiple nodes.

Table 1 below gives a summary of voice encoder scheme. The voice traffic for each scenario can be calculated for each codec.

Assume that G.711 is used as the encoder scheme. Its parameters are:

Frame Size: 4msec
Look-ahead Size: 0msec
DSP Ratio: 1:0
Coding Rate: 64000bits/sec
Number of Frames per Packet: 1

dsp_time = DSP Ratio * Frame Size = 4 msec
steady state packet inter-arrival time = dsp_time
= 4 msec

Number bytes/packet = number of frames per packet * coding rate * Frame Size
= 1*64000 * 4 msec
= 32bytes/pkt

Average Traffic Sent (packets/sec) = 1/4msec
= 250 pkt/sec

Average Traffic Sent (bytes/sec) = 32*250
= 8000bytes/sec

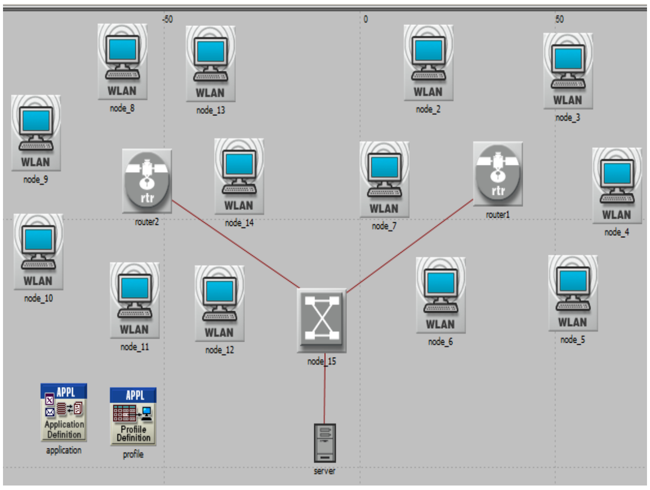

Figure 1: Simulated VoIP over WLAN Scenario

Table1: Voice Encoder Schemes

Codec Type	Name	Frame Size (secs)	Lookahead Size (secs)	DSP Processing Ratio	Coding Rate (bits/sec)	Speech Activity Detection
PCM	G.711 (silence)	10 msec	0 msec	1.0	64 Kbps	Enabled
PCM	G.711	10 msec	0 msec	1.0	64 Kbps	Disabled
ACELP	G.723.1 5.3K	30 msec	7.5 msec	1.0	5.3 Kbps	Disabled
ACELP	G.723.1 5.3K (silence)	30 msec	7.5 msec	1.0	5.3 Kbps	Enabled
RPE-LTP	GSM FR	20 msec	0 msec	1.0	13 Kbps	Disabled
RPE-LTP	GSM FR (silence)	20 msec	0 msec	1.0	13 Kbps	Enabled
ADPCM	G.726 16K	10 msec	0 msec	1.0	16 Kbps	Disabled
ADPCM	G.726 16K (silence)	10 msec	0 msec	1.0	16 Kbps	Enabled
LD-CELP	G.728 12.8K	10 msec	0 msec	1.0	12.8 Kbps	Disabled
LD-CELP	G.728 12.8K (silence)	10 msec	0 msec	1.0	12.8 Kbps	Enabled
CS-ACELP	G.729 A	10 msec	5 msec	1.0	8 Kbps	Disabled
CS-ACELP	G.729 A (silence)	10 msec	5 msec	1.0	8 Kbps	Enabled
ACELP	AMR 4.75K	20 msec	5 msec	1.0	4.75 Kbps	Disabled
ACELP	AMR 4.75K (silence)	20 msec	5 msec	1.0	4.75 Kbps	Enabled

A. Scenario

In the scenario shown in fig. 1 above, the VoIP over WLAN network was simulated using three most popular codecs; G.711, G.729A, and GSM codecs at different times. The delay incurred by voice application packets, while going from a calling party to called party and vice-versa were measured for each encoder. The throughput which refers to the total number of bits (bits/sec) forwarded from the wireless LAN layers to higher layers in all nodes of the network was also analyzed.

B. Result and Analysis

The result of the simulation in fig 2, indicates that the GSM codec has the highest throughput of 20,000 bits/sec, the G.711 has a throughput of 2500 bits/sec which is about 31% of the data expected from the calculation above and G.729 has the lowest value. The GSM codec however has the highest end-to-end delay of about 140ms in fig 3 which is an acceptable value compared to the theoretical acceptable variation [5] [6] [28] [11].

The simulated result in fig. 2 and 3 indicates that the GSM codec will produce best effort network for the VoIP over WLAN giving that it has the highest throughput. The

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advantage of GSM codec over other low rate codecs is its relative simplicity; hence it provides good-quality speech [6]. However, the end-to-end delay is highest compared to G.711 and G.729A codecs, hence, the AMR codecs were standardized as an improvement of the GSM codecs. The end-to-end delay and throughput of the AMR as observed in fig 4 and 5 is an improvement to the GSM codec. It is also refered to as the GSM-AMR codec. The AMR codec will forward more bytes per seconds when compared to the GSM codec.

Comparisons in fig 5, also indicate that the LD-CELP (G.728) codec has a higher throughput than the ACELP and ADPCM (G.726) codecs. The ACELP and ADPCM codecs however has lower end-to-end delay compared to the GSM-AMR, PCM, LD-ACELP codecs as observed in fig 2,3,4 and 5. Result in fig 5, shows that the PCM has the highest throughput of 2400bits/s and a delay of 72ms as observed in fig 2 and fig 3 respectively, whereas ACELP and ADPCM are seen to complement each other with throughput of 650bits/s. In fig 5, the LD-CELP (G.728) codec also presents the highest throughput of about 2400bits/s and a delay of 70ms compared to CS-ACELP (G.729A) and ACELP. CS-ACELP however, produces lower delay compared to LD-CELP according to the simulated wireless network.

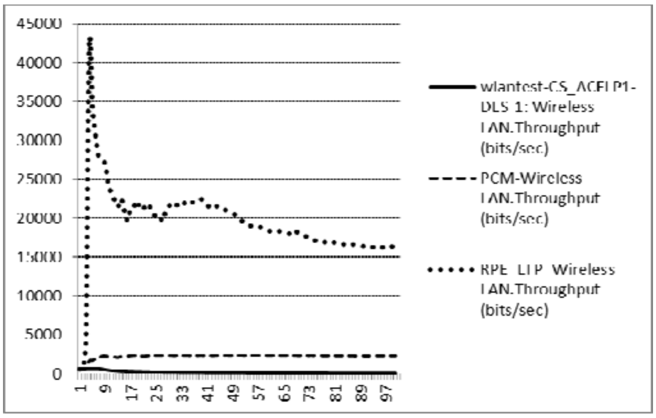

Figure 2: Throughput of CS-ACELP, PCM, GSM Codecs per second

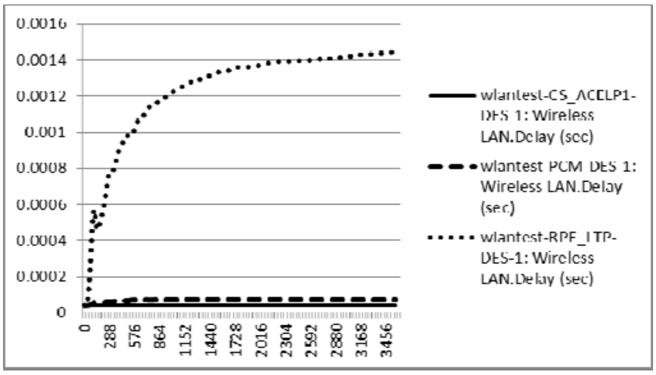

Figure 3: CS-ACELP, PCM, GSM Codecs Delay per second

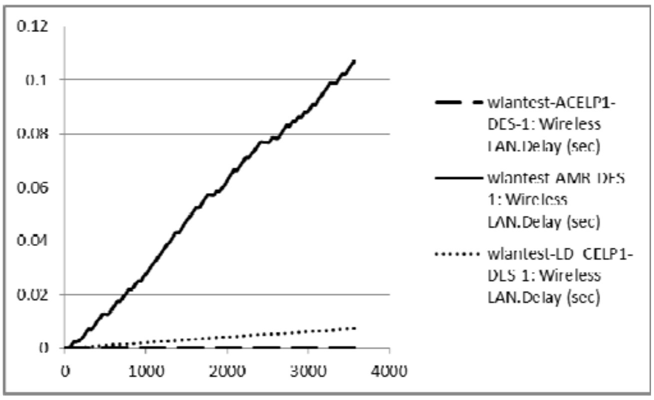

Figure 4: ACELP, AMR, LD-ACELP Codecs Delay per second

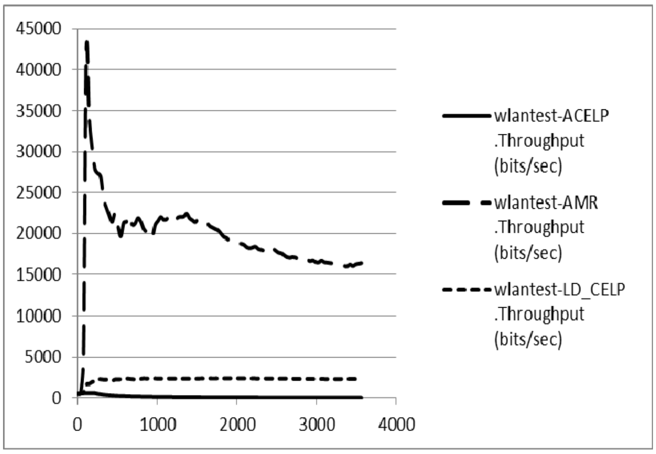

Figure 5: ACELP, AMR, LD-ACELP Codecs Delay per second

IV CONCLUSION AND FUTURE WORK

In this work, we successfully evaluated the effects of various codecs such as G.711, G.723.1, G.729A, G.728, G.726, AMR and GSM codecs on a voice over internet protocol deployed over a wireless local area network. Our result from the simulated network shows that the GSM-AMR codec will give the best-effort quality of service for VoIP over WLAN. The GSM-AMR codec is a digital mobile radio system which provides robust high quality speech together with the flexibility to deliver radio network capacity enhancements by means of low bit-rate operation. Further work should be done on reducing the end-to-end delay variations to enable the codec provide an excellent quality of service.

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