Performance Evaluation of IMS Multimedia over WiMAX

Francis M. Masuabi, Denys Vera, Neco Ventura
Department of Electrical Engineering
University of Cape Town
Rondebosch, 7700
+2721 650 5296
{masuabifm, dvera, neco}@crg.ee.uct.ac.za

Abstract*-Telecommunication networks have evolved steadily to meet the ever increasing demand of users. Hence, the Internet Protocol Multimedia Subsystem (IMS) will make it possible to provide ubiquitous computing to the end-user. WiMAX is based on the IEEE 802.16 standard engineered to provide wireless broadband access, but the main focus of this study, is the use of fixed WiMAX as a backhaul for VoIP and IPTV. A performance evaluation of VoIP and IPTV quality of service was done over the WiMAX network. We examined the performance of these services over various codecs, analyzing the key performance metrics such as delay, latency, jitter and packet loss. With respect to VoIP sessions, the quality of the service was measured using the ITU-T E Model. Our results showed that the G.711 (PCMA and PCMU) was the better VoIP codec according to the E-Model and MPEG2 for IPTV.

Index Terms — **IMS, IPTV, QoS, VoIP, WiMAX.**

I. INTRODUCTION

Worldwide Interoperability for Microwave Access, (WiMAX) is an access network which provides wireless transmission of data in various ways, such as point-to-point links or full mobile cellular-type access. The technology is based on the IEEE 802.16 standard, which is a standards-based technology enabling the delivery of last mile wireless broadband access as an alternative to cable and DSL. WiMAX supports a variety of applications with varied Quality of service (QoS) parameters. The IEEE 802.16 standard groups' services into five classes: unsolicited grant service, real-time polling service, extended real- time polling service, non-real time polling service and best effort. Internet Protocol Television (IPTV) and Voice over IP (VoIP) were investigated using an ideal QoS profile over the WiMAX network.

This work was supported in part by Telkom SA, Nokia Siemens, the National Research Foundation and the Department of Trade and Industry.

IPTV is when digital television is delivered using the IP protocol over a network infrastructure. It covers both live TV (broadcasting) as well as stored video (Video on Demand). The playback of IPTV requires either a personal computer or a "set-top box" connected to a TV. Video content is typically compressed and then sent in a transport stream delivered via Internet Protocol (IP) Multicast in case of live TV or IP Unicast in case of Video on Demand. VoIP on the other hand is voice traffic transported over the Internet. Unlike circuit switched PSTN telephones, VoIP uses packet switching which is more bandwidth efficient. Its effectiveness is due to the compression algorithms used.

PSTNs outperform wireless networks in the deployment of VoIP, and customers expect VoIP to meet and exceed the standard of quality offered by PSTN. Providing IP solutions that correspond with the strict requirements of high quality and reliable voice communication establishes a non-trivial challenge. Delays of up to 100-ms and above are detectable by humans and can impair the interactivity of conversations. Humans are far less tolerant to audio degradation than video degradation by comparison, thus to meet the requirements, it is crucial to minimize the network latency and packet loss as much as possible. Some of the factors that add to the delay are forwarding and queuing delays at the network layer and at the application layer, packetization/depacketization and algorithmic delays are encountered [6]. Packet loss is another factor which impairs voice signals and as a result retransmission cannot be regarded as an option while transmitting voice because of its real-time requirements. Subsequently, service providers have to ensure that delay and packet loss are restrained in order to provide voice qualities of high standards.

Codecs and signaling protocols used for voice applications are important, as they also affect the quality of service performance and need to be taken into account when evaluations are done. But with regard to IPTV, only an optimum amount of bandwidth is needed to withstand the effects of degradation. This paper is organized as follows. Section 2 compares our work with previously published work. Then, section 3 provides the testbed architecture and background. Section 4 provides the VoIP performance metrics where the mean option score is explained. Section 5 and 6 provide procedures used and results obtained. Finally, section 7 concludes this paper.

II. RELATED WORK

Performance evaluation of multimedia is an important facet in communications as the standard of quality always needs to be enhanced. There are various publications closely related to ours, such as the work done by N. Scalabrino et. al [7]. They report results from a fixed Alvarion BreezeMax WiMAX testbed. The voice quality was evaluated using the E-Model which is a tactic of quantifying the quality of a signal. In their test scenario, the G.729 and G.723.1 codecs were considered. The measurements were based on the downlink which was a bottleneck. Having saturated the network with VoIP traffic, they discovered that the G.723.1 codec outperformed the G.729 as a result of the delay saturating after 300 *ms*. From this paper it was seen that the type of codec used during the transmission of VoIP does affect the quality of service.

A. Rumin et. al. [1] detected how many VoIP calls a wireless LAN network can sustain without the performance deteriorating. Two WLAN architectural structures were analysed using the IEEE 802.11b standard. In order to discover the maximum number of voice calls, quality of service parameters involved in real time communications were analysed. Packet loss, jitter and packet loss rate were taken into account. A maximum of 10 VoIP calls could be sustained without degradation. With additional calls, the average delay and jitter escalated to unacceptable values. For ad-hoc networks in reality, results are inconclusive as the performance depends on the exact routing used. This paper took into account the real time metrics, however did not measure the quality of the signals.

III. TESTBED AND BACKGROUND

Fig. 1 illustrates the equipment used to carry out the experimental study for the performance evaluation. The testbed consists of a BreezeMax Micro Base Station (μBST), two subscriber units (SU): the BreexeMax Self Install and the BreezeMax PRO. The testbed uses Frequency Division Duplexing in the 3.5 GHz frequency band. The air interface was emulated over wire using Radio Frequency (RF) cables.

As shown in Fig. 1, two Linux machines (ubuntu 8.04) were used as clients to establish media sessions. When a session is set up i.e. as client 1 makes a call, traffic moves via the SU (CPE-Si) to the μBST to the UCT IMS core, then traverses back using the same path but via the CPE-PRO to client 2 or vice versa. The Monitor PC is used to monitor the conditions of the CPE PRO which comprises of an Outdoor Unit (ODU), an Indoor Unit (IDU) and the self installed CPE Si.

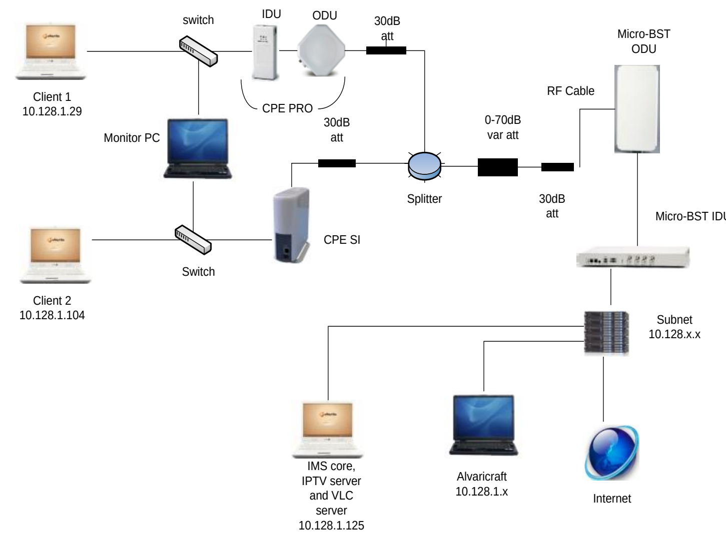

Fig. 1 WiMAX Testbed

The SUs differ as their power budgets vary by 10dB. While the CPE-Si functionality is inadequate at very poor channel conditions (close to the Received Signal Strength Indicator (RSSI) sensitivity threshold), the CPE-PRO is functional with acceptable performance.

The software used in the performance evaluation included the IMS core, UCT IMS client, Alvaricraft and Wireshark. The core was used to register the clients to the IMS. This made it possible to run the UCT IPTV advanced server. The IPTV advanced server is a redirectional server which points to the VLC media streaming server. Alvarion's AlvariCRAFT is a SNMP (Simple Network Management Protocol) application designed for online management of BreezeMAX system components [2]. It provides a guideline in the installation and management of system components. By using AlvariCRAFT, one can easily configure the base station and subscriber units as required while software upgrades can be done on the system. Wireshark, a free Open Source network protocol analyzer was used to monitor traffic between interfaces. This tool was used to measure the performance metrics when traffic traversed from one interface to another.

IV. VOIP PERFORMANCE METRICS

For the purpose of this study we consider delay, jitter, packet loss and the Mean Opinion Score (MOS) as performance metrics. Mean Opinion Score best described by the ITU-T P.800 is the most common measure used for voice quality. It's a subjective method of analyzing speech quality on a scale of 1 to 5, where 1 is the worst and 5 the best. However for this paper, the MOS was obtained from an objective measure called the E-Model. This model provides a computational measure for predictive analysis. It measures the transmission impairments directly and predicts the voice quality with respect to those impairments. Giving an estimation of how an average user would likely rate a call on a MOS scale. Calculation of the transmission rating factor R is as follows:

$$R = R_0 - ls - id - ie + A \qquad (1)$$

where R_0 is the signal- to -noise ratio, ls is a combination of all impairments which occur more or less simultaneously with the voice signal, id represents the impairments caused by delay, ie represents impairments caused by low bit rate codecs and the advantage factor A allows for compensation of impairment factors when there are other advantages of access to the user. However in practice, ITU-T proposed a simplified version of equation (1). The simplified algorithm considers that noise cancellation is encountered in the network and also ignores the advantage factor. The simplified algorithm is shown in equation (2).

$$R = 93.4 - I_d(d) - I_e(codec,loss) \qquad (2)$$

Where I_d is a function of the mouth-to-ear absolute one-way delay (d) and I_e is a function of the used codec and packet loss rate. I_d is a function of delay as shown in equation (3).

$$I_d(d) = 0.024d + 0.11(d - 177.3)H(d - 177.3) \qquad (3)$$

Where H(x) is a step function. The codec impairment factors for the codecs used in this study are shown in Table I.

TABLE I
CODECS WITH IMPAIRMENT FACTORS

Codec	Bit-rate (kbps)	Codec impairment factor (I_e)
PCMA	64	0
PCMU	64	0
GSM	13.6	20

The R rating can be transformed into an equivalent MOS value with the following equation.

$$MOS = 1 + 0.035R + 7 \cdot 10^{-6} R(R - 60)(100 - R) \qquad (3)$$

Although the R rating can be translated into a MOS value, it cannot predict the absolute opinion of an individual.

V. METHODOLOGY

A. Channel conditions

The channel is configured to three levels i.e. best, intermediate and poor quality depending on the value of the step attenuator. The variable range for the step-attenuator is between 20 to 40 dB. A summary of the channel conditions is given in Table II. At poor conditions the CPE- Si ceased to function, thus no parameters were recorded.

TABLE II
CHANNEL CONDITIONS

	Best Quality		Intermediate Quality		Poor Quality
	Client 1 (CPE PRO)	Client 2 (CPE Si)	Client 1 (CPE PRO)	Client 2 (CPE Si)	Client 1 (CPE PRO)
Bandwidth (MHz)	3.5	3.5	3.5	3.5	3.5
Uplink Tx Frequency (MHz)	3454.750	3451.750	3454.750	3451.750	3454.750
Tx Power (dBm)	19.44	21.63	20	21.63	20
Uplink Current Rate	QAM 64 ¾	QAM 64 ¾	QAM 16 ¼	QPSK ¼	BPSK ½
Uplink SNR (dB)	26.30	20.90	17.20	10.90	7.40
Uplink RSSI (dBm)	-76.50	-82.30	-86.50	-93.20	-96.10
Downlink Current Rate	QAM 64 ¾	QAM 64 ¾	QAM 16 ¼	BPSK ¼	BPSK ¼
Downlink SNR (dB)	27	22	18	12	8
Downlink RSSI (dBm)	-74	-83	-85	-93	-94

B. Test Cases

Various test cases were used for IPTV and VoIP to evaluate the performance of multimedia over WiMAX.
With regard to IPTV, a test video was encoded using three different codecs namely MPEG-1, MPEG-2 and MPEG-4. The codecs all have a bit rate of 1024 bps. The video was played for 60 seconds using the UCT IMS client and results were obtained using Wireshark. The various IPTV test cases that were considered in the study are:

- **Case 1:** Investigate how video codecs will perform using different QoS profiles offered by WiMAX. This will show which of the codecs is best for IPTV and also the ideal QoS profile for IPTV.
- **Case 2:** Investigate how many video streams a subscriber unit can sustain while still having good video sessions without packet loss. This is done by increasing the number of streams to the subscriber unit until the videos deteriorate. Case 1 will show which QoS profile and codec is most suited for IPTV and the results thereof will be used for this case.

With respect to audio experiments, a VoIP session was setup using the UCT IMS client as specified in section 3. The codecs supported by the client are G.711 (PCMU and PCMA) and GSM. These codecs were compared to examine which codec best suites VoIP. As such, the best QoS profile for VoIP proved to be Continuous grant (AKA Unsolicited Grant Service) and therefore used for all test cases.

- **Case 1:** A VoIP call was established over the poor channel conditions, under codecs supported by the UCT IMS client. Delay, mean jitter (variation of delay) and packet loss were measured for each codec respectively.
- **Case 2:** WiMAX network was saturated by running three videos on three different clients, a VoIP call and Internet downloads in the background under intermediate circumstances. The saturation is used to determine the effect WiMAX has on the quality of a VoIP call. Voice quality metrics: Delay, mean jitter and packet loss were measured for each codec. An intermediate channel is used under the assumption that it replicates the environment.
- **Case 3:** For each audio codec, the R-Factor was calculated and converted into the mean opinion score to obtain an estimate of the quality as perceived by humans. This will be done on an intermediate channel condition.

VI. RESULTS

A. IPTV Test Cases

The results show that in all the different QoS profiles at low Maximum Information Rate (MIR), Committed Information Rate (CIR) or at small packet sizes for CG, the maximum delta is low. At these values the video does not play. There is insufficient bandwidth making the packets losses high. With an increase in MIR or CIR or packet size the maximum delta also increases. In all the QoS profiles, the maximum delta (delay) increases to values of around 40 milliseconds. For all codecs, MPEG-2 has the least maximum delay followed by the MPEG-1 and then MPEG-4, this is true for all QoS profiles. Figures 2 and 3 show the results of Best Effort and Real-Time QoS profiles respectively.

In terms of Packet Loss, the results show that at the value of 1400 of CIR or MIR, the packet losses decline as the video continues to play. This may be due to the fact that more bandwidth is required for setting up the session. It was observed that MPEG-1 and MPEG-2 packet losses are approximately the same with a slight difference when the CIR or MIR is 1500 kbps. At this value, MPEG-2 experienced no losses while MPEG-1 did. MPEG-4 had the highest packet losses with all the QoS profiles for the video packets. This was an unexpected result which occurred due to the rate control in MPEG-4 not being as effective when compared to the other codecs. Even though the bit rate of the codec was set to1024 kbps, the actual bit rate fluctuated. Significant upward deviation of the rate from 1024 kbps causes more packet losses.

In the second case, the goal is to investigate the number of videos that can be streamed to the subscriber unit, while maintaining a good video session without

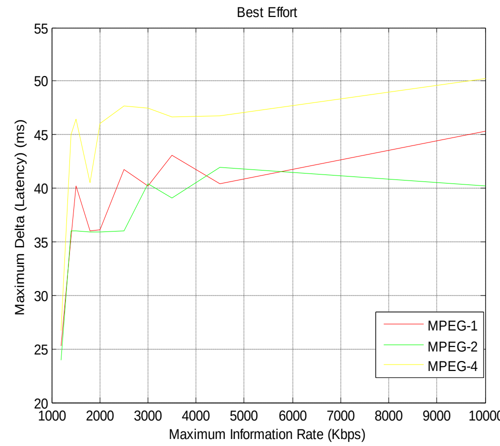

Fig. 2 Maximum Delta using BE

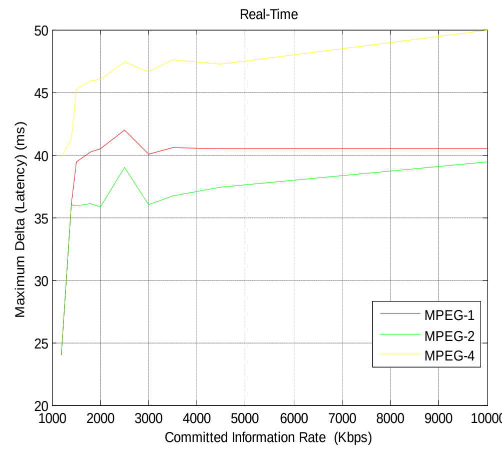

Fig. 3 Maximum Delta Real-Time

significant packet losses. Case 1 showed that Real-Time QoS profile was ideal for IPTV and that MPEG-2 was the better codec compared to the others. The test was done using MPEG-2 and the Real-Time QoS profile. It was found that when values of CIR less than 1200 kbps were used, the packet loss was high and the video would not play. This was expected as all the videos were encoded at a rate of 1024 kbps. In general, when a video is sent to a client, there is packetization overhead that accounts for the extra bits. As a result, all the measurements were taken from 2000 kbps onwards. Between 1200 and 2000 kbps minimal packet losses were incurred. The video played but was choppy. Table III and IV shows the CIR and the number of streams that can be streamed simultaneously at different channel conditions. It was seen that for poor channel conditions, only one stream could be played even when using the largest CIR value.

Table III shows that seven was the maximum number of streams that could be streamed using the largest value of CIR. The results show that the minimum CIR value for two streams is 3000 Kbps. This means that for a household to have two TV sets the value of CIR for the RT QoS profile must not fall under 3000 Kbps.

TABLE III

NUMBER OF STREAMS FOR BEST CHANNEL

CIR (kbps)	Number of Streams
2000	1
3000	2
5000	3
6000	4
8000	6
12000	7

TABLE IV

NUMBER OF STREAMS FOR INTERMEDIATE CHANNEL

CIR (kbps)	Number of Streams
2000	1
3000	2
6000	4
8000	5
12000	5

As expected fewer streams are transmitted in the intermediate condition as compared to the best channel, which results with a maximum of 5 streams as seen in Table IV.

B. VoIP Test Cases

With regard to jitter, in best and intermediate conditions as seen in Fig. 4, the jitter becomes constant at *20 ms*, but in poor conditions the jitter settles in *5, 8, 12 ms* for PCMU, PCMA and GSM respectively which is less than in intermediate conditions. One would expect the jitter in poor conditions to be far more than *20 ms* since the packet loss is recorded to be 36.46%. Given that the packet loss is so high, the jitter in principle ought to be high as well, because packet loss is typically also induced by jitter. When packets at the receiver are out of sequence, in which the packets sent have changed during transmission typically means that jitter had an effect. All the jitter plots settle to a constant value at some point; this means that fixed delay is added to each packet. Figures 5 and 6 present delay of PCMA and GSM under poor conditions respectively. PCMA closely resembles PCMU but the gradient is slightly higher with 0.6254. GSM performs better with a gradient of 0.05045 which is considerably lower than G.711. There are a couple of wild points which possibly represent packets lost during transmission which degrade the quality of the call. Fig.7 presents the average jitter as the packet sizes are increased. The jitter of PCMU and PCMA steeply increases to *20 ms* but eventually settle at *21 ms* when the packet size reaches 448 bytes. GSM on the other hand, initially decreases and settles just below PCMU and PCMA proving to be the better codec in this instance.

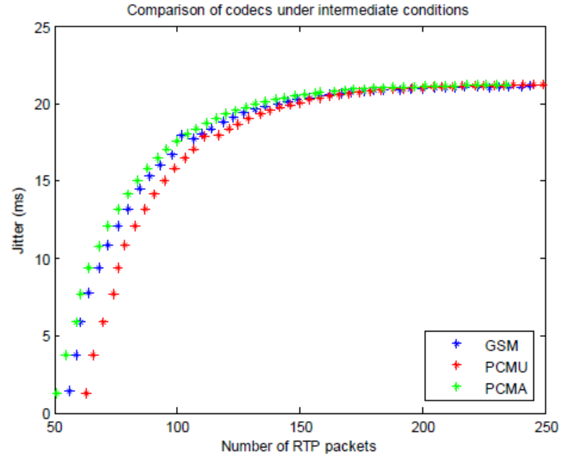

Fig. 4 Jitter plot for all codecs

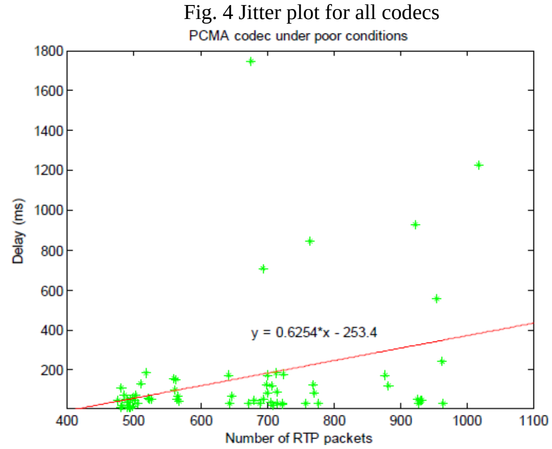

Fig. 5 Delay plot of PCMA under poor conditions

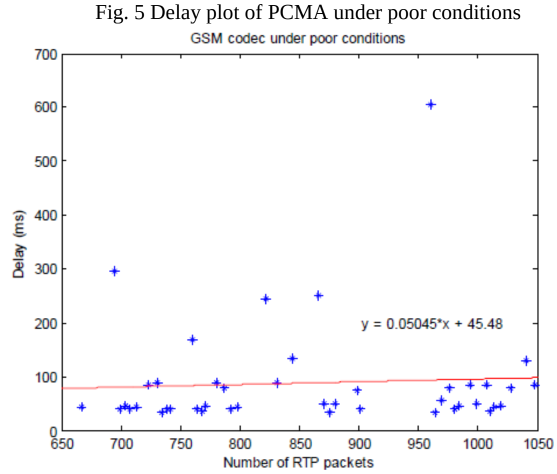

Fig 6. Delay plot GSM under poor conditions

To emulate traffic congestion, the network was stress tested by playing 10 videos using VLC and a call was simultaneously generated using IMS. Delay, jitter and packet loss values of the call in Table V are undesirable. These intolerable values could be due to the bandwidth allocation from the network. When various multimedia is transmitted concurrently, the network bandwidth should ideally be divided in such a way that all media maintains high quality.

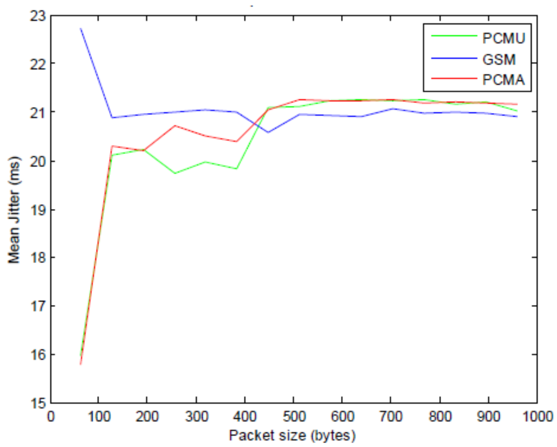

Fig. 7 Packet size vs. Jitter

A video clip demands more bandwidth than a VoIP call, thus with 10 videos playing, VoIP was given the least bandwidth which is reflected in Table V. It must be noted that as the number of videos increase, the picture quality of the videos was bad and the packet loss was extremely obvious.

With reference to the E-Model, the performance of G.711 has better quality than GSM as a result of G.711 having a higher bit rate and simplified compression algorithm. GSM which has a low bit rate uses a compression scheme which may utilize less bandwidth and probably can run more simultaneous calls but at a consequence of reducing clarity, introducing delay and making voice quality more susceptible to packet loss. The results do signify that G.711 has higher MOS values in both intermediate and poor channels thus having better voice quality compared to GSM. The lossy impairment factor of 20 is the major contribution of GSM performing worse than G.711. The MOS values calculated under poor WiMAX conditions are very optimistic. Since when a conversation was held, the quality was dreadful, but according to the measurement, the quality is in the fair to good range.

VII. CONCLUSION

WiMAX is designed with QoS profiles which enable the ISP to classify services into categories depending on their priority. Continuous Grant proved to be the best QoS profile for VoIP, while Real-Time was the best profile for IPTV as they gave better delay and jitter values compared to other QoS profiles. Regarding VoIP, the quality of G.711 proved to be the better codec compared to GSM according to the E-Model. With respect to IPTV, MPEG-2 proved to be the better codec. To help determine how many TV sets a household could have while having good quality; it was shown that in intermediate conditions only 5 sessions could be streamed simultaneously. Performance evaluations such as this study are needed to enhance telephony for better quality of experience for the user.

TABLE V
WIMAX SATURATED WITH MEDIA

Codec	Delay(ms)	Max jitter (ms)	Mean Jitter (ms)	Packet loss (%)
GSM	143.857	35.967	21.017	32.65
PCMU	185.370	32.803	20.990	26.87
PCMA	166.018	31.992	25.772	37.51

Table VI
CALCULATION OF R-FACTOR WITH MOS VALUE UNDER INTERMEDIATE CONDITIONS

Codec	R-Factor	MOS	Ranking
GSM	72.38	3.71	good
PCMU	92.38	4.39	good
PCMA	92.38	4.39	good

REFERENCES

[1] A.C. Rumin, E.I.C. Guy. *Establishing how many VoIP calls a Wireless LAN can support without Performance degradation*, WMuNeP'06,Torremolinos, Malaga, 2006, Spain.

[2] Alvarion BreexeMAX manual.

[3] D. Vera. *Evaluation of WiMAX as a viable access technology to carry IP multimedia,* undergraduate THESIS. University of Cape Town. 2008.

[4] E. Francis, M. Retnasothie, K.Ozdemir, T. Ycek, H. Celebi, J. Zhang & R. Muththaiah,*Wireless IPTV over WiMAX: Challenges and Applications*, IEEE Wamicon, ,Clearwater, FL, 2006.

[5] F. Masuabi. *Evaluation of VoIP QoS over WiMAX, undergraduate* THESIS.University of Cape Town. 2008.

[6] M. Bakshi.*VoIP/Multimedia over WiMAX (802.16)* [ONLINE] Available: http://students.cec.wustl.edu/edu/~mb5/wimax_voip.html

[7] N. Scalabrino, F.De. Pellegrini, I. Chlamtac, A. Ghittino, S. Pera, *Performance Evaluation of WiMAX testbed under VoIP traffic*, InProc, WINTECH'06, 2006, pp. 97-98.

Francis Masuabi received her B.Sc. (Hons) degree from the University of Cape Town in 2008. She is currently working towards her MSc. at the same institution. Her research interests include mobility management in 3GPP Evolved Packet Core.

Denys Vera received his B.Sc. in Elec Comp Eng from the University of Cape Town in 2009. He is currently working towards his MSc. at the same institution. His research interests include next generation IPTV.