Comparison of VoIP Performance over WiMAX, WLAN and WiMAX-WLAN Integrated Network Using OPNET

Anindita Kundu[1], Suman Bhunia[1], Iti Saha Misra[2], and Salil K. Sanyal[2]

[1] School of Mobile Computing and Communication, Jadavpur University, Kolkata, India
kundu.anindita@gmail.com, sumanbhunia@gmail.com
[2] Electronics and Telecommunication Engineering, Jadavpur University, Kolkata, India
iti@etce.jdvu.ac.in, s_sanyal@ieee.org

Abstract. Voice over IP is expected to be in practice in the next generation communication networks. The target of this paper is to analyse its performance in the next generation networks. Among the most competing networks like WiMAX, Wifi, etc., WiMAX having higher bandwidth accommodates more users but with degraded performance. Hence, an integrated network using WiMAX backbone and WLAN hotspots has been developed and VoIP has been setup using SIP. Since, OPNET 14.5.A provides a real life simulation environment, it is chosen as the simulation tool. Quality of the service is critically analysed with parameters like jitter, MOS and delay for various voice codecs. Finally, it is concluded that the WiMAX-WLAN integrated network provides improved and optimal performance over WLAN and WiMAX network with respect to network capacity and quality of service.

Keywords: WiMAX, WLAN, Integrated network, backbone, hotspot, VoIP, Codecs, OPNET.

1 Introduction

Wireless networking has become an essential part in the modern telecommunication system. The demand of high speed data transfer with quality has led to the evolution of technologies like WiMAX and WLAN and is still increasing. Hence, new ways to enhance quality and speed of connectivity are being searched for.

WLANs [1] are mostly designed for private wired LANs and have been enormously successful for data traffic but voice traffic differs fundamentally from data traffic in its sensitivity to delay and loss [2]. Voice over WLAN is popular, but maintaining the speech quality is still one of many technical challenges of the VoIP system. VoIP is spreading rapidly and there is need to support multiple concurrent VoIP communications but WLAN support handful number of users [3] [4].

The IEEE 802.11 MAC specifies two different mechanisms, namely the contention-based Distributed Coordination Function (DCF) [1] and the polling-based Point Coordination Function (PCF) [1]. The DCF uses a carrier sense multiple access with collision avoidance (CSMA/CA) scheme for medium access and the optional

N. Meghanathan et al. (Eds.): NeCoM, WiMoN, and WeST 2010, CCIS 90, pp. 316–325, 2010.

four way handshaking request-to-send/clear-to-send [1] mechanism (RTS/CTS). Incapability of providing differentiation and prioritization based upon traffic type results in providing satisfactory performance for best-effort traffic only, but inferior support for QoS requirements posed by real time traffic. These requirements make the DCF scheme a less feasible option to support QoS for VoIP traffic. The PCF mode enables the polled stations to transmit data without contending for the channel. Studies on VoIP over WLAN in PCF mode [5] shows that the polling overhead is high with increased number of stations in a basic service set (BSS). This results in excessive delay and poor performance of VoIP under heavy load conditions. Thus, both DCF and PCF have limited support for real-time applications. Supporting VoIP over WLAN using DCF mode poses significant challenges, because the performance characteristics of their physical and MAC layers are much worse than their wired counterparts and hence considered in our system.

WiMAX (Worldwide Interoperability for Microwave Access) [6][16] on the other hand is designed to deliver a metro area broadband wireless access (BWA) service. So, while wireless LAN supports transmission range of up to few hundred meters, WiMAX system ranges up to 30 miles [6]. Unlike a typical IEEE 802.11 WLAN with 11Mbps bandwidth which supports very limited VoIP connections [4], an IEEE 802.16 WiMAX with 70Mbps bandwidth [7] can support huge number of users. These motivations led to study and comparison of the VoIP quality of service in IEEE 802.11b WLAN and IEEE 802.16 WiMAX network.

IEEE 802.16 support 5 types of service classes, namely UGS (Unsolicited Grant Service), rtPS (real time Polling Service), nrtPS (non-real time Polling Service), BE (Best Effort Service), ertPS (extended rtPS service) [8]. UGS supports fixed-size data packets at a constant bit rate (CBR). It supports real time applications like VoIP or streaming applications but wastes bandwidth during the off periods. rtPS supports variable bit rate(VBR) real-time service such as VoIP. Delay-tolerant data streams such as an FTP is designed to be supported by the nrtPS. This requires variable-size data grants at a minimum guaranteed rate. The nrtPS is similar to the rtPS but allows contention based polling. Data streams, such as Web browsing, that do not require a minimum service-level guarantee is supported by BE service. BE connections are never polled but receive resources through contention. ertPS was introduced to support VBR real-time services such as VoIP and video streaming. It has an advantage over UGS and rtPS for VoIP applications as it carries lower overhead than UGS and rtPS [9] and hence is modeled in the system.

Since, WiMAX is expected to create the opportunity to successfully penetrate the commercial barrier by providing higher bandwidth, establishing wireless commons becomes an important factor. Also, bandwidth crunch and network integration are some of the major technical and social challenges regarding the future of the community-based Wi-Fi networks [10]. According to [10], the foundation of the WiMAX PTP commons is the process of hot-spot interconnection and integration. Instead of global Internet connectivity, many current applications and businesses are expected to be better utilized by using the localized Wi-Fi constellation.

With a step towards the next generation, it is expected that an integrated network as shown in Figure 1, comprising of both the WiMAX and WLAN network and using mobile nodes with dual stack is expected to provide a better performance than a similar WiMAX or WLAN network.

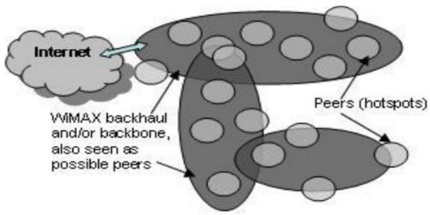


Fig. 1. Wi-Fi integration using WiMAX [10]

VoIP has been widely accepted for its cost effectiveness and easy implementation. A VoIP system consists of three indispensable components, namely 1) codec, 2) packetizer, and 3) playout buffer. Analog voice signals are compressed, and encoded into digital voice streams by the codecs. The output digital voice streams are then packed into constant-bit-rate (CBR) voice packets by the packetizer. A two way conversation is very sensitive to packet delay jitter but can tolerate certain degree of packet loss. Hence a playout buffer is used at the receiver end to smooth the speech by removing the delay jitter.

Quality of noise sensitive VoIP is usually measured in terms of jitter, MOS and packet end-to-end delay. Perceived voice with zero jitter, high MOS and low packet end-to-end delay is considered to be the best. With the two competing wireless networks namely WLAN and WiMAX, this paper analyses the perceived voice quality as measured using OPNET simulation environment.

2 Voice over IP

Voice is analog and is converted to digital format before transmitting over Internet. This process is called encoding and the converse is called decoding and both are performed by voice codecs [11]. With bandwidth utilization becoming a huge concern, voice compression techniques are used [11] to reduce bandwidth consumption. Voice compression by a codec adds an additional overhead of algorithmic delay. Thus, a codec is expected to provide good voice quality even after compression, with minimum delay.

Table 1 shows the bandwidth requirements of some common codecs. G.711 is the international standard for encoding telephone audio. It has a fixed bit rate of 64kbps. G.723 and G.729 are low bit rate codecs at the expense of high codec complexity. G.723 is one of the most efficient codecs with the highest compression ratio and is used in video conferencing applications. G.729 is an industry standard with high bandwidth utilization for toll-quality voice calls [12]. G.726 uses ADPCM speech codec standard, and transmits at rates of 16, 24, 32, and 40 kbps. G.728 officially codes speech at 16 Kbit/s using low-delay code excited linear prediction [13]. For example, during a call using G711 as codec, the amount of data transfer for both uplink and downlink will be 87.2 x 2 = 174.4Kbps = 0.1703 Mbps = 10.21 MB per minute. So, G 711 uses 10.21 Mb/min per VoIP call where as G 729 uses 0.5MB/min per voice call in the same way.

Table 1. Bandwidth Requirement of Some Common Codecs [11][14]

Codecs	Algorithm	Bandwidth (Kbps)	Ethernet Bandwidth Usage (Kbps)
G 711	PCM(Pulse Code Modulation)	64	87.2
G 729	CS-ACELP (Conjugate Structure Algebraic-Code Excited Linear Prediction)	8	31.2
G 723.1	Multi Rate Coder	6.3	21.9
G 723.1	Multi Rate Coder	5.3	20.8
G 726	ADPCM(Adaptive Differential Pulse Code Modulation)	32	55.2
G 726	ADPCM(Adaptive Differential Pulse Code Modulation)	24	47.2
G 728	LD-CELP (Low-Delay Code Excited Linear Prediction)	16	31.5

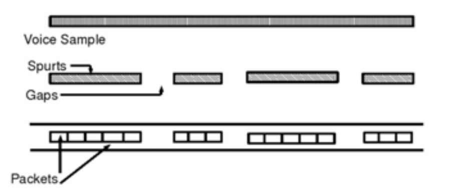


Fig. 2. Packet Creation based on voice activity

Moreover, recently voice codecs are developed to detect talk-spurt [15] and silence lengths [15] within a conversation. Silence in a communication period leads to packetization of the background noise and sending it over the network. This causes bandwidth wastage. Usually, during a conversation we talk 35% of the time and remain quiet rest of the time [15]. With silence suppression during the silence period, the codec does not send data as shown in figure 2. This decreases channel utilisation and thereby saves bandwidth.

Voice communication is noise sensitive. Noise causes the signal to reach the destination with a lead or lag in the time period. This deviation is called jitter. Lead causes negative jitter and lag causes positive jitter and both degrade the voice quality. The time taken by voice to be transmitted from the mouth of the sender to the ear of the receiver is called packet end-to-end delay. The packet end-to-end delay should be very less for voice communication. Perceived voice quality is typically estimated by the subjective mean opinion score (MOS), an arithmetic average of opinion score. MOS of a particular codec is the average mark given by a panel of auditors listening to several recorded samples. It ranges from 1(unacceptable) to 5 (excellent). It depends on delay and packet dropped by the network. The E-model, an analytical model defined in ITU-T recommendation, provides a framework for an objective on-line quality estimation based on network performance measurements like delay and loss and application level factors like low bit rate codecs. The result of the E-model is the calculation of the R-factor (best case 100 worst case 0) [5].

$$R = R_0 - I_s - I_d - I_e + A. \quad (1)$$

Where R_0 groups the effects of noise, I_s includes the effects of the other impairments related to the quantisation of the voice signal, I_d represents the impairment caused due to delay, I_e covers the impairments caused by the low bit rate codecs and packet losses. The advantage factor A compensates for the above impairments under various user conditions. A is 10 for mobile telephony but 0 for VoIP [5]. R_0 is considered to be 94.77 and I_s is considered to be 1.43 in OPNET 14.5.A. The relation between MOS and R-factor:

$$MOS = 1 + 0.035R + 7.10\text{-}6R(R - 60)(100 - R) \quad (2)$$

The purpose of this modeling is to compare the performance parameters for the voice codecs considering both with and without silence suppression in WiMAX 802.16d, WLAN 802.11b and their integrated network and thereby show that the integration provides optimal network capacity and quality of service.

3 Simulation Environment

Figure 3 show WiMAX network setup. The Wireless Deployment Wizard of OPNET is used to deploy a 7 celled WiMAX network, with multiple subscriber stations.

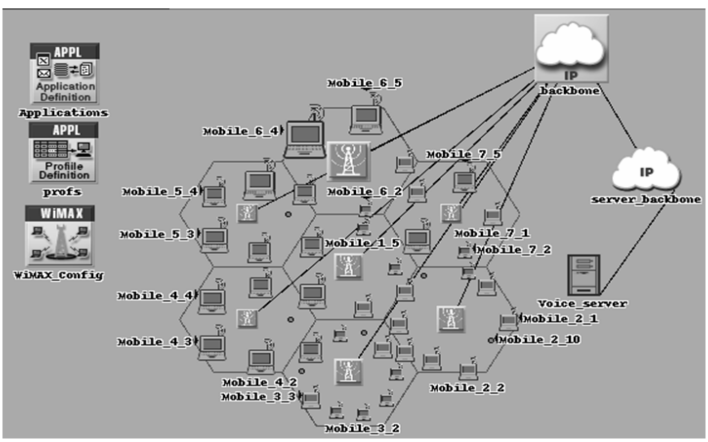


Fig. 3. Network Model for WiMAX

The base stations are connected to the voice server configured as the SIP server via an IP backbone and a server backbone. These nodes represent the service provider company network. The Base Station and subscriber station parameters are as shown in Table 2. The number of subscribers in cell 2 and cell 3 are 10 and VoIP calls are configured between them in mesh.

Table 2. WiMAX and WLAN Network Parameters

WiMAX Parameters	Values	WLAN Parameters	Values
WiMAX Service Class	ertPS	Physical Characteristics	Direct Sequence (DSSS)
BS Transmission Power	10W	Data Rate	11Mbps
SS Transmission Power	0.5W	Transmission Power	0.005 W
PHY profile	WirelessOFDMA 20MHz	Buffer Size	2048000 bits

Similar to the WiMAX network, a WLAN 802.11b network is also deployed by the Wireless Deployment Wizard of OPNET where a 7 celled WLAN network with multiple subscriber stations. Unlike the WiMAX network, the WLAN network has access points (APs) in place of the base stations and the mobile nodes of WiMAX are replaced by mobile nodes of WLAN. The APs are also connected to the core network. The APs are connected to the voice server configured as the SIP server via an IP backbone and a server backbone. These nodes represent the service provider company network. Similar to the WiMAX network, the VoIP calls are setup between the subscribers of cell 2 and cell 3 in mesh. The parameters of the access points and the subscriber stations are as shown in Table 2.

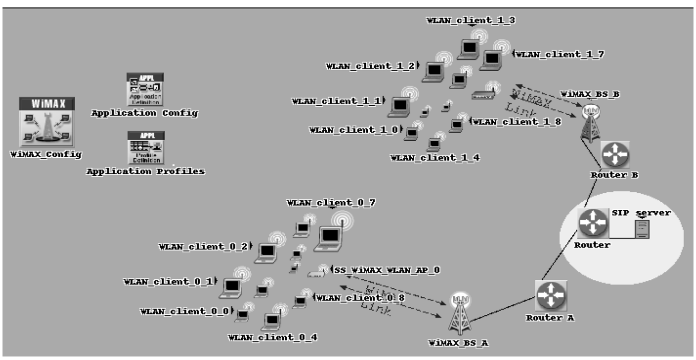


Fig. 4. Network Model for WiMAX-WLAN Integrated Network

Like figure 1, a WLAN integrated network using WiMAX is developed as shown in figure 4. Two WiMAX Base Stations are connected to each other and a SIP server via routers. A special type of node called SS_WiMAX_WLAN_AP having dual stack of both WiMAX and WLAN as shown in figure 5, is configured as the subscriber station of WiMAX network and Access Point for the WLAN network. This node is a bridge between the WLAN subscriber stations and the WiMAX base station. There are 10 WLAN subscriber stations under each SS_WiMAX_WLAN_AP. VoIP calls are setup between 10 users of two such SS_WiMAX_WLAN_AP. The parameters of the WiMAX Base Station and the WLAN subscriber stations are same as that shown in Table 2.

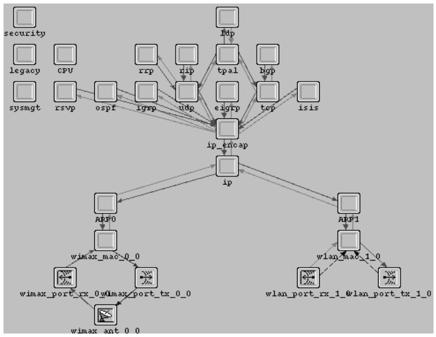
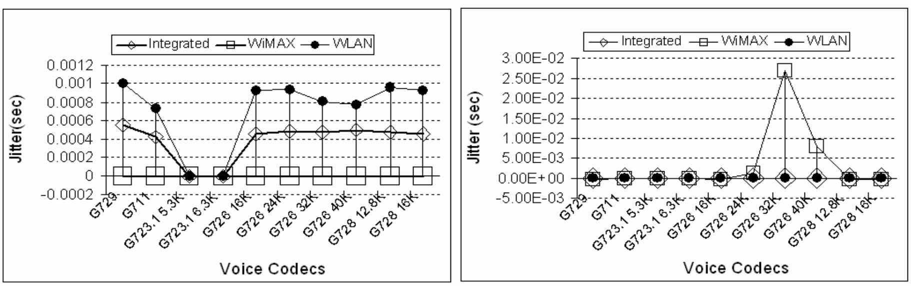


Fig. 5. Protocol Stack of SS_WiMAX_WLAN_AP

4 Simulation Results and Discussion

The variation of jitter with variation of the voice codecs with and without silence suppression is shown in Fig 6a and 6b. From 6a, we see that the average voice jitter is almost 0 for WiMAX implying very good quality of voice while WLAN has a positive jitter varying from about 0.0007 to 0.001 seconds. The integrated network shows jitter variation from about 0.0004 to 0.0006 seconds. For G 723.1, the average jitter is almost 0 irrespective of the network indicating a very good performance. This is because the bit rate of G 723.1 is 6.3 or 5.3 Kbps which results in generation of small packets. But modem and fax signals cannot be carried by G 723.1 [15]. It can be used only for narrow band communications.

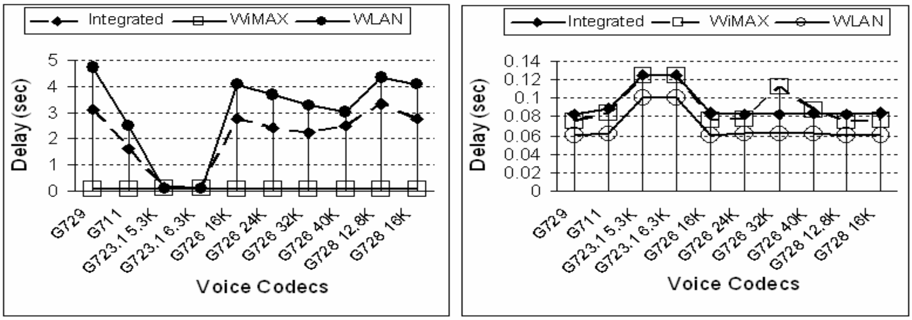


(a) (b)

Fig. 6. Average jitter, (a) Without silence suppression (b) With silence suppression

With silence suppression, as shown in Fig 6b, the result is different. Like G711, G 726 has its roots in the PSTN network. It is primarily used for international trunks to save bandwidth. Unlike G711, G 726 uses 32Kbps to provide nearly the same quality of voice [17] because 32 Kbps is the de facto standard. As shown in Figure 6b, the average voice jitter is almost 0 for all codecs in both WLAN and WLAN-WiMAX integrated network where as the WiMAX network shows a slight deviation for the voice codec G 726. The deviation is highest for G 726 with 32kbps and is about

0.03sec. Though G 726 supports data rate of 16 , 24, and 40Kbps also, the 24 and 16 Kbps channels are used for voice in Digital Circuit Multiplication Equipment (DCME) and the 40 Kbps is for data modem signals (especially modems doing 4800 Kbps or higher) in DCME. But G 726 with 32 Kbps follows the de facto standard and the jitter is within the consideration limit.



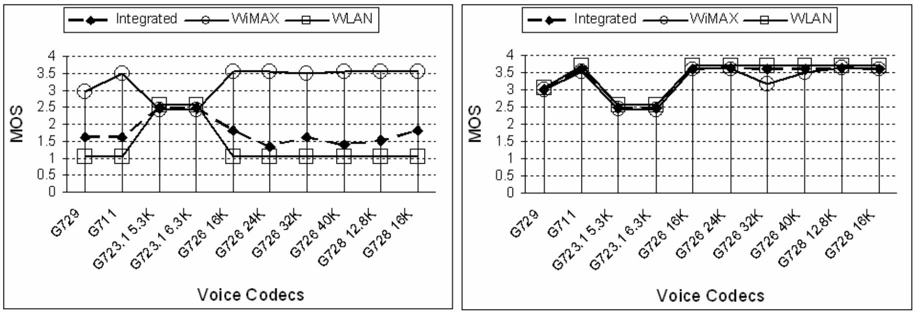
(a) (b)

Fig. 7. Average Packet End-to-end Delay (a) Without silence suppression (b) With silence suppression

As shown in Figure 7a, the packet end-to-end delay for voice without silence suppression is less than 0.5 seconds for WiMAX where as for WLAN high bit rate codecs it is very high. This is because the silence periods is also packetised and send thereby creating huge bandwidth requirement and congestion in the WLAN network. Integration results in increased capacity. Hence, the integrated network shows delay less than the WLAN network as increased capacity results in less congestion.

Figure 7b shows, the packet end-to-end delay with silence suppression for WiMAX is more than WLAN and the integrated network has the same almost as that of WiMAX network. This is because with silence suppression, the number of packets to be send decreases thereby releasing the congestion in the WLAN network and also the distance to be traversed by the packet for WiMAX is much higher than the distance to be covered by the packet in case of WLAN and even for the integrated network the distance traversed by the packet is almost is as huge as that of WiMAX. For WLAN the packet end-to-end delay is about 0.06 seconds except for G 723.1 for which it is about 0.1 seconds where as for WiMAX and WLAN-WiMAX integrated network, the delay varies from 0.08 seconds for G 729 and G 711 to 0.13 seconds for G 723.1.

Figure 8a shows the variation of MOS for all networks with variation of the voice codecs without silence suppression. As mentioned before, MOS depends on the packet end-to-end delay and packets dropped. As the figure shows, MOS value obtained for WiMAX is above 3, for WLAN is almost 1 and for the integrated network is about 1.5 except for G 723.1 for which it is about 2.5. This is because G 723.1 being a low bit rate codec creates numerous packets which results in increased delay due to packet reassembly. Hence, low MOS in case of WiMAX. On the other hand it has low bandwidth requirement and hence less packets are dropped for this codec in WLAN network. Hence higher MOS compared to the other codecs.



(a) (b)

Fig. 8. Average voice MOS (a) Without silence suppression (b) With silence suppression

The result with silence suppression is totally different and is shown in Fig 8b. As shown in figure, WLAN perform better among the three networks with a MOS value of above 3.5 except for G711 for which it is about 3 and for G 723.1 for which it is about 2.5. This is because with the silence suppression the number of packets in the network decreases thereby releasing the congestion in the WLAN network. This decreases the amount of packet dropped considerably and thereby increases the MOS value. The codec G 723.1 is a low bit rate codec having bit rate of 5.3 kbps and 6.3kbps and the packet end-to-end delay is larger for G 723.1 in comparison to the other codecs. Hence the MOS decreases.

5 Conclusion

In this paper extensive simulation based performance analysis for VoIP application is done over a WiMAX based BWA network, a WLAN (IEEE 802.11b), and also on a WLAN-WiMAX integrated network. For this we have used OPNET 14.5.A simulation platform. Multiple competing traffic sources using SIP signalling over the networks are generated and the trace of traffic and measurements for different performance parameters discussed in this paper are obtained.

Close observation of the results reveal that WiMAX network performs better on the basis of jitter, MOS and packet end-to-end delay than conventional WLAN 802.11b in case of voice applications when no silence suppression is considered, i.e. with no bar on the bandwidth usage, but when more users are to be accommodated, bandwidth becomes a constraint and hence silence suppression has to be used. With silence suppression, WLAN provides a better voice quality than WiMAX. Though, the WiMAX network provides high capacity, degradation of the voice quality is observed i.e. reflected from the MOS value. WLAN on the other hand has less capacity but provides a better voice quality. Hence, the integrated deployment of WiMAX and WLAN is expected to be competent enough to provide optimal voice quality with optimal network capacity.

Acknowledgment. The authors deeply acknowledge the support from DST, Govt. of India for this work in the form of FIST 2007 Project on "Broadband Wireless Communications" in the Department of ETCE, Jadavpur University.

References

1. Crow, B.P., Widjaja, I., Kim, J.G.: IEEE 802.11 Wireless Local Area Networks. IEEE Communications (1997)
2. Malone, D., Clifford, P., Leith, D.J.: On Buffer Sizing for Voice in 802.11 WLANs. IEEE Communications Letters (2007)
3. http://voip.biz-news.com/news/tags/en_US/VoIP
4. Smart Mobile™: Next-Generation Wlan Architecture For High Performance Networks, A Trapeze Networks White Paper (November 2006)
5. Chen, D.Y., Garg, S., Kappes, M., Trivedi, K.S.: Supporting VBR VoIP traffic with IEEE 802.11 WLAN in PCF mode. In: Proceedings of OPNETWork 2002, Washington D.C. (August 2002)
6. Retnasothie, F.E., Ozdemir, M.K., Y' Cek, T., Celebi, H., Zhang, J., Muththaiah, R.: Wireless IPTV over WiMAX: Challenges and Applications. In: IEEE WAMICON (2006)
7. Cai, L.X., Shen, X. (S)., Mark, J.W., Cai, L., Xiao, Y.: Voice Capacity Analysis of WLAN With Unbalanced Traffic. IEEE Transations on Vehicular Technologies 55(3) (May 2006)
8. Nagarajan, N., Rajeev, Kaarthick, B.: Performnance analysis of Video Conferencing and Multimedia application Services over WiMAX. In: IEEE International Advance Computing Conference (IACC 2009), Patiala, India, March 6-7 (2009)
9. Chu, G., Wang, D., Mei, S.: A QoS Architecture for the MAC protocol of IEEE 802.16 BWA System. In: IEEE 2002 ICC, Circuits and Systems, July 2002, vol. 1, pp. 435–439 (2002)
10. Angelov, B., Rao, B.: The Progression of WiMAX Toward a Peer-to- Peer Paradigm Shift. Annual Review of Communications 59 (2007)
11. http://whirlpool.net.au/wiki/?tag=VoIP_Codecs
12. http://en.wikipedia.org/wiki/G.726
13. Trad, A., Munir, F., Afifi, H.: Capacity Evaluation of VoIP in IEEE 802.11e WLAN Environment. In: IEEE CCNC 2006 Proceedings (2006)
14. http://wiki.snom.com/Bandwidth_requirements_and_QoS
15. Hersent, O., Petit, J., Gurle, D.: IP Telephony- Deploying Voice-Over-IP Protocols. John Wiley & Sons, Ltd., Chichester (2005)
16. Ghosh, A., Wolter, D.R., Andrews, J.G., Chen, R.: Broadband Wireless Access with WiMax/802.16: Current Performance Benchmarks and Future Potential. IEEE Communications Magazine (February 2005)
17. http://www.voip-info.org/wiki/view/ITU+G.726S