Improving R-score of Adaptive VoIP Codec in IEEE 802.16 networks

Tsung-Yin Lee and Jen-Yi Pan,
Department of Communications Engineering,
National Chung Cheng University Chia-Yi, Taiwan, Republic of China
E-mail: gyinlee@msn.com, jypan@comm.ccu.edu.tw

Abstract—**The IEEE 802.16 system called WiMAX (Worldwide Interoperability for Microwave Access) provides quality of service (QoS) of several types for different service. The WiMAX is expected to support QoS for real time application, such as Voice over IP (VoIP). In this paper, when network congestion occurs, the VoIP bit-rate needs to be adapted to achieve the best speech quality. We propose a new scheme called Adaptive VoIP Level Coding (AVLC). VoIP is sensitive to delay and loss. According to different network conditions such as varied modulation, packet delay, packet loss, and residual time slot, we use G.722.2 codec to adapt each connection's data rate. Simulation experiments are conducted to test the performance (network delay, packet loss, and R-score) of the proposed mechanisms. In result, we increase the R-score about 40% to 50%.**

I. INTRODUCTION

WiMAX provides a last-mile broadband access technology. Orthogonal frequency division multiplexing (OFDM) adopted in WiMAX increases channel bandwidth and capacity. WiMAX can transmit in the distance of up to 30 miles requiring line of sight (LOS) [1]. WiMAX supports five quality-of-service (QoS) types to transmit data [2], such as Unsolicited Grant Service (UGS), real-time Polling Service (rtPS), non-real-time polling service (nrtPS), extended real-time Polling Service (ertPS) and best efforts service (BE). It is suitable to transmit variable real-time data in rtPS and ertPS.

WiMAX can support large range, high speed and QoS guarantee to transmit data, so many real time applications are well-served in WiMAX like video conference, VoIP. Telecommunication Standardization Sector of the International Telecommunication Union (ITU-T) defines some voice modulation methods using the technique of Adaptive Multi Rate-Wide Band (AMR-WB) especially suitable for voice transmission in variable data rates such as G.722.2 [3]. G.722.2 can use different bit-rates to convey voice signal. In different network conditions, we use different VoIP bit-rates to transmit our speech data.

VoIP is sensitive to delay and loss. We transmit VoIP data in WiMAX, and the physical layer's burst profile specifies connection's modulation scheme and coding rate such as binary phase shift keying (BPSK), quaternary phase shift keying (QPSK), and so on. When the signal becomes stronger or the connection quality (like signal-to-noise ratio) is better now, we use more advanced coding rate. When signal becomes weak or connection quality is worse, we use slower modulation scheme and coding rate to reduce transmission error. When the network quality gets better or worse, we can adjust the VoIP bit-rates for each connection according to network conditions like delay, packet loss, and residual time slot, as shown in Fig.1. The residual time slots are the time slots which are not scheduled to transmit any data. When the number of residual time slots approaches to zero, we estimate the network becoming congestion. Thus, we can reduce the network delay and packet loss to obtain better communication quality in the network.

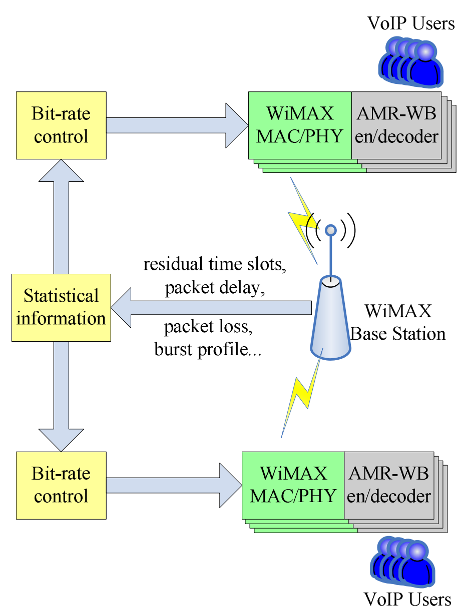

Fig.1 System Architecture

In addition, we use R-factor [4] [5] to assess communication quality. Simulation experiments are conducted to verify our scheme. We assume that the number of VoIP sources will gradually increase. Finally, we use network simulator 2 (NS2) [6] to simulate our scheme, and we can increase the R-score by 40% to 50% in result. The 11Mbps WiMAX link can support more than 250 VoIP connections that have average R-score over 55.

The rest of the paper is organized as follows. In section II, we discuss VoIP related with R-factor and AMR-WB G.722.2. In section III, we introduce medium access control (MAC) layer and physical (PHY) layer in WiMAX. In section IV, we explain that how to adaptive VoIP on WiMAX. In section V, we use

NS2 with National Institute of Standards and Technology (NIST) [7] WiMAX module to simulate our scheme. We draw a conclusion and future works in the last section.

II. VoIP AND CODEC

VoIP is sensitive to delay and loss, so this section will discuss the effects of packet delay and packet loss on R-factor.

2.1 G.722.2

G.722.2 is a speech coding standard developed by ITU-T [3]. It has 9 different bit rates (6.60kbps~23.85kbps). The lowest bit rate providing excellent speech quality in a clean environment is 12.65kbps. Using lower G.722.2 bit-rates can save bandwidth, but the speech quality will degrade comparing to higher bit-rates'. Although reduced VoIP bit-rates will lose some speech quality, it can decrease delay and packet loss. Then we can improve the performance. The different codecs affect of R-factor discussing in [8], and we will describe the detail in section 2.3.

2.2 R-factor

The ITU-T E-model [4] introduces a meter "R-factor" to gauge voice quality. The transmission rating scale ranges from R = 0 (lowest possible quality) to R = 100 (optimum quality). R-factor combines different aspects of voice quality impairment. The formula for the R-factor is as follows:

$$R = R_0 - I_S - I_e - I_d + A \tag{1}$$

R_0 is the basic signal-to-noise ratio by typical circuit switched network. The factor I_s is the sum of all impairments which may varies higher or lower during the voice transmission. A is the other advantage factor. We can rewrite R-factor's formula with default values:

$$R = 93.2 - I_e - I_d \tag{2}$$

Thus, we only have two factors that affect the R-value. I_e is the loss impairment factor that is caused by packet loss and codec loss. I_d is the delay impairment factor that is caused by mouth-to-ear delay. We will describe the detail in section 2.3 and 2.4.

TABLE I
Quality of R-factor

R-factor	User Experience
90	Excellent
80	Good
70	Fair
60	Poor
50	Bad

2.3 Loss Effect

Loss Effect mainly has two parts; one of them is affected by algorithm, and the other is affected by packet loss. In [8], there are many experiments about loss effect on different codec, including G.722.2. According to [8], we can compute loss effect as follows:

$$I_e = I_{e,WB} + (129 - I_{e,WB})\left(\frac{P_{pl}}{P_{pl} + B_{pl}}\right) \tag{3}$$

Where $I_{e,WB}$ is impairment factor and, B_{pl} is packet loss robustness factor for speech codec, using a least-squares approximation with experiments. Table II lists two parameters. P_{pl} is the loss probability due to the network losses.

TABLE II
A part of G.722.2 parameters [6]

G.722.2 (kbps)	$I_{e,WB}$	B_{pl}
12.65	11	13.0
14.25	10	14.1
15.85	7	13.1
18.25	5	12.5
19.85	4	12.3
23.05	1	13.0

2.4 Delay Effect

The total mouth-to-ear delay is composed of three components: codec delay, network delay, and playout delay [9]. Codec delay has two parts: one is the algorithm delay, and the other is packetization delay. Codec delay of G.722.2 is 25ms including frame size (20ms) and lookahead (5ms). Network delay is the end to end delay for packet transmission. Playout delay stands for the delay that is from the reception of packets to start playing. So the total delay is as follows:

$$D_{all} = D_{network} + D_{playout} + D_{codec} \tag{4}$$

VoIP is delay sensitive, and 177.3ms is a critical delay time in VoIP. When we calculate total delay, we can compute I_d and write the final formula as follows [5]:

$$I_d = 0.024D_{all} + 0.11(D_{all} - 177.3)H(D_{all} - 177.3) \tag{5}$$

Where H(x) is an indicator function: H(x) = 0 if x < 0, and H(x) = 1 otherwise.

III. WIMAX PROTOCOL

In this section, we introduce a part of PHY and MAC protocol in WiMAX.

3.1 Media Access Control Layer

In WiMAX [2], Base Station (BS) will periodically broadcast Downlink Channel Descriptor (DCD) and Uplink Channel Descriptor (UCD). DCD and UCD include burst profile. Each connection's coding rate and modulation scheme will be recorded in the burst profile. In addition, BS will also broadcast downlink map (DL-MAP) and uplink map (UL-MAP) periodically. The Uplink Interval Usage Code (UIUC) and Downlink Interval Usage Code (DIUC) record each connection's burst profile and we can get UIUC and DIUC in UP-MAP and DL-MAP. In the downlink subframes, both the DL-MAP and UL-MAP messages are transmitted, which specify the allocation of transmission opportunity for each connection. Besides, the UCD count and DCD count in UL-MAP and DL-MAP indicate which frame will carry UCD and DCD.

3.2 Physical Layer

Burst profile records different level of modulation schemes in connection, and we can get burst profile in UCD and DCD. Each connection can use different burst profile. The faster modulation scheme we use, the more data we can transmit. In different signal-to-noise ratio (SNR), WiMAX can adjust burst profile of each connection.

TABLE III
Modulation and coding schemes for WiMAX [10]

Modulation Scheme	Coding rate	bits/symbol	SNR for 1% PER (dB)
BPSK	1/2	0.5	1.5
QPSK	1/2	1.0	6.4
QPSK	3/4	1.5	8.2
16QAM	1/2	2.0	13.4
16QAM	3/4	3.0	16.2
64QAM	1/2	4.0	21.7
64QAM	3/4	4.5	24.4

IV. PROPOSED SCHEME

According to section II and section III, we present a new scheme called Adaptive VoIP Level Coding (AVLC). When the network congestion occurs, we read DIUC and UIUC's information to get each connection's burst profile, and we will adapt VoIP bit-rates according to G.722.2 bit-rates and each connection's burst profile. We must change transmission bit-rates, so we use rtPS to transmit our VoIP data.

First, we should detect network environments. Network delay and packet loss are the two main factors that affect VoIP. We update the value of $AVLC_{value}$ as follows:

$$AVLC_{value}(t)=\begin{cases}\min\{AVLC_{value}(t\text{-}1)+1, AVLC_{max}\} \\ \quad \text{if } D(t)>D_{max} \text{ or } PL(t)>L_{max} \text{ and } N_{rts} =\sim 0 \\ \max\{AVLC_{value}(t\text{-}1)-1, 0\} \\ \quad \text{if } D(t)<D_{min} \text{ or } PL(t)<L_{min} \text{ and } N_{rts} >\sim 0\end{cases} \quad (6)$$

Where $AVLC_{value}(t)$ is the adaptive codec value at time t, we can change connection's VoIP bit-rates according to Fig.2. The G.722.2 bit-rates can determine as follows:

$$\beta_{G.722.2}(N,t)=\beta_{hb}-\alpha\left(\frac{AVLC_{value}(t)}{\beta_{bs}(N,t)}\right) \quad (7)$$

We have N connections: each connection has different numbers from 1 to N. $\beta_{G.722.2}$ is the determinant VoIP bit-rates of each connection at time t. β_{hb} is the highest G.722.2 bit-rates (23.05kbps) that we use to adjust. β_{bs} is the transmission rate (bits/symbol) of connection at time t. According to TABLE III, β_{bs} will change by different modulation scheme. α is the VoIP control parameter which is a constant. When the network quality change quickly, we can set the α larger, and we can adjust faster when α is larger. In (7), when the G.722.2 (kbps) is between the two G.722.2 bit-rates, we choose the one which is close to our value.

Different burst profile in the same network conditions should use different VoIP bit-rates. The better modulation scheme we used, the slighter of bit-rates of VoIP will be reduced. Because the occupied physical slot will decrease as the faster modulation scheme, we adapt the connections that using slower modulation scheme and coding rate first can save more bandwidth.

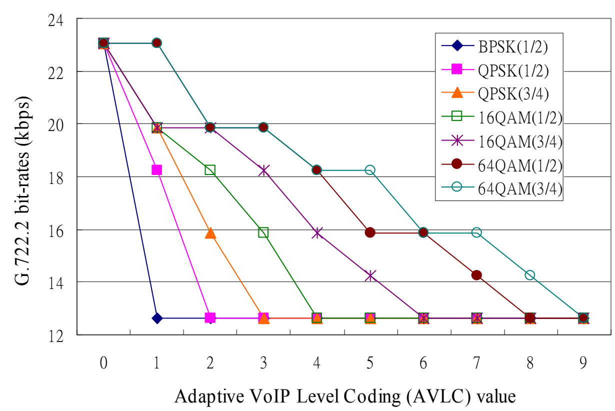

Fig. 2 AVLC value VS G.722 bit-rates (α=5)

D(t) is the average network delay at time t, and L(t) is the average packet loss rate at time t. According to (5), the 177.3ms is a critical time value of VoIP quality. If we use G.722.2 (23.05kbps), packet loss rate decreases more than 20 R-score (figure 3). We set the two values as follows:

$$D_{max}=177.3ms-D_{playout}-D_{codec} \text{ and } L_{max}=3\% \quad (8)$$

Finally, we set D_{min} and L_{min} smaller than D_{max} and L_{max}. If the network satisfies both two conditions (EX: $D_{min}<D(t)$ and $L_{max}>L(t)$), BS will also check the residual time slot N_{rts}. If network status gets better, we will also change $AVLC_{value}$ to adjust VoIP bit-rates for each VoIP session.

```
1: initial value ( BS can set different values (2~6) )
2: AVLC_value ← 0;
3: α ← 5;
4: D_max ← 177.3ms – D_playout – D_codec;
5: L_max ← 3%;
6: D_min ← lower than D_max;
7: L_min ← lower than L_max;
8: collect D(t), L(t), N_rts periodically;
9: if D(t) > D_max or L(t) > L_max and N_rts=0 than
10:   AVLC_value(t) ← min (AVLC_value(t-1) + 1, AVLC_max);
11: else if D(t) < D_min or L(t) < L_min and N_rts > 0 than
12:   AVLC_value(t) ← max (AVLC_value(t-1) – 1, 0);
13: end if
14: according to TABLE III and (7), we set VoIP bit-rates of
    each connections;
15: loop (repeat step 8~14);
```

Fig. 3 Adaptive VoIP Level Coding (AVLC) scheme

According to Table II and (3), as we know that if the VoIP bit-rates are reduced, the loss impairment factor will be raised. But when we reduce network delay and packet loss enough, we can decrease loss effect to gain better communication quality.

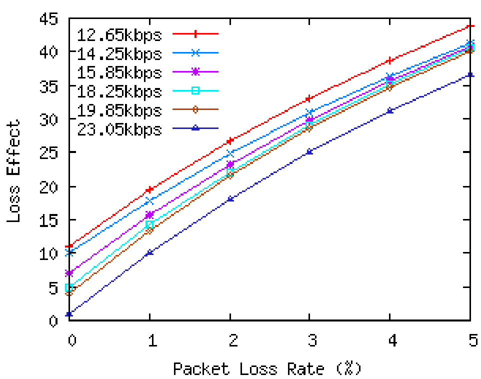

Fig. 4 Different G.722.2 codec vs. loss effect

When the network conditions are changed (user number, SNR, delay, loss and so on), the AVLC will adjust the VoIP bit-rates of each connections. AVLC will be adjusted from Fig.3 and the following several steps:

. Step1: BS collects packets and analyzes packets information (packet loss, network delay, residual time slot) periodically.

. Step2: according to packet loss, network delay and default value (codec delay, playout delay, impairment factors and packet loss robustness factors), we compute I_e, I_d and R-score.

. Step3: we read DIUC and UIUC's information to get each connection's modulation scheme and coding rate.

. Step4: based on step3 and step4, we can decide every connection's VoIP bit-rates.

. Step5: if VoIP bit-rates change, we transmit information to related SS.

· Step6: repeat step1 to step5.

V. Simulation

We have two methods. One is using AVLC (12.65kbps to 23.05kbps), and the other is using fixed VoIP bit-rates (23.05kbps). We draw R-score, network delay, packet loss and source numbers based on two experiments. Total simulation time is 56 seconds. When the simulation time is from 0 to 50 seconds, we increase VoIP source number from 0 to 250. After the 50th second, we fix VoIP source number to 250.

We use NS2 and NIST WiMAX module to carry out the simulation of the experiment. We modify NIST WiMAX module to add rtPS class, and BS will poll the group of rtPS class every 20ms. Some of the parameters used in the AVLC scheme are set as follows: $AVLC_{max}$ = 9, D_{max} = 177.3ms – $D_{playout}$ (60ms) – D_{codec} (25ms) = 92.3ms, L_{max} = 3%, D_{min} = 30ms, and L_{min} = 1%. We assume that all packets are transmitted and received without errors and the transmission delay is negligible. The VoIP streams were generated by a G.722.2 codec. Other simulation parameters are shown in the TABLE IV.

TABLE IV
Simulation Parameters

System	OFDM/TDD
Channel Bandwidth	11MHz
Beam Pattern	Omni Antenna
Frame Duration	4 ms
$D_{playout}$	60 ms
D_{codec}	25 ms
VoIP traffic	VBR (AMR-WB)
burst profile distribution	uniform
β_{hb}	23.05kbps
α	5

In Fig.5 and Fig.6, we present the network delay and packet loss for both adaptive and non-adaptive schemes. At the 28th second, the network has reached the critical point to congestion, and we start our adaptive scheme. Although the VoIP source will keep increasing before 50th second, the network delay and packet loss are better than the non-adaptive scheme.

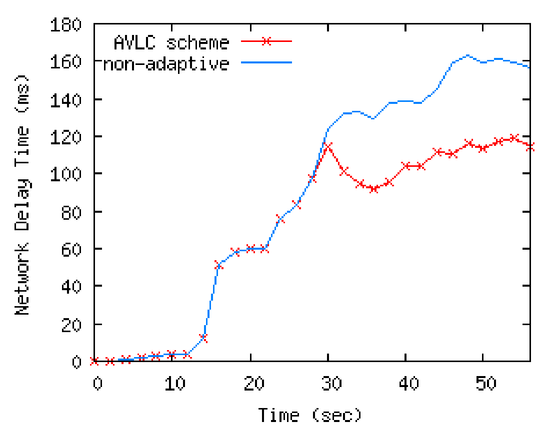

Fig. 5 network delay vs. time

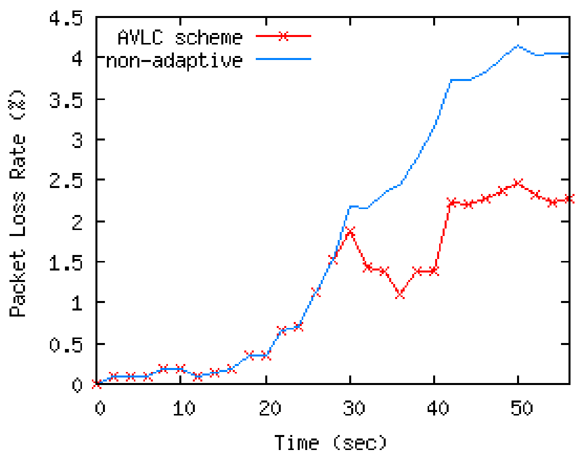

Fig. 6 packet loss rate vs. time

In Fig.7, we show the R-score for both adaptive and non-adaptive schemes. R-score is closely related to delay and loss. After adjustment, we improve the R-score by 40% to 50%. According to Section IV and Fig.4, if we reduce VoIP bit-rates in the same condition on packet loss, the loss impairment factor will rise. We can get better result when we reduce packet loss and network delay enough.

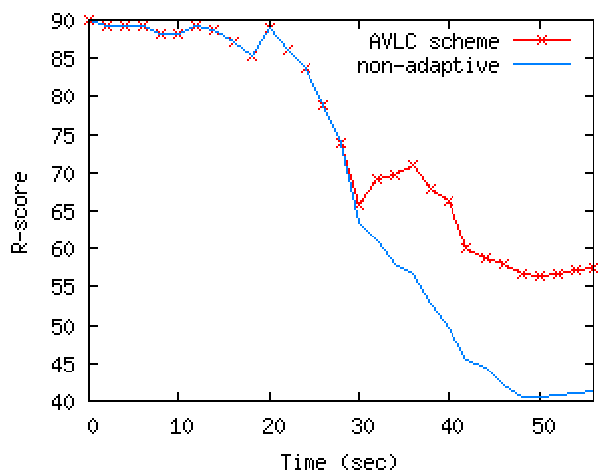

Fig. 7 R-score vs. time

In Fig.8, in the network with high quality from 0 to 28 seconds, we can use fast G.722.2 codec (23.05kbps). At the 28th second, when the network becomes congested, we will decrease the rate gradually (based on each connection's burst profile). Until the 44th second, we already adjust all connection to the lowest VoIP bit-rates (12.65kbps). In Fig.6, the packet loss rate is a rise after 40 second. Although all VoIP bit-rates can't adapt, we also get higher communication quality through AVLC.

VI. Conclusion and Future Work

In this paper, we studied the two most critical effects (packet loss and delay) in VoIP and we use our scheme to adapt VoIP codec over WiMAX. According to each connection's burst profile, we can adapt VoIP bit-rates to enhance VoIP quality. In our scheme, we reduce packet loss and network delay, so we could improve the R-score and allow more users to join the network. Under the same network conditions, we increased the R-score by 40% to 50%.

In the future, we will consider different conditions to adapt VoIP bit-rates, like utility requirement, fairness requirement and so on. We will also compare it with different codecs and compute R-score in future experiments.

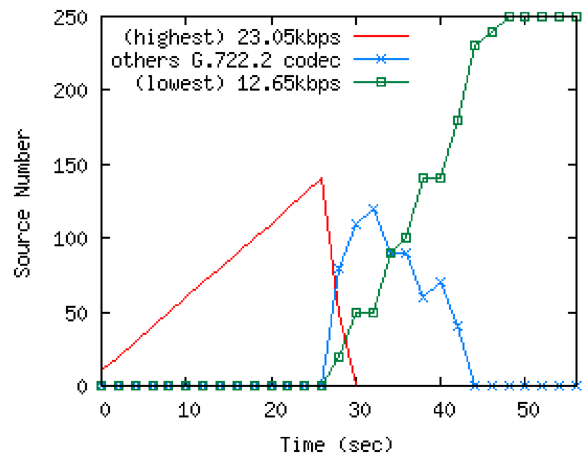

Fig. 8 number of VoIP sources using different G.722.2 codec

Acknowledgements

The authors would like to thank the Industrial Technology Research Institute and the Ministry of Economic Affairs (Contract No.MOEA97-EC-17-A-03-R7-0619) of the Republic of China, Taiwan, for financially supporting this research. This work was also supported partially by National Science Council of Republic of China under Grants NSC 97-2221-E-194-025 and 96-2219-E-194-008.

References

[1] S.J. Vaughan-Nichols, "Achieving wireless broadband with WiMAX", IEEE Computer, Vol.37, NO.6, pp.10-13, June 2004.
[2] IEEE standard 802.16d 2004, "IEEE Standard for Local and metropolitan area networks", October 2004.
[3] ITU-T Recommendation G.722.2, "Wideband coding of speech at around 16kbit/s using adaptive multi-rate wideband (AMR-WB)", January 2002.
[4] ITU-T Recommendation G.107, "The E-Model, a computational model for use in transmission planning", December 1998.
[5] R. G. Cole and J. H. Rosenbluth, "Voice over IP performance monitoring", ACM SIGCOMM Computer Communication Review, Vol.31, NO.2, pp. 9-24 , April 2001.
[6] The Network Simulator – NS-2, http://www.isi.edu/nsnam/ns.
[7] National Institute of Standards and Technology WiMAX Module, http://www.nist.gov.
[8] Sebastian Möller, Alexander Raake, Nobuhiko Kitawaki Akira Takahashi, and Marcel Wältermann, "Impairment Factor Framework for Wide-Band Speech Codecs", IEEE Transactions on Audio, Speech, and Language Processing, Vol.14, NO.6, pp.1969-1976, November 2006.
[9] Shamik Sengupta, Mainak Chatterjee, and Samrat Ganguly, "Improving Quality of VoIP Streams over WiMax", IEEE Transactions on Computers, Vol.57, NO.2, pp.145-156, February 2008.
[10] Christian Hoymann, "Analysis and performance evaluation of the OFDM-based metropolitan area networks IEEE 802.16", Computer Networks, Vol.49, NO.3, pp.341–363, October 2005.