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Author(s)	Lin, XH; Niu, L; Wang, H; Kwok, YK
Citation	Proceedings of the 7th IEEE International Wireless Communications and Mobile Computing Conference (IWCMC 2011), Istanbul, Turkey, 4-8 July 2011, p. 337-342
Issued Date	2011
URL	http://hdl.handle.net/10722/158720
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On Exploiting the On-Off Characteristics of Human Speech to Conserve Energy for the Downlink VoIP in WiMAX Systems

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Abstract—Energy conservation is a critical issue in the emerging standard IEEE 802.16e/m WiMAX supporting mobility. To guarantee QoS requirements in real-time services such as VoIP, traditional energy saving strategies adopt constant listen-sleep intervals, ignoring the On-Off characteristics of human speech. However, statistically the silence period can account for nearly 60% of the whole speech in time scale. Therefore, neglecting this fact can lead to unnecessary periodical listening in the silence duration, and, in turn, can result in excessive waste of battery energy.

In this paper, we adopt a hybrid energy management for the downlink simplex VoIP. We also give an evaluation model to analyze the performance of the scheme. Guided by this model, we obtain the optimal window adjustment parameters. Extensive simulation results have validated the analytical model, and indicated that, compared with the traditional scheme, the hybrid scheme can achieve as much as 90% reduction in energy dissipation during silence period, while meeting the QoS requirements satisfactorily at the same time.

Keywords—Energy Conservation, WiMAX, VoIP, Sleep Mode, QoS

I. INTRODUCTION

The ever growing demand on high-speed and ubiquitous wireless Internet access has spurred the standard of WiMAX (Worldwide Interoperability for Microwave Access), which aims at providing broad bandwidth at low cost for residential and business areas [1-7]. Due to the high energy consumption involved in these services by the communication and computation units, how to intelligently manage the energy is a hot research topic.

Sleep mode is a state in which a mobile station (MS) conducts pre-negotiated periods of absence from the serving base station (BS) air interface. The periods are characterized by the unavailability of the MS, as observed from the serving BS, to DL or UL traffic [3, 4]. When there is no traffic between BS and MS, through the signaling exchange, the MS can enter sleep mode and power down relevant communication units to save energy. Specifically, the MS can initiate the sleep operation by sending an MOB_SLP-REQ message, which defines the requested sleep profile, to the BS. In this sleep profile, some parameters such as initial-sleep window, final-sleep window base, final-sleep window exponent, listening window, and start frame number for first sleep window, are included. On receiving the MOB_SLP-REQ message, the BS may comply with this profile as recommended and respond by sending an MOB_SLP-RSP message back to the MS. When MS receives the MOB_SLP-RSP message, it can enter sleep mode. Meanwhile, during the sleep period, there may be downlink traffic addressed to the MS which is temporally unavailable to the BS. Therefore, the BS must buffer the coming traffic for the MS. To probe the downlink traffic, from time to time, the MS must wake up in the listening window specified in the profile and receive the MOB_TRF-IND message sent from BS. If the traffic

indication flag in the message is negative, which means no coming traffic, the MS can again enter sleep state. Otherwise if the flag is positive, the sleep mode is deactivated, and the MS must enter active mode to receive packets. According to the characteristics of traffics, three power saving classes have been defined in the standard. Power saving class (PSC) of type I is recommended for best effort services and non-real time traffics such as WEB browsing. Due to the busy behavior of the traffic arrival, the sleep window in PSC-I is doubled each time when the traffic indication flag is negative, until the window size reaches the maximum, thus avoiding unnecessary listening. PSC-II is recommended for unsolicited grant services (UGS) and real time connections such as VoIP. To guarantee the QoS, the sleep interval is constant and adjusted according to arrival interval of the coded VoIP packets. PSC-III is recommended for multicast connections as well as for management operations, and is not discussed in this paper. In dealing with energy conservation in WiMAX, much research has been done.

However, most of the literature works are various enhancements or analysis based on PSC-I, and for the sake of analytical simplicity, Poisson traffic model has been employed in performance evaluation. Nevertheless, Poisson assumption and PSC-I are not applicable to VoIP traffic because: (1) packet arrival in VoIP does not obey Poisson process; and (2) PSC-I is unsuitable for VoIP due to the fact that the delay and packet loss rate constraints cannot be properly handled. To satisfy the QoS requirements, PSC-II, which adopts constant sleep and listening intervals, is recommended for VoIP by the standard. However, according to speech model recommended by ITU [8], human speech consists of alternating talk-spurt and silence (pause) periods, both complying with negative exponential distribution, with mean 1.004 and 1.587 seconds, respectively. Statistically, total silence periods can amount to nearly 61% of the speech in time scale. Therefore, constant listening in PSC-II may incur unnecessary wakeup during the silence period, and, in turn, may expend more energy.

To tackle this problem, a hybrid scheme is proposed by Choi in [7]. Specifically, in the hybrid scheme, during the talk-spurt period, fixed listening and sleep intervals are adopted to fit the constant-bit-rate arrival of the coded voice packets. While in the silence period, exponential sleep window adjustment is used to probe the traffic arrival, thus reducing unnecessary listening. This scheme is proved to be effective in energy conservation. However, the scheme proposed in [7] cannot be applied directly to downlink simplex VoIP traffic without the consideration of the influence of sleep window adjustment on the QoS performance. Given the delay and loss rate constraints, how to select the sleep window parameters such that the tough QoS requirements can be satisfied? A large sleep window can lead to severe distortion in the voice quality, while a small window can incur more energy consumption. We

must strike a balance between QoS and energy expenditure. Moreover, in the adjustment of sleep window, the speed of natural human speech should also be taken into consideration.

In this paper, based on the hybrid method [7], we strive to find the optimal sleep window parameters under the constraints of QoS requirement imposed by the simplex VoIP traffic. The paper is organized as follows. In Section II, we present the background of human speech model and basic idea of hybrid power saving scheme. To evaluate the performance of the scheme, in Section III, we give the mathematical analysis model, and derive the delay, loss rate and number of invalid wakeup. This is followed by the analysis results validated by the simulation experiment in Section IV. Guided by these results, we also obtain the optimal window adjustment parameters. Finally, we conclude the paper in Section V.

II. HUMAN SPEECH MODEL AND THE HYBRID SCHEME

In this paper, we consider a downlink simplex VoIP traffic model. It is a simplex transmission of speech from the base station to the mobile user, and only the downlink voice is supported. Normally, it is adopted in many voice broadcast scenarios such as talk show, news broadcasting, oral presentation, etc. According to human speech model, both the probability density function (PDF) for silence and talk-spurt durations with hangover in monologue can be approximated by two exponential functions [8 - 9]. Specifically, the cumulative distribution function (CDF) for the duration of talk-spurt can be expressed

as $P(t < \tau) = 1 - e^{-\frac{\tau}{T_\lambda}}$, while the CDF for the silence period is $P(t < \tau) = 1 - e^{-\frac{\tau}{T_s}}$, with $T_\lambda = 1.004, T_s = 1.587$, respectively.

To avoid unnecessary listening during the silence period, in hybrid scheme, the sleep window is doubled when the traffic indication is negative, until the window size reaches the maximum. Specifically, the evolution of sleep window is given by

$$T_n = \begin{cases} T_1, & n = 1 \\ \min(2^{n-1}T_1, T_{\max}), & n > 1 \end{cases} \quad (1)$$

Where T_1 and T_{\max} are the initial and the maximum sleep window size, respectively.

In the talk-spurt period, the listening and sleep intervals are constant values, and during the listening interval, the MS can receive the downlink VoIP traffic. The hybrid power saving scheme is illustrated in Fig.1.

III. PERFORMANCE ANALYSIS

VoIP service has tight QoS requirements – average delay and packet loss rate. To avoid severe deterioration on the performance, these two constraints must be properly handled. In the hybrid scheme, the most significant factor influencing the performance is sleep window adjustment. A large sleep window can avoid frequent listening and thus save energy. However, it can also render excessive delay

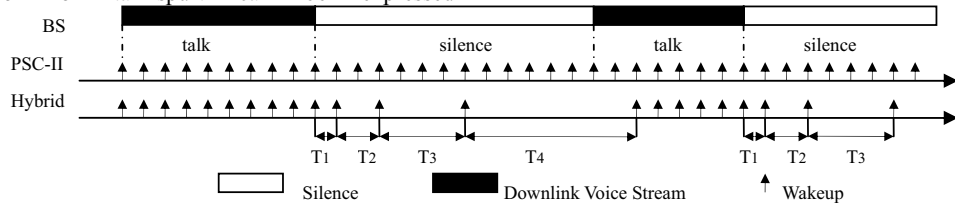


Fig.1 Hybrid Power Saving Scheme

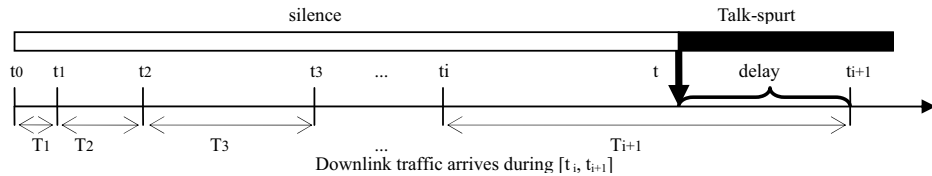


Fig.2 The Checking Point for the Traffic Arrival

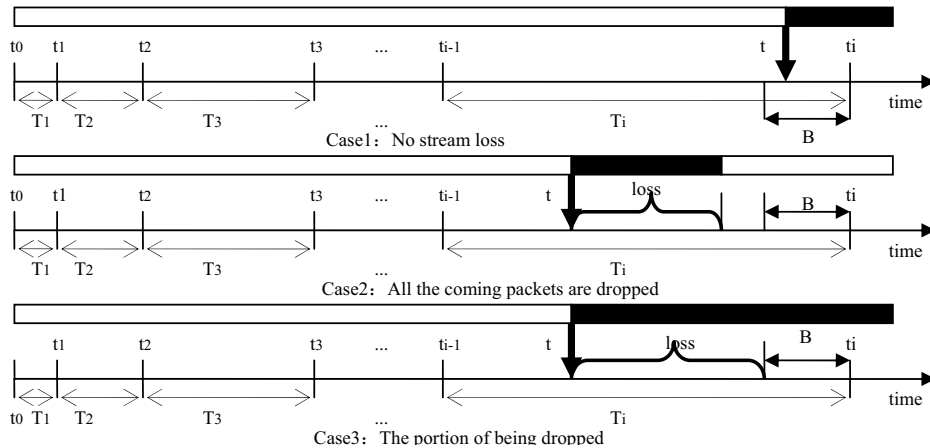


Fig.3 Talk-spurt Arrival in Different Time Interval

and loss rate. On the other hand, constant listening has better communication performance. However, it is more energy consuming and thus undesirable in view of energy efficiency. In system optimization, it is a waste to spend extra energy on achieving communication performance than needed. Therefore, how to properly tune the sleep window to conserve energy while still meet the QoS requirement is a challenging issue. To get the optimal tuning parameters, in this section, we give the mathematical analysis on the system performance.

A. Average Talk-spurt Delay

Talk-spurt delay in this paper is defined as the period from the time the first talk-spurt packet reaches the buffer at the BS to the time the MS begins the reception of this talk-spurt packets. We assume that, in the silence period, the sleep window T_i ($i \geq 1$) is adjusted according to (1).

We also specify that $T_{\max} = 2^K T_1$, where K is the maximum sleep window index, which is also the maximum value that i can reach. Without loss of generality, let t_0 be the beginning time of a silence period, which is also the starting point of the first sleep window. For the MS, the wakeup time for checking the downlink traffic is t_i , where $i > 0$ and $t_i - t_{i-1} = T_i$. If downlink traffic arrives at time t , where $t \in [t_{i-1}, t_i]$, the talk-spurt delay can be written as $t_i - t$, because MS must sleep until t_i to check the downlink traffic indication as illustrated in Fig.2. As the sleep window size T_i is doubled each time when the traffic indication flag is negative, t_i can be express as:

$$t_i = \begin{cases} 0, & i = 0 \\ T_1 \sum_{j=0}^{i-1} 2^j, & 1 \leq i \leq K+1 \\ T_1 \sum_{j=0}^K 2^j + (i-K-1)T_{\max}, & i > K+1 \end{cases} \quad (2)$$

According to the speech model mentioned above, the PDF for the silence period can be written as $\frac{1}{T_s} e^{-\frac{t}{T_s}}$. The average delay for the talk-spurt $E[D]$ can be calculated by:

$$E[D] = \sum_{i=0}^K \int_{t_i}^{t_{i+1}} \frac{1}{T_s} e^{-\frac{t}{T_s}} (t_{i+1} - t) dt + \sum_{i=0}^{\infty} \int_{t_{K+1} + iT_{\max}}^{t_{K+1} + (i+1)T_{\max}} \frac{1}{T_s} e^{-\frac{t}{T_s}} (t_{K+1} + (i+1)T_{\max} - t) dt \quad (3)$$

The first integral in Equation (3) can be written as $(t_{i+1} - t_i) e^{-\frac{t_i}{T_s}} + T_s (e^{-\frac{t_{i+1}}{T_s}} - e^{-\frac{t_i}{T_s}})$. The second integral in equation (3) can be expressed as

$$T_{\max} e^{-\frac{t_{K+1} + iT_{\max}}{T_s}} + T_s (e^{-\frac{t_{K+1} + (i+1)T_{\max}}{T_s}} - e^{-\frac{t_{K+1} + iT_{\max}}{T_s}})$$

Combining these two integrals, $E[D]$ can be re-written as:

$$\begin{aligned} E[D] &= \sum_{i=0}^K \underbrace{(t_{i+1} - t_i)}_{=T_{i+1}=2^i T_1} e^{-\frac{t_i}{T_s}} + T_s \sum_{i=0}^K \underbrace{(e^{-\frac{t_{i+1}}{T_s}} - e^{-\frac{t_i}{T_s}})}_{=-e^{-\frac{t_0}{T_s} + e^{-\frac{t_{K+1}}{T_s}} = e^{-\frac{t_{K+1}}{T_s} - 1}} \\ &+ T_{\max} e^{-\frac{t_{K+1}}{T_s}} \sum_{i=0}^{\infty} e^{-\frac{iT_{\max}}{T_s}} + T_s e^{-\frac{t_{K+1}}{T_s}} \sum_{i=0}^{\infty} \underbrace{(e^{-\frac{(i+1)T_{\max}}{T_s}} - e^{-\frac{iT_{\max}}{T_s}})}_{=-e^0 = -1} \\ &= \sum_{i=0}^K 2^i T_1 e^{-\frac{t_i}{T_s}} + T_s (e^{-\frac{t_{K+1}}{T_s}} - 1) \\ &+ T_{\max} e^{-\frac{t_{K+1}}{T_s}} (1 - e^{-\frac{T_{\max}}{T_s}})^{-1} - T_s e^{-\frac{t_{K+1}}{T_s}} \\ &= \sum_{i=0}^K 2^i T_1 e^{-\frac{t_i}{T_s}} + T_{\max} e^{-\frac{t_{K+1}}{T_s}} (1 - e^{-\frac{T_{\max}}{T_s}})^{-1} - T_s \end{aligned} \quad (4)$$

Where T_1, T_{\max} can be calculated from equation (1), and t_{K+1} can be calculated from (2).

B. Loss Rate

When the MS is in sleep state during the silence period, the coming downlink VoIP traffic must be buffered at the BS. However, there is delay constraint imposed on the VoIP stream because: (1) buffer capacity at the BS for each MS is limited; and (2) long buffering time can render packet dropping due to delivery expiration. Therefore, we assume that buffer capacity allocated to each MS is B , i.e., the BS can only buffer B seconds' voice stream for the MS.

Again, let $t_1, t_2, t_3, t_4, \dots$ be the downlink traffic checking points, and the starting and ending time for the talk-spurt be t and t' , respectively. For the analysis of the stream loss, we have three cases as illustrated in Fig.3:

- Case 1: if $t \in [t_i - B, t_i]$, all the coming VoIP packets during this interval are buffered and there is no stream loss;
- Case 2: if $t \in [t_{i-1}, t_i - B]$ and $t' \in [t_{i-1}, t_i - B]$, all the coming VoIP packets are dropped due to expiration;
- Case 3: if $t \in [t_{i-1}, t_i - B]$ and $t' \in [t_i - B, \infty)$, the portion of being dropped due to expiration is $t_i - B - t$.

Let $j = \min \arg \{i \mid T_i > B\}$. For Case 2, the loss rate can be calculated as:

$$\begin{aligned} E(L_1) &= \sum_{i=j}^{\infty} \int_{t_{i-1}}^{t_i - B} \underbrace{\frac{1}{T_s} e^{-\frac{t}{T_s}}}_{\text{means } t \in [t_{i-1}, t_i - B]} dt \int_{t_{i-1}}^{t_i - B} \underbrace{\frac{1}{T_s} e^{-\frac{t'}{T_s}}}_{\text{means } t' \in [t_{i-1}, t_i - B]} \underbrace{(t' - t) dt'}_{\text{lost portion}} \\ &= \sum_{i=j}^{\infty} \int_{t_{i-1}}^{t_i - B} \left(\frac{T_s}{T_s} e^{-\frac{t}{T_s}} - \frac{t_i - B - t + t_{i-1}}{T_s} e^{-\frac{t}{T_s}} \frac{t_{i+1} - B - t}{T_s} \right) dt \end{aligned} \quad (5)$$

For Case 3, the loss rate can be calculated as:

$$E(L_2) = \sum_{i=j}^{\infty} \int_{t_{i-1}}^{t_i - B} \underbrace{\frac{1}{T_s} e^{-\frac{t}{T_s}}}_{\text{means } t \in [t_{i-1}, t_i - B]} \underbrace{e^{-\frac{t_i - B - t}{T_s}}}_{\text{means } t' - t > t_i - B - t} \underbrace{(t_i - B - t) dt}_{\text{lost portion}} \quad (6)$$

Therefore, the total average loss rate $E(L)$ is the combination of $E(L_1)$ and $E(L_2)$, i.e.

$$E(L) = E(L_1) + E(L_2) \quad (7)$$

C. Average Number of Invalid Wakeup

If the MS wakes up and receives negative traffic indication, this wakeup is invalid. To reduce energy consumption due to invalid wakeup, in the proposed hybrid scheme, the sleep window is increased exponentially. The average number of invalid wakeup $E(N_w)$ in one silence period can be expressed by:

$$\begin{aligned} E[N_w] &= \sum_{i=0}^K i \int_{t_i}^{t_{i+1}} \frac{1}{T_s} e^{-\frac{t}{T_s}} dt \\ &+ \sum_{i=0}^{\infty} (K+i+1) \int_{t_{K+1}+iT_{\max}}^{t_{K+1}+(i+1)T_{\max}} \frac{1}{T_s} e^{-\frac{t}{T_s}} dt \\ &= \sum_{i=0}^K i \left(e^{-\frac{t_i}{T_s}} - e^{-\frac{t_{i+1}}{T_s}} \right) \\ &= \sum_{i=1}^K e^{-\frac{t_i}{T_s}} - Ke^{-\frac{t_{K+1}}{T_s}} \\ &+ \sum_{i=0}^{\infty} (K+i+1) \left(e^{-\frac{t_{K+1}+iT_{\max}}{T_s}} - e^{-\frac{t_{K+1}+(i+1)T_{\max}}{T_s}} \right) \\ &= -e^{-\frac{t_{K+1}}{T_s}} (1 - e^{-\frac{T_{\max}}{T_s}}) \left[K \sum_{i=0}^{\infty} e^{-\frac{iT_{\max}}{T_s}} + \sum_{i=0}^{\infty} (i+1) e^{-\frac{iT_{\max}}{T_s}} \right] \\ &= \sum_{i=1}^K e^{-\frac{t_i}{T_s}} + e^{-\frac{t_{K+1}}{T_s}} (1 - e^{-\frac{T_{\max}}{T_s}})^{-1} \end{aligned} \quad (8)$$

IV. NUMERICAL RESULTS AND PERFORMANCE ANALYSIS

The tuning of the sleep window size significantly influences the performance of the hybrid scheme in the QoS provision and energy consumption. In this section, we change the initial and maximum sleep window, and present the numerical results on average talk-spurt delay, loss rate, and number of invalid wakeup. We hope to spend the least amount of energy just sufficient to meet the QoS requirements. To validate the accuracy of performance analysis in Section III, we also perform extensive simulations.

According to the QoS specifications in ITU document G.114 [10], for satisfactory transmission quality, the maximum end-to-end delay of VoIP should not exceed 250 ms. The factors contributing to the end-to-end delay include speech processing delay, codec delay, and backbone transmission delay. Therefore, in system design, we should leave some redundancy for these delay components and require that the average talk-spurt delay should not exceed 200 ms. In addition, to avoid severe speech quality distortion at MS, the maximum packet loss rate must be less than 10%.

At the transmitting side, the speech is sampled, coded, and encapsulated into IP streaming packets according to the format specifications. We assume that, during the talk-spurt period, the VoIP packet is generated for every 20 ms. Accordingly, during this active duration, the MS should also wake up every 20 ms to receive the downlink traffic. In the silence period, the sleep window is adjusted exponentially according to equation (1). After the sleep window, the MS wakes up to receive the downlink traffic

TABLE 1 SIMULATION PARAMETERS

Parameter	Value
Listening Power	0.5 W
Sleep Power	0.05W
Fame Length	5 ms
Mean Talk-spurt Period	1.004 s
Mean Silence Period (Base)	1.587 s
Delay Tolerance	< 200ms
Loss Tolerance	< 10%
Buffering Length at BS	100 ms
Initial Battery Energy Level	1000 Joule

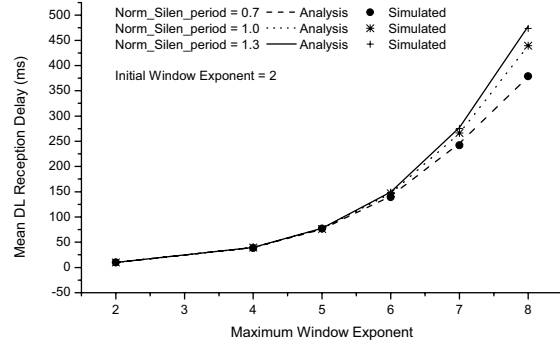


Fig.4 Mean Talk-spurt Delay vs. Maximum Sleep Window Exponent (Initial Window Exponent = 2)

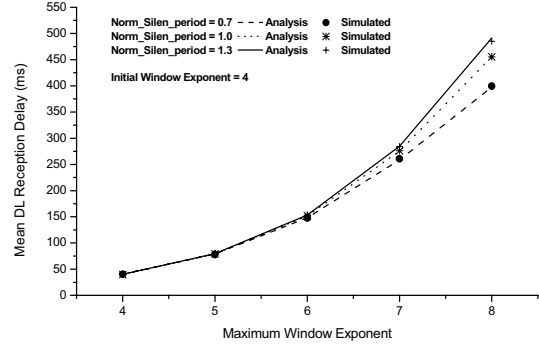


Fig.5 Mean Talk-spurt Delay vs. Maximum Sleep Window Exponent (Initial Window Exponent = 4)

indication broadcast from the BS. We assume that the reception of the traffic indication needs 1 ms. If negative indication is presented, the MS does not need to receive the rest frame and goes directly into sleep state. Otherwise if positive indication is presented, the MS should keep active for the whole frame length (5 ms) to receive its downlink packets. The physical parameters for analysis and simulation are listed in TABLE 1.

A. Average Talk-spurt Delay

Fig.4 and Fig.5 are the mean downlink talk-spurt delay versus the varied maximum sleep window exponent, with the initial window exponents 2 and 4, respectively.

The initial sleep window T_1 in equation (1) is calculated by:

$$\begin{aligned} & \text{Initial Sleep Window} \\ &= \text{Frame Length} \times 2^{\text{Initial Window Exponent}} \end{aligned} \quad (9)$$

The maximum/final sleep window T_{\max} is calculated by:

$$\begin{aligned} \text{Maximum Sleep Window} \\ = \text{Frame Length} \times 2^{\text{Maximum Window Exponent}} \end{aligned} \quad (10)$$

Thus, K in equation (2) can be expressed as:

$$\begin{aligned} K = \text{Maximum Window Exponent} \\ - \text{Initial Window Exponent} \end{aligned} \quad (11)$$

According to the above equations, the initial sleep window in Fig.4 and Fig.5 is 20 ms and 80 ms, respectively, while the maximum sleep window in both figures is 1280 ms. Note that the average duration of the silence period can vary in terms of language, gender, or even individual personality. In our simulations, we change the silence duration by multiplying the Mean Silence Period in TABLE 1 by "Norm_Silen_period," which reflects the speed of natural human speech.

In both figures, we can see that the simulated results match well with the analytical results, which validates the accuracy of the delay model in Section 3. The mean delay increases with the maximum window. This is natural because the MS spends more time on sleeping, resulting in more delay in the traffic reception. As expected, for the same maximum window exponent, the mean delay increases with the silence period due to the increase in the interval of two consecutive talk-spurts. Note that in the system model, if initial window exponent equals to maximum window exponent (the first point in each plot), the hybrid scheme will become PSC-II, i.e., sleep interval is constant. This, of course, will result in the shortest delay. However, as illustrated in latter section, more energy will be consumed due to more frequent listening.

As the BS can buffer stream for 100 ms, to reduce unnecessary wakeup, we further increase the initial window exponent from 2 to 4, which corresponds to the first sleep window of 80 ms (exponent of 5 or above is not selected because it might render stream loss in exactly the first sleep window). It is observed that the mean delay has slight increase when the initial sleep window is prolonged, as shown in Fig.5.

In both figures, a valuable observation is that, if mean talk-spurt delay of less than 200 ms can be satisfied, the maximum window exponent should not exceed 6, i.e., maximum sleep window $T_{\max} = 320 \text{ ms}$.

B. Stream Loss Rate

Fig.6 and Fig.7 are the mean stream loss rate versus the maximum sleep window. Again, the simulation results fit quite well with the analytical lines. We can observe that a larger maximum window exponent can lead to more stream loss due to longer sleeping period. Similarly, the increase in the silence period can incur more loss, which is more obvious when the maximum window exponent is large. In this paper, we have set the buffering length to 100 ms, thus no stream is lost if the maximum window exponent does not exceed 4. Comparing Fig.7 with Fig.6, we find that increasing initial window exponent from 2 to 4 has slight influence on the loss rate. It is illustrated in Fig.6 and Fig.7 that, to keep loss rate less than 10%, the maximum window exponent should not exceed 6, or equivalently, the final sleep window is 320 ms.

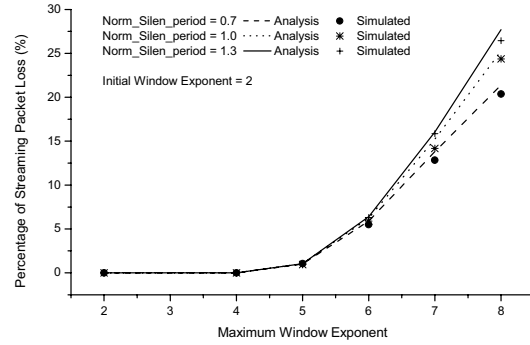


Fig.6 Loss Rate vs. Maximum Sleep Window Exponent (Initial Window Exponent = 2)

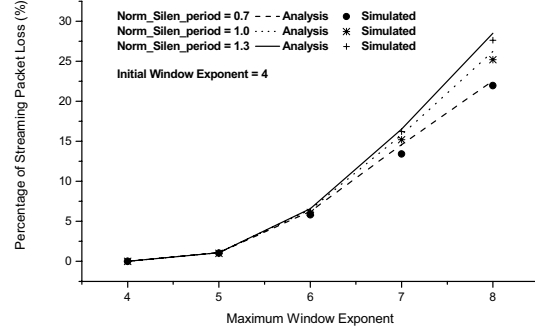


Fig.7 Loss Rate vs. Maximum Sleep Window Exponent (Initial Window Exponent = 4)

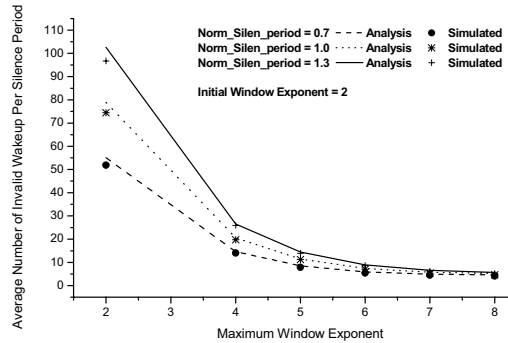


Fig. 8 Number of Invalid Wakeup vs. Maximum Sleep Window Exponent (Initial Window Exponent = 2)

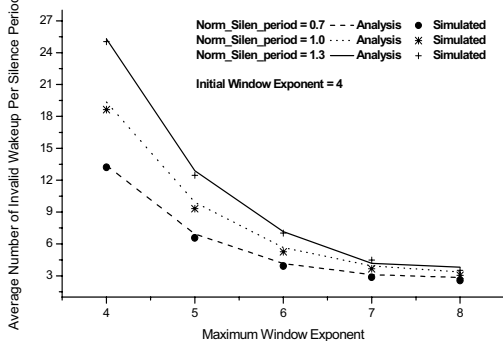


Fig.9 Number of Invalid Wakeup vs. Maximum Sleep Window Exponent (Initial Window Exponent = 4)

C. Number of Invalid Wakeup per Sleep Period

We calculate the average number of invalid wakeup according to equation (9) and perform simulations to validate the accuracy. These results are plotted in Fig.8 and Fig.9. From the figures, it is shown that, with the increase

of the maximum window exponent, the number of invalid wakeup significantly decreases because the sleeping period is prolonged. For the same maximum window exponent, the extension in the traffic silence period results in more invalid wakeups, because the MS more frequently probes the traffic arrival. In Fig.9, the initial window exponent increases from 2 to 4, and it is observed that the number of invalid wakeup is considerably reduced. Thus, an initial exponent of 4 can further enhance the energy efficiency.

D. Energy Saving Effects

From the above discussion, we know that, to provide satisfactory QoS for the downlink VoIP, the maximum window exponent should be set to 6. In addition, as the BS has a buffering length of 100 ms for each VoIP stream, the initial sleep window exponent can be set to 4 (corresponding to the sleep window size of 80 ms). Therefore, given the prescribed QoS requirement, to conserve the energy to the best effort, [Initial Window Exponent = 4, Maximum Window Exponent = 6] is the optimal working parameter for the hybrid scheme.

To demonstrate the energy conserving effect, we use PUS-II as the baseline. As the packet is generated every 20 ms in the speech active period, to fit this arrival rate, the constant sleep interval in PSC-II is also set to 20 ms.

Fig.10 is the percentage of energy saved for listening to the downlink traffic indication in the speech silence period versus the normalized silence period. We can see that the hybrid scheme can reduce energy wasted in unnecessary wakeup for about 89%-94%. The gain is more obvious with

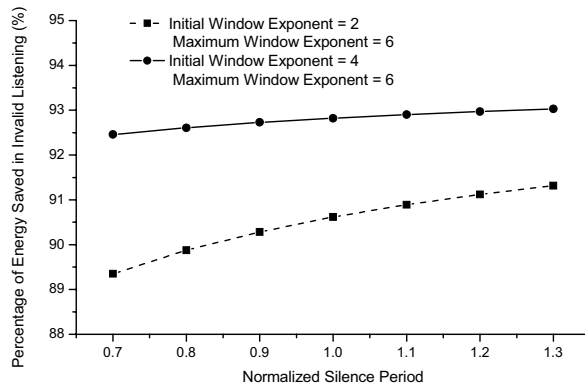


Fig.10 Percentage of Energy Saves in Listening vs. Normalized Silence Period

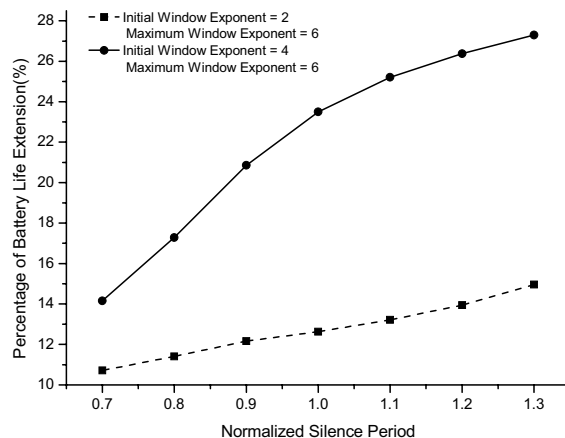


Fig.11 Battery Life Extension vs. Normalized Silence Period

the increase of the speech silence period.

To see more clearly on the effect of energy conservation, we perform simulations to test how long the battery life can be extended with the hybrid scheme. The initial battery capacity and the energy consumptions in different states are listed in TABLE 1. Fig.11 is the life extension versus normalized silence period. Due to the reduction in unnecessary listening in the speech silence period, compared with traditional PSC-II, the battery life in the hybrid scheme can be extended for about 11%-20%. The improvement is more obvious with longer silence period. This proves the effectiveness of the hybrid scheme.

V. CONCLUSION

In this paper, we adopted a hybrid energy conservation scheme for the downlink simplex VoIP in WiMAX system. We have also given an analysis model to evaluate the system performance. Guided by the model, we obtain the optimal window adjustment parameters, with which the battery life can be extended for about 11-20%, while the tough QoS requirements can still be satisfied. Therefore, by adopting the hybrid scheme, we can strike a proper balance between energy conservation and QoS provision. In addition, for the hybrid scheme, no major revision is required on the original power saving strategies, and hence, it is well compatible with the state-of-art industrial platforms and products.

Acknowledgement: The research was jointly supported by research grant from Natural Science Foundation of China under project number 60602066 and 60773203, and grants from Foundation of Shenzhen City under project number JC200903120069A, SG200810220145A, JC201005250035A, and JC201005250047A. Corresponding author of the paper: Dr. Hui Wang {wanghsz@szu.edu.cn}

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