On the Use of a Power-Saving Mode for Mobile VoIP Devices and Its Performance Evaluation

Hyun-Ho Choi, *Member, IEEE*, Jung-Ryun Lee, *Member, IEEE*, and Dong-Ho Cho, *Senior Member, IEEE*

***Abstract*—The widespread use of mobile devices in the IP network has lead to a new attempt to apply a power-saving mode (PSM) to real-time traffic such as Voice over IP (VoIP). This paper evaluates the performance of the PSM when the PSM is used for VoIP services of mobile devices. Taking the activity of each conversational party into account, we consider two different kinds of PSMs: one is employed during the talk-spurt periods and the other is employed during the mutual silence periods of two conversational parties. The performance of each PSM is analyzed with respect to buffering delay, the probability of packet drop, and power consumption of a mobile VoIP device. Thereafter, the maximum bound of sleep interval in each period is derived, which minimizes the power consumption of the mobile device without violating the quality-of-service (QoS) of VoIP. In the various network environments, the analysis and simulation results show that the proposed PSM for VoIP significantly decreases the power consumption while satisfying the end-to-end delay and packet drop probability constraints of a VoIP connection.**

***Index Terms*—Power-saving mode, sleep interval, voice over IP (VoIP), quality of service (QoS).**

I. Introduction

MOBILE devices (cellular phones, laptops, PDAs, etc.) usually rely on portable power sources, such as batteries. Since batteries provide a limited amount of energy, it is important to design efficient power-saving mechanisms to prolong their lifetime. Most wireless systems adopt a power-saving mode (PSM) to reduce the energy consumption of mobile stations (MS). The common approach of PSM is *discontinuous reception*, that is, MSs periodically power off their reception units (go into a sleep mode) instead of continuously listening to the radio channel. Since an MS in sleep mode wakes up only at predefined listening intervals, it can conserve much of its energy [1]-[3].

Recently, voice over Internet protocol (VoIP) services in wireless networks are widely spreading as service providers try to offer wireless multimedia services based on the All IP infrastructure, and customers want to enjoy voice communications at low cost. The VoIP protocol carries voice packets with a reduced data rate using speech data compression techniques, and voice packets are sent by multiplexing with other data packets over the IP, so the transmission efficiency is improved and the communication cost is reduced. Because of these advantages, typical wireless access systems, such as EVDO, WiFi, and WiMAX, use the VoIP protocol to offer telephony services [4]-[6].

As the majority of services that mobile devices use in wireless telecommunication environments are currently voice and the power-saving function is indispensable to mobile devices, the current wireless access systems provide PSMs for VoIP services. The IEEE 802.16e mobile WiMAX specifies the *Power-Saving Class of Type 2* and the IEEE 802.11e WLAN provides *Automatic Power Save Delivery* as their power management schemes for real-time VoIP services [6], [7]. In these PSM schemes, an MS repeats sleep and listening alternately according to the generation interval of VoIP packets. Its sleep and listening intervals are constant because the VoIP packet arrives with a certain periodicity.

It has been recognized that there is a tradeoff between the energy conservation of an MS and the quality-of-service (QoS) performance of transmitted packets [8]. In the PSM for VoIP, the same tradeoff happens according to the length of the sleep interval. If the sleep interval is equal to the packet generation interval of the VoIP codec with a short range of 10 to 30 ms [9], it is sufficient to satisfy VoIP QoS, such as end-to-end delay and packet loss rate, but it is difficult to acquire enough power-saving gain due to the relatively short sleep interval. On the contrary, if the length of the sleep interval is set to a larger value, more power is saved, but it causes an additional delay to VoIP traffic, and so may not satisfy VoIP QoS. Therefore, in the PSM for VoIP, it is crucial to determine the length of the sleep interval by taking into account the performance of PSM and the QoS requirements of VoIP.

In this paper, we investigate the PSM for mobile VoIP devices in wireless networks. We evaluate the performance of the VoIP PSM and derive a theoretical maximum bound of sleep interval that minimizes the total power consumption of an MS while still guaranteeing VoIP QoS. Our approach considers two states of VoIP communication: talk-spurt and mutual silence[1]. Since there is no packet transmission between

Hyun-Ho Choi is with Communication Lab., Samsung Advanced Institute of Technology (SAIT), Yongin, Republic of Korea (e-mail: hyun.ho.choi@samsung.com).

Jung-Ryun Lee is with the School of the Electrical Engineering, Chung-Ang University, Seoul, Republic of Korea (e-mail: jrlee@cau.ac.kr).

Dong-Ho Cho is with Department of Electrical Engineering and Computer Science, Korea Advanced Institute of Science and Technology (KAIST), Daejeon, Republic of Korea (e-mail: dhcho@ee.kaist.ac.kr).

[1]Enhanced voice codecs use a silence suppression scheme that prevents voice packets from being transmitted during silent periods [10]-[12]. It is known that silent periods occupy about 60 percent of the total duration of a VoIP call and mutual silence periods (i.e., both caller and callee are silent) occupy about 20 percent [13].

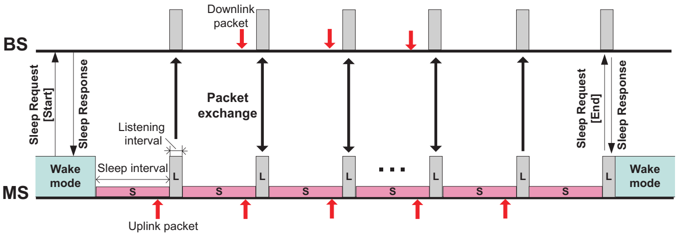

Fig. 1. Operation of power-saving class of type 2 in IEEE 802.16e.

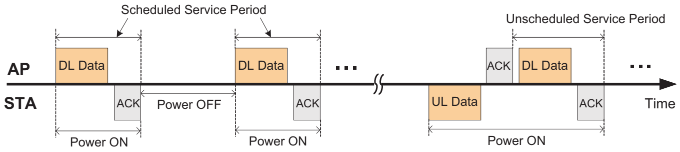

Fig. 2. Operation of automatic power save delivery in IEEE 802.11e.

an MS and a BS (base station) in the mutual silence period, the sleep interval in the mutual silence period can be longer than that in the talk-spurt period.

The rest of this paper is organized as follows. In Section II, the standards and literature related to the PSM for VoIP are introduced. In Sections III and IV, PSMs in the talk-spurt period and in the mutual silence period are analyzed, respectively. In Section V, some implementation issues are stated. In Section VI, analysis and simulation results are presented. Section VII concludes this paper.

II. Related Work

The IEEE 802.16e standard defines the *power-saving class of type 2* (PSC II) for the connection of real-time variable rate data transfer, such as VoIP and video-streaming [6]. Fig. 1 illustrates a basic sleep mode operation of PSC II. To employ PSC II, two parameters, the sleep interval and the listening interval, need to be determined. Since real-time packets arrive periodically, the sleep and listening intervals in every sleep cycle have constant length and they are repeated alternatively throughout PSC II operation. During the listening interval, the MS and BS exchange their real-time packets with each other.

The IEEE 802.11e standard specifies *automatic power save delivery* (APSD) as an enhanced power management scheme for real-time services [7]. Fig. 2 shows the basic operation of the APSD. There are two types of APSD: scheduled APSD and unscheduled APSD. In the scheduled APSD, pre-arranged wake-up times allow the access point (AP) to deliver packets buffered for the station. Receiving buffered packets from the AP at a scheduled time reduces the station's wake time spent synchronizing with the system and contending for the channel with other stations. In the unscheduled APSD, the receipt of a packet from the station indicates that the station is still awake to receive packets buffered at the AP. Accordingly, the station can receive downlink packets immediately after sending uplink packets without any transmission of a power save-poll (PS-Poll) control frame.

Chen *et al.* [14] proposed a new power management mechanism named *unscheduled power-save delivery* (UPSD) for VoIP services in the IEEE 802.11 WLAN and evaluated its performance considering both a voice only scenario and a voice in the presence of data traffic scenario. Pérez-Costa *et al.* [15] mentioned the problem that the 802.11 legacy PSM can result in downlink delays unacceptable for the QoS of VoIP and proposed an *adaptive power save mode* (APSM) algorithm, which adapts the PS-Poll sending interval to the inter-arrival time of the downlink data frame. Thus, the APSM is effective on power-saving while satisfying stringent requirements of applications. Shih *et al.* [16] presented an *on-demand polling* (ODP) scheme as a power-efficient MAC protocol. The AP periodically polls the stations according to the polling list during talk-spurt periods, but the station is removed from the polling list during silent periods, so the station consumes less power. Tsao *et al.* [17] suggested a transmission scheme that dynamically eliminates the WLAN MAC acknowledgement frames of real-time packets in order to reduce the power consumption of video and voice over WLAN stations.

III. Power Saving Mode for Talk-spurt Period

First, we make an end-to-end delay model of the VoIP packet considering a general packet-based IP network environment and then analyze the PSM in the talk-spurt period.

A. End-to-end Delay Model

We establish an end-to-end delay model to describe each delay attribute that the VoIP packet experiences during delivery via the network. Fig. 3 illustrates the considered end-to-end delay model for VoIP traffic[2]. On the basis of [18], a VoIP call's end-to-end delay can be split into several terms as follows:

$$\begin{aligned} D_{e2e} &= D_{enc} + D_{pack} + \sum_{r=1}^{R}(D_{mnet} + D_{que}^{r}) \\ &+ D_{buf} + D_{play} + D_{dec} \end{aligned} \tag{1}$$

where R is the number of routers in the packet transmission path and each delay term has the following meaning:

- D_{enc}: voice-encoding delay by the voice coder at the source.
- D_{pack}: packetization delay needed to fill a packet payload with encoded/compressed speech.
- D_{mnet}: minimal network delay in each network router, which consists mainly of the propagation delay (5 microseconds per kilometer), the transmission delay (which depends on the packet size and the link bandwidth), and the route lookup delay.
- D_{que}^{r}: queueing delay in the r-th router.
- D_{buf}: packet buffering delay in BS or MS.
- D_{play}: playout delay (or dejitter delay) required by decoder for smoothing the inter-arrival time of the voice packets.

[2]In this paper, we regard a VoIP call as a land-to-mobile call in which the other side is a VoIP telephone connected to the wired packet-based network, but this model can be extended to accommodate mobile-to-mobile calls without difficulty by having both ends be part of a symmetric wireless network.

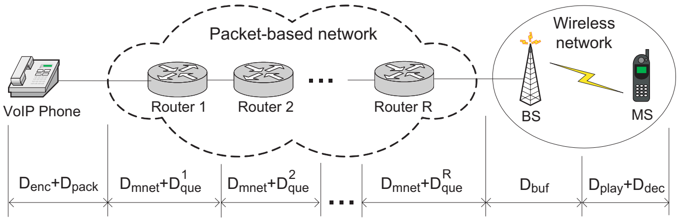

Fig. 3. End-to-end delay model for VoIP traffic.

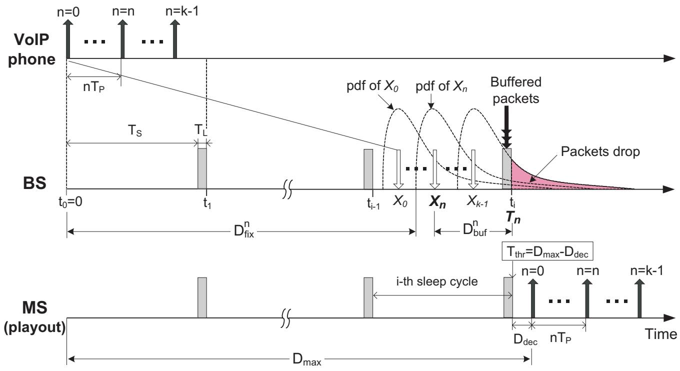

Fig. 4. Analytic model for PSM in talk-spurt period.

- D_{dec}: voice-decoding delay by the voice decoder at the destination.

Each term in the end-to-end delay model can be divided into fixed parts and variable parts. We can assume that D_{enc}, D_{dec}, D_{pack} and D_{mnet} are fixed delays, because the former three delays have a constant value decided by the type of codec used, and the latter D_{mnet} delay has a small variation compared with the complete end-to-end delay. Among the variable delay terms, D_{que}^r occupies the greatest part of the end-to-end delay and follows an exponential distribution judging from empirical measurements [19]. D_{buf} is a buffering delay generated by the power-saving operation. D_{play} is decided according to the delay variance.

In this end-to-end delay model, we are interested in how much delay a VoIP packet arriving at the BS experiences. Let X be the distribution of delay between the VoIP phone and the BS. Then X is expressed as

$$\begin{aligned} X &= D_{enc} + D_{pack} + \sum_{r=1}^{R}(D_{mnet} + D_{que}^r) \\ &= \sum_{r=1}^{R} D_{que}^r + D_{fix} = Hypo(R) + D_{fix} \end{aligned} \tag{2}$$

where D_{fix} is a fixed delay given by $D_{enc} + D_{pack} + \sum_{r=1}^{R} D_{mnet}$. That is to say, X follows a hypoexponential distribution composed of R exponential distributions with different rates in series. This is reasonable as several studies have verified experimentally that the network propagation delay conforms to a heavy-tailed asymmetric distribution (i.e., hypoexponential distribution) with a certain constant delay [19]-[21]. The probability density function (pdf) of X is given by

$$f_X(t) = \begin{cases} \sum_{r=1}^{R} C_r \lambda_r e^{-\lambda_r(t-D_{fix})} & \text{if } t \geq D_{fix} \\ 0 & \text{if } t < D_{fix} \end{cases} \tag{3}$$

where λ_r is a rate parameter of exponential distribution D_{que}^r and C_r is a constant given by $\prod_{q \neq r} \frac{\lambda_q}{\lambda_q - \lambda_r}$.

B. Analysis in Talk-spurt Period

Considering the periodicity of VoIP packet generation, we make an approach to control the length of the sleep cycle as a multiple of the generation interval of the VoIP packet, as follows:

$$T_S + T_L = k \cdot T_P \tag{4}$$

where T_S and T_L are the length of the sleep interval and listening interval, respectively, T_P is a packet generation interval specified by the VoIP codec, and k is a positive integer. In most of the PSMs, the listening interval, T_L, is set to a short constant value considering the system capacity and the VoIP packet size in order to reduce energy consumption as much as possible. Therefore, if we find the maximum k that satisfies VoIP QoS (i.e., end-to-end delay and packet loss rate), we can know the maximum sleep interval.

To analyze the PSM in the talk-spurt period, we construct a simplified analytic model as shown in Fig. 4. This model describes the transmission process of voice packets in the VoIP phone, BS and MS, according to the time axis. Here, we assume that the routing path of both downlink and uplink packets is symmetric and so consider only the downlink case. Since the sleep cycles with equal length are repeated continuously, we are allowed to consider only one sleep cycle with the length of kT_P and only k voice packets generated during kT_P. When the start time is zero, the VoIP phone transmits a voice packet at time nT_P ($n = 0, 1, 2, \cdots, k-1$). Using (2), the distribution of the arrival time of the n-th voice packet, X_n, is expressed as

$$\begin{aligned} X_n &= X + nT_P \\ &= \sum_{r=1}^{R} D_{que}^r + D_{fix} + nT_P \\ &= \sum_{r=1}^{R} D_{que}^r + D_{fix}^n \end{aligned} \tag{5}$$

where $D_{fix}^n = D_{fix} + nT_P$. So, X_n follows a hypoexponential distribution with fixed delay D_{fix}^n. From (3), the pdf of X_n becomes

$$f_{X_n}(t) = \begin{cases} \sum_{r=1}^{R} C_r \lambda_r e^{-\lambda_r(t-D_{fix}^n)} & \text{if } t \geq D_{fix}^n \\ 0 & \text{if } t < D_{fix}^n. \end{cases} \tag{6}$$

1) Buffering Delay: We define the i-th sleep cycle in the BS as $[t_{i-1}, t_i]$, where $t_i = i(T_S + T_L)$. Then we can calculate the probability that the n-th voice packet arrives in the i-th sleep cycle, p_i^n, as follows.

$$\begin{aligned} p_i^n &= P\{X_n \in [t_{i-1}, t_i]\} \\ &= \int_{t_{i-1}}^{t_i} f_{X_n}(t)dt \end{aligned}$$

$$
\begin{aligned}
&= \int_{t_{i-1}}^{t_i} \sum_{r=1}^{R} C_r \lambda_r e^{-\lambda_r (t - D_{fix}^n)} dt \\
&= \left[-\sum_{r=1}^{R} C_r e^{-\lambda_r (t - D_{fix}^n)} \right]_{t'_{i-1}=max(D_{fix}^n, t_{i-1})}^{t'_i = max(D_{fix}^n, t_i)} \\
&= \sum_{r=1}^{R} C_r \left\{ e^{-\lambda_r (t'_{i-1} - D_{fix}^n)} - e^{-\lambda_r (t'_i - D_{fix}^n)} \right\} \quad (7)
\end{aligned}
$$

where we define $t'_i = max(D_{fix}^n, t_i)$.

The voice packet arriving in the i-th sleep cycle is transmitted to the MS in the i-th listening interval. Let T_n be the point in time when the n-th voice packet is delivered to the MS, as shown in Fig. 4. Then, the average T_n is calculated by

$$
\begin{aligned}
E[T_n] &= \sum_{i=1}^{\infty} p_i^n \cdot t_i \\
&= \sum_{i=1}^{\infty} p_i^n \cdot i(T_S + T_L) \\
&= \sum_{i=1}^{\infty} i(T_S + T_L) \\
&\quad \cdot \sum_{r=1}^{R} C_r \left\{ e^{-\lambda_r (t'_{i-1} - D_{fix}^n)} - e^{-\lambda_r (t'_i - D_{fix}^n)} \right\} \quad (8)
\end{aligned}
$$

The average arrival time of the n-th voice packet is given by

$$E[X_n] = \sum_{r=1}^{R} \frac{1}{\lambda_r} + D_{fix}^n. \quad (9)$$

Suppose the n-th voice packet generated at the VoIP phone arrives in the i-th sleep cycle of the BS. Since this packet must be delivered to the MS in the i-th listening interval, the buffering delay of the n-th voice packet is represented by $D_{buf}^n = T_n - X_n$. From (8) and (9), the average buffering delay of the n-th voice packet is expressed as

$$E[D_{buf}^n] = E[T_n] - E[X_n]. \quad (10)$$

Finally, the average buffering delay of all voice packets in the talk-spurt period is calculated by

$$D_{buf,T} = \frac{1}{k} \sum_{n=0}^{k-1} E[D_{buf}^n]. \quad (11)$$

2) Drop Probability: Assuming that there is no packet transmission error in the wired or wireless link, we may consider only the packet loss by PSM operation. Therefore, the QoS constraint of VoIP is expressed as [22]

$$P_{drop} = Pr\{D_{e2e} > D_{max}\} \leq \delta \quad (12)$$

where D_{max} is a maximum tolerable end-to-end delay and δ is a maximum tolerable packet drop rate.

To prevent voice packets from being dropped, the MS must receive them before the threshold time $T_{thr} = D_{max} - D_{dec}$, as shown in Fig. 4. Let m be the largest integer satisfying $m(T_S + T_L) \leq T_{thr}$, then the BS must deliver voice packets to the MS by the m-th listening interval to prevent packet drops. Thus, $t_m = m(T_S + T_L)$ becomes the threshold time point at the BS, and all voice packets that arrive at the BS after t_m are dropped. Therefore, the drop probability of the n-th voice packet is given by

$$
\begin{aligned}
P_{drop}^n &= \int_{t_m}^{\infty} f_{X_n}(t) dt \\
&= \int_{t_m - nT_P}^{\infty} f_X(t) dt \\
&= \int_{t_m - nT_P}^{\infty} \sum_{r=1}^{R} C_r \lambda_r e^{-\lambda_r (t - D_{fix})} dt \\
&= \sum_{r=1}^{R} C_r e^{-\lambda_r (t_m - nT_P - D_{fix})}. \quad (13)
\end{aligned}
$$

The average drop probability of all voice packets in the talk-spurt period is given by

$$
\begin{aligned}
P_{drop,T} &= \frac{1}{k} \sum_{n=0}^{k-1} P_{drop}^n \\
&= \frac{1}{k} \sum_{n=0}^{k-1} \sum_{r=1}^{R} C_r e^{-\lambda_r (t_m - nT_P - D_{fix})} \\
&= \frac{1}{k} \sum_{r=1}^{R} C_r e^{-\lambda_r (t_m - D_{fix})} \sum_{n=0}^{k-1} e^{\lambda_r n T_P} \\
&= \frac{1}{k} \sum_{r=1}^{R} C_r e^{-\lambda_r (t_m - D_{fix})} \frac{1 - e^{k \lambda_r T_P}}{1 - e^{\lambda_r T_P}}. \quad (14)
\end{aligned}
$$

Since $P_{drop,T}$ is an equation of unknown k, we can obtain a maximum k value, k_{max}, as solving the inequality $P_{drop,T} \leq \delta$. Let $T_{Smax,T}$ be the maximum sleep interval for the talk-spurt period. From (4), $T_{Smax,T}$ is determined by

$$T_{Smax,T} = k_{max} \cdot T_P - T_L. \quad (15)$$

3) Power Consumption: Let E_S and E_L denote the consumed energies per unit time in the sleep interval and the listening interval, respectively, then the power consumption in the talk-spurt period is obtained by

$$
\begin{aligned}
PW_T &= \frac{T_{Smax,T} E_S + T_L E_L}{T_{Smax,T} + T_L} \\
&= \frac{(k_{max} \cdot T_P - T_L) E_S + T_L E_L}{k_{max} \cdot T_P}. \quad (16)
\end{aligned}
$$

IV. Power Saving Mode for Mutual Silence Period

We first investigate a typical two-way conversational model called the *Brady* model and then analyze the PSM in the mutual silence period.

A. Two-way Conversational Model

Brady [13] proposed a general six-state model that provides a good statistical analysis of two-way conversation. Fig. 5 shows the Brady model and the values of the state transition parameters. This figure is divided into quadrants, each of which represents a different state for parties A and B engaged in a conversation. Note that the state transitions for party A have the same characteristics as those for party B.

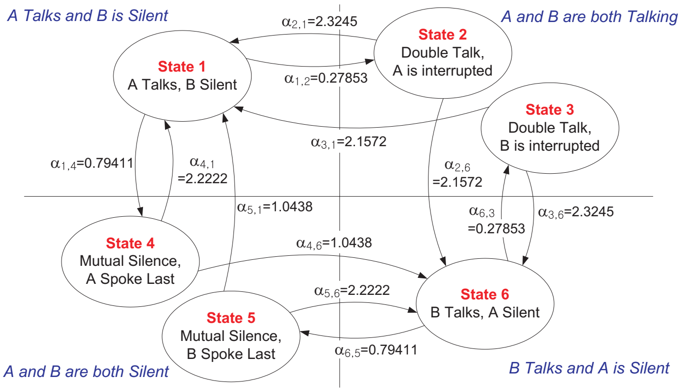

Fig. 5. Brady model for two-way conversation.

Let π_i be the steady state probability that the Markov state stays in state i in the long run, and then the probability vector $\Pi = [\pi_1 \ \pi_2 \ \pi_3 \ \pi_4 \ \pi_5 \ \pi_6]$. The transition rate matrix P from the Brady model is described as

$$P = \begin{bmatrix} \alpha_{1,1} & \alpha_{1,2} & 0 & \alpha_{1,4} & 0 & 0 \\ \alpha_{2,1} & \alpha_{2,2} & 0 & 0 & 0 & \alpha_{2,6} \\ \alpha_{3,1} & 0 & \alpha_{3,3} & 0 & 0 & \alpha_{3,6} \\ \alpha_{4,1} & 0 & 0 & \alpha_{4,4} & 0 & \alpha_{4,6} \\ \alpha_{5,1} & 0 & 0 & 0 & \alpha_{5,5} & \alpha_{5,6} \\ 0 & 0 & \alpha_{6,3} & 0 & \alpha_{6,5} & \alpha_{6,6} \end{bmatrix} \quad (17)$$

where $\alpha_{i,i} = -\sum_{j=1, j\neq i}^{6} \alpha_{i,j}$. Therefore, π_i can be obtained by the balance equation $\Pi = \Pi \cdot P$ and the normalized condition $\sum_{i=1}^{6} \pi_i = 1$. The computation results show that the average percentage of the mutual silence state $(\pi_4 + \pi_5)$ is about 20% and the average duration of mutual silence $(\frac{1}{\alpha_{4,1}+\alpha_{4,6}})$ is about 306 ms.

B. Analysis in Mutual Silence Period

To analyze the performance of the PSM in the mutual silence period, we construct a simplified analytic model as shown in Fig. 6. The PSM for the mutual silence period finishes due to the arrival of voice packets by the restart of talk-spurt period. Let Y be the distribution of lengths of the mutual silence periods. From the Brady model, the pdf of Y is derived as

$$f_Y(t) = \lambda e^{-\lambda t} \quad (18)$$

where arrival rate $\lambda = \alpha_{4,1} + \alpha_{4,6}$. Namely, the mutual silence period follows the exponential distribution with rate λ. Since the exponential distribution has a memoryless property, each sleep cycle in the mutual silence period has the same property, so we may consider only one sleep cycle among the total sleep cycles in the mutual silence period, as shown in Fig. 6.

1) Buffering Delay: Delay margin is the time difference between the total propagation delay and the maximum tolerable end-to-end delay. Let Δ be the delay margin of voice packets arriving in the BS. Using (2), Δ is given by

$$\Delta = D_{max} - D_{dec} - X. \quad (19)$$

As shown in Fig. 6, we let W denote the length of one sleep cycle in the mutual silence period and define the duration $U = W - \Delta$. Note that if all packets arrive during U, they are all dropped. Given that the first packet arrives during U, the actual packet drop duration, V, becomes

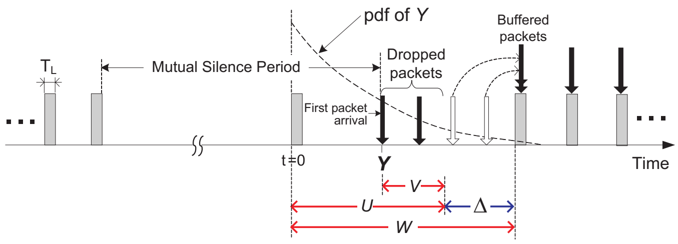

Fig. 6. Analytic model for PSM in mutual silence period.

$$V = U - Y \qquad \text{if } Y \in U. \quad (20)$$

Let p be the probability that the first packet arrives in U, then p is given by

$$p = P\{Y \in U | Y \in W\} = \frac{1 - e^{-\lambda U}}{1 - e^{-\lambda(U+\Delta)}}. \quad (21)$$

Assuming that there is no state transition during V (i.e., a silent period does not occur immediately after a talk-spurt occurs), we can calculate the average packet drop duration, $E[V]$, as follows.

$$\begin{aligned} E[V] &= E[V \mid Y \in U] \cdot p \\ &= E[U - Y \mid Y \in U] \cdot p \\ &= \{U - E[Y \mid Y \in U]\} \cdot p \end{aligned} \quad (22)$$

where

$$\begin{aligned} E[Y \mid Y \in U] &= \int_0^U t \cdot f_{Y|U}(t) dt \\ &= \int_0^U t \cdot \frac{f_Y(t)}{P\{Y \in U\}} dt \\ &= \int_0^U \frac{t \cdot \lambda e^{-\lambda t}}{1 - e^{-\lambda U}} dt \\ &= \frac{1}{\lambda} - \frac{U e^{-\lambda U}}{1 - e^{-\lambda U}}. \end{aligned} \quad (23)$$

From (22) and (23), the average packet drop duration is obtained by

$$\begin{aligned} E[V] &= \left(U - \frac{1}{\lambda} + \frac{U e^{-\lambda U}}{1 - e^{-\lambda U}} \right) \cdot p \\ &= \left(U - \frac{1 - e^{-\lambda U}}{\lambda} \right) \cdot \frac{1}{1 - e^{-\lambda(U+\Delta)}}. \end{aligned} \quad (24)$$

In the mutual silence period, the buffering delay becomes the time duration during which the BS buffers voice packets until it transmits them in the nearest listening interval. Therefore, the average buffering delay in the mutual silence period is expressed as

$$\begin{aligned} D_{buf,S} &= E[V] + E[\Delta] \\ &= E[V] + D_{max} - D_{dec} - E[X] \end{aligned} \quad (25)$$

where $E[X] = \sum_{r=1}^{R} \frac{1}{\lambda_r} + D_{fix}$.

2) Drop Probability: The packet drop probability in the mutual silence period, $P_{drop,S}$, can be defined as the ratio of the average packet drop duration to the duration of the sleep cycle, as follows.

$$\begin{aligned} P_{drop,S} &= \frac{E[V]}{W} \\ &= \frac{E[V]}{E[Y] + D_{buf,S}} \\ &= \frac{E[V]}{1/\lambda + E[V] + E[\Delta]}. \end{aligned} \quad (26)$$

which is an equation of unknown U. From (24) to (26), we can obtain the maximum U value, U_{max}, as solving the inequality $P_{drop,S} \leq \delta$. Let $T_{Smax,S}$ denote the maximum sleep interval for the mutual silence period. As shown in Fig. 6, $T_{Smax,S}$ is decided by

$$T_{Smax,S} = U_{max} + E[\Delta] - T_L. \quad (27)$$

3) Power Consumption: Finally, the power consumption in the mutual silence period is given by

$$PW_S = \frac{T_{Smax,S}E_S + T_L E_L}{T_{Smax,S} + T_L}. \quad (28)$$

V. Implementation Issues

In company with the theoretical approach, we need to discuss how to obtain the maximum sleep intervals and apply them in practical network environments. To calculate the maximum sleep interval for the talk-spurt period, it is important to accurately estimate the delay distribution of arriving VoIP packets. There are a few methods to estimate the one-way network propagation delay. One of the easiest is to use the real-time transport protocol (RTP). Since the VoIP packet is transmitted by the RTP protocol, its delay can be measured by using the time stamp field in the RTP header, or by the round-trip time computation method using the real-time transport control protocol (RTCP) [23]. Another likely way to estimate the delay is to use a network management tool. As a VoIP assessor, *Chariot* [24] provides various measurement results such as the number of routers, the service time in each router, jitter, QoS information, as well as the one-way delay of the VoIP packet.

On the other hand, to calculate the maximum sleep interval for the mutual silence period, it is essential to estimate the distribution of the mutual silence periods. We can estimate it by measuring the length of mutual silence period throughout the VoIP call. The mutual silence period can be recognized by the arrival of a *Silence Insertion Descriptor* (SID) frame that is sent by the sender's VoIP codec at the beginning of the silent period [25]. This SID frame is smaller in size than a VoIP data packet and includes only information about background noise that is used to generate artificial noise at the receiving side's decoder during the silent period. Therefore, both the MS and BS can identify a mutual silence period by simply comparing the sizes of the received packets.

By the way, to make the distribution of mutual silence period accurate, enough measurement data about the length of mutual silence is required and it takes a time to get enough samples of mutual silence in practical VoIP communications. Because of this problem, we can consider a heuristic method that applies a truncated *binary exponential backoff* (BEB) algorithm to determine the sleep intervals in the mutual silence period [26]. Namely, the length of the i-th sleep interval during the mutual silence period, T_i, increases exponentially by $T_i = \min(2^{i-1}T_{min},\ T_{max})$. This BEB algorithm is a general method that may be suitably applied to the unknown situation when we cannot know when the traffic arrives, so it is appropriate for the mutual silence period.

TABLE I
PARAMETER SETUP

Parameter	Value	Description
T_P	20 ms	packet generation interval of voice codec
T_L	5 ms	listening interval
E_S	0.045 W	power consumption in sleep interval
E_L	1.5 W	power consumption in listening interval
R	2∼10	number of routers
$1/\lambda_r$	2∼10 ms	average service time in the routers
D_{enc}	3 ms	voice-encoding delay
D_{pack}	20 ms	packetization delay
D_{mnet}	10 ms	minimal network delay
D_{dec}	3 ms	voice-decoding delay
D_{max}	270 ms	maximum tolerable end-to-end delay
δ	0.03	maximum tolerable packet drop rate

VI. Results and Discussions

For performance evaluation, we used the parameters shown in Table I. For the parameters related to the voice codec, we considered the ITU-T G.729B codec with a silence suppression scheme, which is mostly used for VoIP applications [11]. We set the listening interval to 5 ms, which corresponds to one frame length in the 802.16e specification [6]. Since the size of a VoIP packet is very small compared to the capacity of 802.16e (maximum 30 Mbps), a T_L of 5 ms is sufficient for the delivery of buffered VoIP packets. We considered that MS has two power strengths that are used in the listening and sleep states [27]. We assume that the number of routers varies from 2 to 10 and the service time of each router follows a Gaussian distribution with average $1/\lambda_r$. The fixed delay terms are based on [22]. For VoIP QoS, we set the maximum end-to-end delay to 270 ms and the maximum packet drop rate to 0.03, which correspond to the *user-satisfaction* level of VoIP quality [28].

A. Results of PSM for Talk-spurt Period

Fig. 7 shows the normalized power consumption and average buffering delay in the PSM for talk-spurt period (PSM-T) versus the length of the sleep cycle (kT_P). The normalized power consumption is a relative power consumption normalized by the power consumption when the sleep cycle is equal to the generation interval of the VoIP packets, T_P, (i.e., $k = 1$). We have validated the analytical model by a simulation program using Network Simulator version 2 (NS-2). As the length of the sleep cycle increases, the power consumption significantly decreases, but the buffering delay increases. Obviously, there is a trade-off relationship between power consumption and buffering delay.

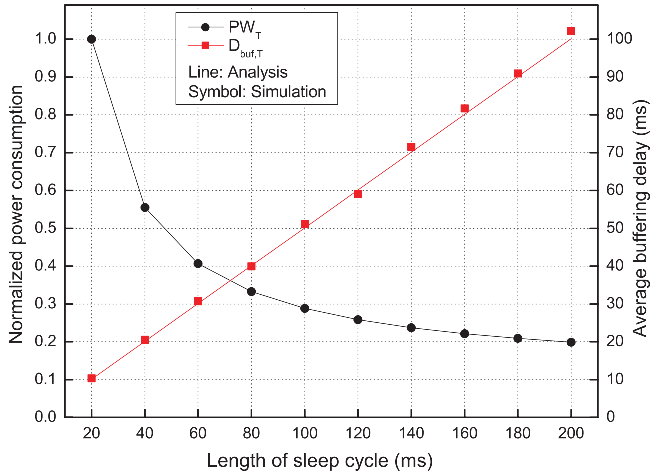

Fig. 7. Normalized power consumption and average buffering delay in PSM-T vs. length of sleep cycle.

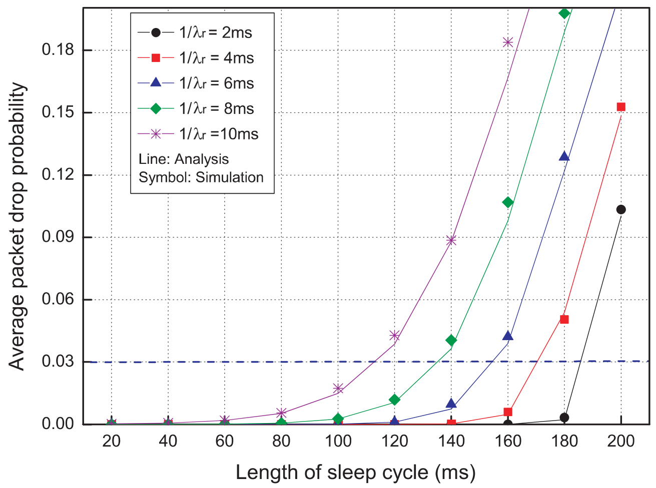

Fig. 8. Average packet drop probability in PSM-T vs. length of sleep cycle when $R = 6$.

Fig. 8 shows the average packet drop probability versus the length of the sleep cycle according to the average service time ($1/\lambda_r$) in each router when there are six routers in the routing path. As the length of the sleep cycle increases, the average drop probability increases exponentially because the longer the sleep cycle, the larger the buffering delay. Furthermore, as the service time is increased, the average drop probability becomes greater because the network delay of arriving voice packets is increased. From this result, we can know the maximum length of the sleep cycle that satisfies the requirement of the packet drop probability being less than δ. That is, in each line, the length of the sleep cycle whose drop probability is 0.03 becomes the maximum sleep cycle.

Fig. 9 shows the maximum sleep interval and the normalized power consumption according to various VoIP communication environments (i.e., variations in the number of routers and the average service time). The maximum sleep interval is a result of theoretical analysis and the normalized power consumption indicates a minimum power consumption when the maximum sleep interval is applied. Note that this result of maximum

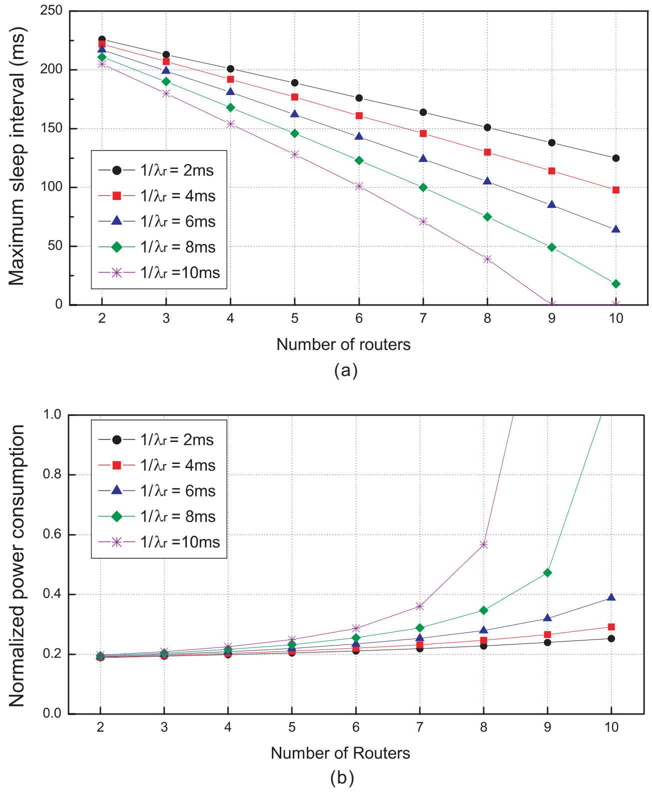

Fig. 9. (a) maximum sleep interval and (b) normalized power consumption in PSM-T vs. number of routers.

sleep interval agrees with the results of Fig. 8. As the number of routers and the average service time are increased, the maximum sleep interval is decreased and eventually the power consumption is increased. This is because the network delay of voice packets is increased and there is not a sufficient delay margin to permit sleeping as the number of routers and the service time are increased. When the maximum length of the sleep cycle is zero, it means that there is no available sleep interval, so the MS cannot use a PSM and should always be awake.

B. Results of PSM for Mutual Silence Period

Fig. 10 shows the normalized power consumption and average packet drop probability in the PSM for mutual silence period (PSM-S) versus the length of the sleep cycle. According to the tradeoff relationship, the power consumption falls and the average packet drop probability increases as the length of the sleep cycle increases. When the length of the sleep cycle is small, there is no packet drop because the delay margin during the mutual silence period is larger than the length of the sleep cycle. However, as the length of the sleep cycle increases, the buffering delay during the PSM-S operation is increased, so the packet drop rate is increased.

Fig. 11 shows the maximum sleep interval and the normalized power consumption according to the variations in the number of routers and the average service time. As with PSM-T, the maximum sleep interval is decreased and the power consumption is increased as the number of routers and the average service time are increased. As expected, PSM-S has

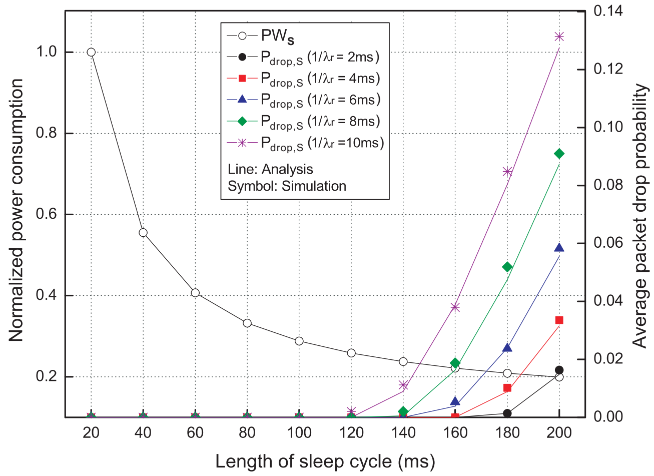

Fig. 10. Normalized power consumption and average packet drop probability in PSM-S vs. length of sleep cycle when $R = 6$.

a longer maximum sleep interval and less power consumption than PSM-T under the same network environments because there is no packet transmission in the mutual silence period. This result shows that the use of the maximum sleep interval in the mutual silence period saves much more energy when PSM-S is used than when it is not used.

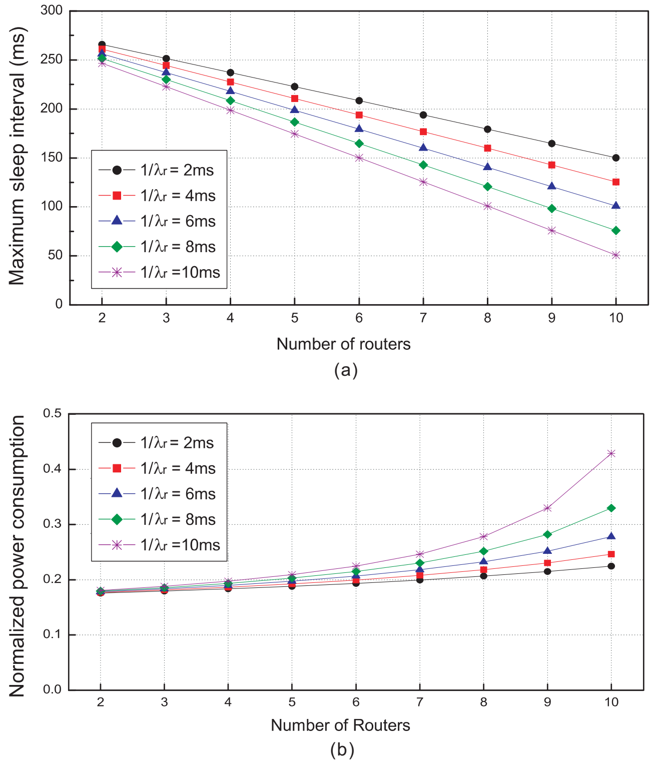

Fig. 11. (a) maximum sleep interval and (b) normalized power consumption in PSM-S vs. number of routers.

VII. Conclusions

In this paper, we analyzed the VoIP PSM and derived the efficient sleep interval placements applicable during the talk-spurt periods and the mutual silence periods, respectively, while satisfying the given QoS constraint of a VoIP connection. The analysis and simulation results showed that the maximum sleep interval is strongly affected by the VoIP network environment and the QoS constraint of VoIP, and the VoIP PSM using the maximum sleep interval could minimize the power consumption of the VoIP device, saving up to 80% power, while satisfying the constraints of the VoIP QoS in various network environments. These results indicate that it is feasible for the VoIP PSM to lengthen the sleep interval within the maximum bound of sleep interval because it gives a significant power-saving effect while satisfying the given VoIP QoS. In addition, this study is expected to be applied to other applications with different traffic attributes for the purpose of power conservation of mobile devices.

References

[1] Suckchel Yang, Myungsik Yoo, Yoan Shin, "Adaptive discontinuous reception mechanism for power saving in UMTS," *IEEE Communications Letters*, vol. 11, issue 1, pp. 40-42, Jan. 2007.

[2] Tuan-Che Chen, Jyh-Cheng Chen, and Ying-Yu Chen, "Maximizing Unavailability Interval for Energy Saving in IEEE 802.16e Wireless MANs," *IEEE Trans. on Mobile Computing*, vol. 8, no. 4, pp. 475-487, Apr. 2009.

[3] T. R. Park and M. J. Lee, "Power saving algorithms for wireless sensor networks on IEEE 802.15.4," *IEEE Commun. Mag.*, vol. 46, no. 6, pp. 148-155, June 2008.

[4] 3rd Generation Partnership Project 2 (3GPP2), "cdma2000 high rate packet data air interface specification," Tech. Rep. C.S20024-A v1.0, Apr. 2004.

[5] *IEEE Std 802.11, 1999 Edition*, "Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specifications," Aug. 1999.

[6] *IEEE Std 802.16e-2005*, "Part 16: Air Interface for Fixed and Mobile Broadband Wireless Access Systems," Feb. 2006.

[7] *IEEE Std 802.11e*, "Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) specifications, Amendment 8: Medium Access Control (MAC) Quality of Service Enhancements," Nov. 2005.

[8] Dinh Thi Thuy Nga, M.-G. Kim, and M. Kang, "Delay-guaranteed Energy Saving Algorithm for the Delay-sensitive Applications in IEEE 802.16e Systems," *IEEE Trans. on Consumer Electronics*, vol. 53, issue 4, pp. 1339-1347, Nov. 2007.

[9] R. V. Cox and P. Kroon, "Low Bit-Rate Speech Coders for Multimedia Communication," *IEEE Commun. Mag.*, vol. 34, no. 12, pp. 34-41, Dec. 1996.

[10] Recommendation G.723.1 Annex A, "Dual rate speech coder for multimedia communications transmitting at 5.3 and 6.3 kbit/s, ANNEX A: Silence Compression Scheme," Approved in Nov. 1996.

[11] Recommendation G.729 Annex B, "A silence compression scheme for G.729 optimized for terminals conforming to Recommendation V.70," Approved in Oct. 1996.

[12] V. Abreu-Sernandez and C. Garcia-Mateo, "Adaptive Multi-Rate Speech Coder for VoIP Transmission," *IEE Electronics Letters*, vol. 36, issue 23, pp. 1978-1980, Nov. 2000.

[13] P. T. Brady, "A model for generating on-off speech patterns in two-way conversation," Bell Syst. Tech. J., pp. 2445-2472, Sept. 1969.

[14] Y. Chen, N. Smavatkul, and S. Emeott, "Power management for VoIP over IEEE 802.11 WLAN," *IEEE Wireless Communications and Networking Conference*, vol. 3, pp. 1648-1653, Mar. 2004.

[15] X. Pérez-Costa and D. Camps-Mur, "APSM: bounding the downlink delay for 802.11 power save mode," *IEEE ICC 2005*, vol. 5, pp. 3616-3622, May 2005.

[16] Chih-Yung Shih, Yih-Shen Chen, Ray-Guang Cheng, and Chung-Ju Chang, "A power-efficient MAC protocol for integrated voice and data services over 802.11e HCF," *IEEE Vehicular Technology Conference*, vol. 4, pp. 2032-2036, May 2004.

[17] Shiao-Li Tsao, Chung-Huei Huang, and Tzu-Ming Lin, "Energy conserving packet transmission schemes for video and voice over WLAN," *IEEE Consumer Communications and Networking Conference*, vol. 2, pp. 758-762, Jan. 2006.

[18] J. Janssen, D. De Vleeschauwer, M. Buchli, G.H. Petit, "Assessing voice quality in packet-based telephony," *IEEE Internet Computing*, vol. 6, issue 3, pp. 48-56, May-June 2002.

[19] T. Yensen, J. P. Lariviere, I. Lambadaris, and R. A. Goubran, "HMM delay prediction technique for VoIP," *IEEE Trans. on Multimedia*, vol. 5, issue 3, pp. 444-457, Sep. 2003.

[20] B. Kao, H. Garcia-Molina, and D. Barbara, "Aggressive transmissions of short messages over redundant paths," *IEEE Trans. on Parallel and Distributed Systems*, vol. 5, pp. 102-109, Jan. 1994.

[21] A. Fasbender, D. Kesdogan, and O. Kubitz, "Analysis of Security and Privacy in Mobile IP," 4th International Conference on Telecommunication Systems, Modelling and Analysis, Nashville, TN, USA, Mar. 1996.

[22] M. Yavuz, et al., "VoIP over cdma2000 1xEV-DO Revision A," *IEEE Commun. Mag.*, vol. 44, pp. 88-95, Feb. 2006.

[23] RFC 3550, "RTP: A Transport Protocol for Real-Time Applications," *IETF*, July, 2003.

[24] http://www.netiq.com, "Chariot: Evaluating Data Networks for VoIP"

[25] A. Estepa, et al., "A New Approach for VoIP Traffic Characterization," *IEEE Commun. Lett.*, vol. 8, no. 10, pp. 644-646, Oct. 2004.

[26] Byung-Jae Kwak, Nah-Oak Song, L.E. Miller, "Performance analysis of exponential backoff," *IEEE/ACM Trans. on Networking*, vol. 13, issue 2, pp. 343-355, Apr. 2005.

[27] E. S. Jung, and N. H. Vaidya, "An energy efficient MAC protocol for wireless LANs." *IEEE INFOCOM 2002*, vol. 3, pp. 1756-1764, Jun. 2002.

[28] ITU-T Recommendation G.114, "One-way transmission time," May, 2003.