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Power Control for Packet Voice Service with Application to EDGE Wireless System

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Using the enhanced data rates for GSM evolution (EDGE) system with cyclic frequency hopping as an example, we apply a Kalman-filter power control method based on interference tracking to packet voice service in wireless networks. Our results show that the power-control method significantly improves the spectral efficiency by enabling the 1/3 frequency reuse while maintaining a stringent requirement of 2% packet loss probability for voice service. Specifically, for allocated spectrum of 1.8, 3.6 and 5.4 MHz, the 1/3 reuse with the Kalman power control can yield 102.5%, 49.5% and 32.5% improvement in spectral efficiency, respectively, over the 3/9 reuse (regardless of whether or not power control is used). We also compare the performance of the Kalman method with a traditional signal-to-interference-ratio method and a control method that is based on the last interference measurement. We find that appropriate selection of power for the first packet of each talk spurt and the filtering function for noisy measurements are crucial in providing high system capacity for packet voice service. For the EDGE system, we also identify a need for shortening the power update period, which is 480 ms in the specifications.

KEY WORDS: EDGE system; Kalman filtering; packet error performance; packet voice service; power control; spectral efficiency; wireless network.

1. INTRODUCTION

The European Telecommunication Standard Institute (ETSI) has established the protocol standards for the enhanced data rates for GSM evolution (EDGE) system [1] as one of our future time-division-multiple-access (TDMA) wireless networks. Using packet-switching technology and the existing 200 kHz GSM channels, the EDGE system employs a link-adaptation technique to support the highest data rate approaching 480 kbits/s. The EDGE system is expected to serve as a platform for integrated services including packet voice and data.

Transmission power control has been widely practiced to manage interference in cellular networks [2,3]. Existing algorithms can be classified as *signal-based* [4] and *signal-to-interference-ratio (SIR) based* [3,5] power control. SIR-based power control iteratively adjusts power according to previous SIR measurements, and typically performs well for calls with relatively long holding time. Recently, for wireless packet networks with bursty transmission, Leung [6] proposes a power-control method that is based on prediction of interference power by use of a Kalman filter. The purposes of this work are twofold. First, we explore the applicability and quantify the performance gain of the Kalman power control for packet voice service in the EDGE system. Second, we examine exchange of control information needed for the power-control method. Although we use the EDGE system to make our discussion concrete, ideas in this paper are generally applicable to other wireless packet networks.

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In the following, section 2 outlines the Kalman power control and its pertinent operations of the EDGE system. In section 3, we use simulation to quantify the performance gain of the power control for packet voice service in terms of spectral efficiency and coverage. We also compare performance between the Kalman and SIR-based methods. Finally, section 4 presents our conclusion.

2. EDGE SYSTEM AND POWER CONTROL

The EDGE system uses existing 200 kHz channels (carriers) in the GSM. Each carrier is divided into time slots and eight adjacent slots form a TDMA frame [7]. In one of the proposals, the EDGE system has nine modulation and coding schemes (MCSs) [8] and uses a link-adaptation technique to adapt transmission to one of the schemes according to the link quality. Since we focus on packet voice here, it is assumed that packets of all calls are transmitted at MCS-2 (using GMSK modulation) to achieve robustness and a data rate of 11.2 kbits/s per time slot, adequate for voice applications. Furthermore, this paper considers only voice service and the use of power control for integrated voice and data services is an area of our future work.

Typically, a vocoder generates one voice packet (also known as voice frame) per 20 ms. Each voice packet is treated as a radio-link-control (RLC) block. In turn, each RLC block is divided into four bursts, which are transmitted in a designated time slot over four successive TDMA frames, one burst per frame. For simplicity, we treat these four frames carrying the four bursts as one single frame (for one voice packet) in this paper.

For packet voice service, each call alternates between talking mode, at which packets are generated periodically by the vocoder, and silence period. To enable channel assignment and relinquishment on a per talk-spurt basis, fast signaling mechanisms between mobile stations (MSs) and base stations (BSs) are needed. For this purpose, Qiu et al. [9] has proposed a set of new signaling protocols for maintaining satisfactory voice clipping (i.e. loss of first few packets of a talk spurt). Since we focus on power control, such new protocols are assumed to be available so that voice clipping is not an issue and our primary performance concern is the packet loss probability for each call.

The Kalman-filter method [6] is used to control transmission power. Each MS continuously measures the interference-plus-noise power (referred to as *interference power* for brevity) for a small set of downlink channels, which is being used or may be used to carry voice traffic from the BS to the MS. These measurements are fed into a Kalman filter to predict future interference power on these channels. Similar processes are also performed at each BS to track interference power for uplink transmission.

Since voice packets of a talk spurt are transmitted in the same time slot over successive TDMA frames (i.e. a voice channel) in the EDGE system, let us focus only on *that time slot* and index the frames by n. For a transmitter, either a MS or BS, its transmission power in the time slot of frame n is set to be

$$p(n) = \min\{\gamma^* \tilde{i}(n)/g(n), p_{\max}\} \quad (1)$$

where γ^* is the SINR target for the voice service, $g(n)$ is the path gain between the transmitter and the intended receiver in frame n (which can be estimated by use of control channel for handoff purposes), $p_{\max}$ is the maximum power level and $\tilde{i}(n)$ is the interference power (in mW) for the slot in frame n predicted by the Kalman filter. The update equations for the Kalman filter are:

$$\tilde{I}_{n+1} = \hat{I}_n \quad (2)$$

$$\tilde{P}_{n+1} = \hat{P}_n + Q_n \quad (3)$$

$$K_n = \tilde{P}_n(\tilde{P}_n + R_n)^{-1} \quad (4)$$

$$\hat{I}_n = \tilde{I}_n K_n (Z_n - \tilde{I}_n) \quad (5)$$

$$\hat{P}_n = (1 - K_n)\tilde{P}_n \quad (6)$$

where $\tilde{I}_n$and $\hat{I}_n$ are the a priori and a posteriori estimate of interference power (in dBm) for frame n, $\tilde{P}_n$ and $\hat{P}_n$ are the a priori and a posteriori estimate of error variance, K_n is the Kalman gain, and Q_n and R_n are the variance for the process and measurement noise, respectively. Note that $\tilde{i}(n)$ is the linear-scale equivalent in mW of $\tilde{I}_n$ in dBm. Let Z_n and $\bar{Z}_n$ be the measured interference power for frame n and the average interference over the last W frames prior to frame n, respectively. Hence, we have

$\bar{Z}_n = 1/W \sum_{i=n-W+1}^{n} Z_i$ and the variance of interference measurements is

$$V_n = \frac{1}{W-1} \sum_{i=n-W+1}^{n} (Z_i - \bar{Z}_n)^2 \quad (7)$$

To illustrate our ideas here, Q_n and R_n are estimated by

$$Q_n = \xi V_n \quad (8)$$

and

$$R_n = \zeta V_n \quad (9)$$

respectively, where ξ and ζ are constant between 0 and 1. Note that Q_n and R_n can be obtained by other ways to optimize the filter performance and the issue is addressed in our subsequent work.

The Kalman method requires exchange of control information between transmitter and receiver outlined in Figs. 1 and 2. As shown in Fig. 1, when a call is in a silent period for downlink transmission, the associated MS continuously measures and predicts by use of the Kalman filter the interference power on several channels. When the next talk spurt starts, the BS sends a paging message to the MS over a control channel. In turn, the MS includes the predicted interference power for a few voice channels in the paging response message. The BS selects (possibly making use of the interference predictions) and informs the MS of the chosen channel in the resource-assignment message. Then, the BS can start transmitting voice packets. While receiving packets, the MS continues to measure and predict interference power for the given set of channels, including the channel where packets are received. Periodically, it sends the interference prediction for the receiving channel back to the BS via a control channel. With the new prediction $\tilde{i}(n)$, the BS adjusts its power according to (1). Similar operations apply to the uplink transmission in Fig. 2.

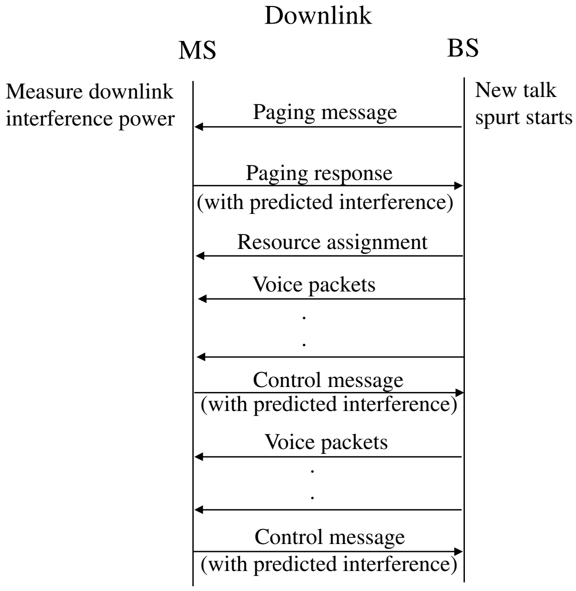

Fig. 1. Control messages for downlink transmission.

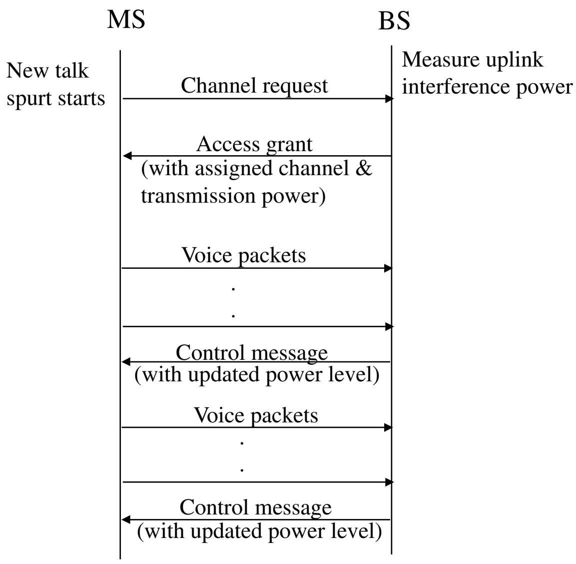

Fig. 2. Control messages for uplink transmission.

3. PERFORMANCE STUDY

3.1. Simulation Model

A total of 37 cells in a hexagonal layout is simulated. Each cell has three sectors, each of which is served by a BS antenna at the center of the cell. The 3-dB beamwidth of each BS antenna is 60°, while MSs have omni-directional antennas. The BS antenna has a front-to-back gain ratio of 25 dB. Frequency reuse factor of 1/3 and 3/9 are considered. Each radio link is characterized by a path-loss model with an exponent of 3.5 and lognormal shadow fading with a dB standard deviation of 6. Cell radius is assumed to be 1 km and the path loss at

100 m from the cell center is −73 dB. Unless stated otherwise, thermal noise at each receiver is fixed and equal to −116 dBm (for the 200 kHz GSM channel with 5 dB noise factor). Transmission power is limited between 1 and 30 dBm. Each sector is populated with 100 MSs randomly and each of them selects the BS that provides the strongest signal power. All MSs remain at the fixed locations. Unless stated otherwise, ξ and ζ in (8) and (9) are 1 and 0.5, respectively.

For each packet transmission, the SINR is measured at the receiver and rounded to its closest integer value in dB. The packet error is determined based on the SINR value and the corresponding error probability (which is averaged over Rayleigh fading with cyclic-frequency hopping) in Table I. Packet error probability is zero for SINR exceeding 23 dB. With these error probabilities, γ^* in (1) is selected to minimize the overall packet error rate (i.e., averaged over all MSs) and we found that $\gamma^* = 15$ dB provides the best results.

The durations of a talk spurt and a silent period are exponentially distributed with an average of 1 and 1.35 s, respectively. As a packet is generated every 20 ms, the number of packets in a talk spurt is geometrically distributed with an average of 50. Mechanisms such as those in Ref. [9] for fast channel assignment on a per talk-spurt basis are used to avoid voice clipping. When a talk spurt starts, the BS randomly assigns one available channel to carry the talk spurt. If no channel is available, the entire talk spurt is lost (or blocked).

Table I. Packet error probability vs. SINR

Measured SINR (dB)	Packet error probability
0	0.9001
1	0.8364
2	0.7547
3	0.6610
4	0.5649
5	0.4638
6	0.3645
7	0.2748
8	0.2024
9	0.1410
10	0.0949
11	0.0659
12	0.0452
13	0.0274
14	0.0171
15	0.0103
16	0.0070
17	0.0045
18	0.0026
19	0.0016
20	0.0010
21	0.0006
22	0.0004
23	0.0002
24	0.0000

3.2. Performance Results

We define that the quality of packet voice service is satisfactory if (a) the blocking probability of both new call and talk spurts due to channel unavailability is less than 2%, and (b) packet error rate does not exceed 2% [10] for calls associated with at least 90% of MSs in each sector (i.e., a 90% coverage requirement). By assuming that talk spurts arrive according to a Poisson process, the voice capacity is the maximum traffic load in Erlangs while maintaining the satisfactory service quality.

For comparison, we also studied the performance of a traditional SIR power control [5]. Specifically, the power of the first packet of a talk spurt is chosen to fully compensate its path loss and shadow fading, subject to the maximum power $p_{\max}$. Power for subsequent packets (indexed by n) are adjusted according to

$$p(n) = \min\{p(n-1)\gamma^*/\beta(n-1), p_{\max}\} \qquad (10)$$

where $\beta(n-1)$ is the measured SINR for packet $n{-}1$. Let us refer to this power control as the G-SINR method because the power is selected based on signal path gain (G) and SINR measurements. We note that an exponential smoothing can be applied to the SINR measurements in the method proposed in Ref. [5]. As measurements are assumed to be error free at this point, the smoothing is not included in (10) to improve convergence. We shall examine another method that selects power for the first packet differently, and how its performance is affected by measurement errors in section 3.4.

For 1.8, 3.6 and 5.4 MHz spectrum allocated for the 1/3 reuse, each sector of a cell is assigned with 24 (i.e., 3 200 kHz carriers times 8 slots), 48 and 72 voice channels, respectively. Similarly, for the 3/9 reuse, each sector has one third of these many channels. Figure 3 shows the downlink spectral efficiency in terms of Erlang/cell/MHz. This figure assumes that the SINR and interference power are measured accurately, and that the measurements

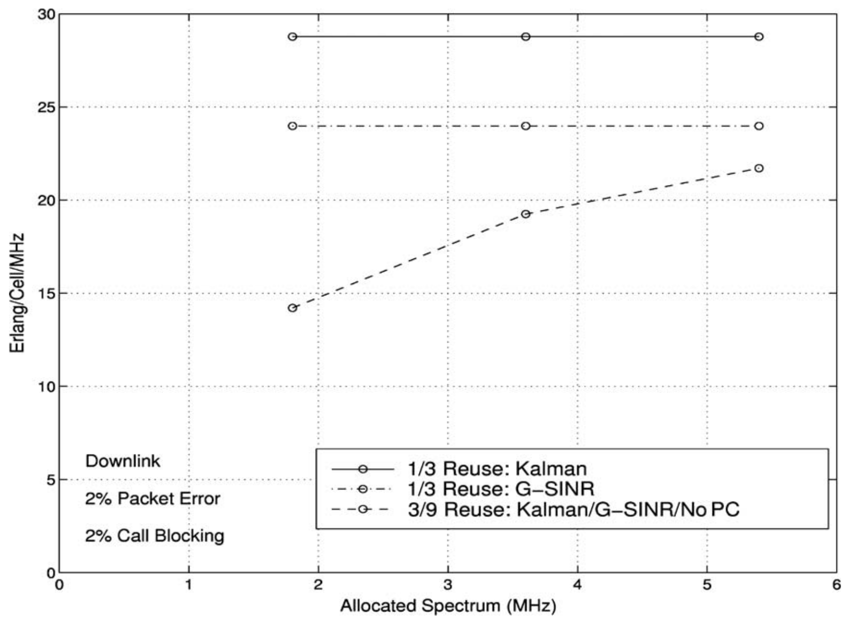

Fig. 3. Spectral efficiency for the 1/3 and 3/9 reuse.

for one voice packet can be used to control power for the next packet.

We make several observations. First, our results for the 3/9 reuse show that the Kalman and G-SINR power control enable each voice channel to carry 100% of traffic load, and no power control can support 70% traffic while providing satisfactory service quality. However, the limiting factor for the spectral efficiency of the 3/9 reuse is the blocking probability for talk spurts. Thus, the Kalman method, G-SINR method and no power control yield the same spectral efficiency in Fig. 3. As the allocated spectrum increases, the trunking efficiency and thus the spectral efficiency are improved. On the other hand, since the 1/3 reuse is the lowest reuse factor, each sector is allocated with the maximum possible number of channels for a given spectrum, thus avoiding the trunking inefficiency. The limiting factor for the 1/3 reuse is the packet loss probability, which is mainly determined by the carried traffic load of each channel and thus the interference. In this case, since the voice capacity is almost directly proportional to the maximum feasible load on each channel (as traffic load is balanced among all channels by the random channel assignment), the spectral efficiency become independent of the actual spectrum allocation in Fig. 3.

Second, we found that the Kalman and G-SINR power control support each channel to carry a maximum of 30% and 25% of traffic load, respectively, to maintain the stringent 2% loss probability with 90% coverage. For the 1/3 reuse, the spectral efficiency for the power-control methods is 28.78 and 23.98 Erlangs/cell/MHz, respectively. That is, the Kalman power control yields about 20% improvement over the G-SINR control, as shown in Fig. 3. Furthermore, for the 1.8, 3.6 and 5.4 MHz spectrum allocation, the 1/3 reuse with the Kalman power control provides 102.5%, 49.5% and 32.5% improvement in spectral efficiency, respectively, over the 3/9 reuse with the Kalman, G-SINR or no power control. We choose to compare with the 3/9 reuse here because it gives a capacity higher than the 1/3 reuse with no power control.

Besides the 20% improvement of the Kalman power control over the G-SINR control, our results also reveal that the former method is more robust than the latter in terms of coverage. More specifically, Figure 4 shows the impact of coverage due to power update period. To obtain these results, MSs continue to measure and track interference power for each packet transmission, but the transmission power is updated (changed) periodically according to the given update period. As mentioned earlier, the 90% coverage requirement is met by the methods when the system runs at their respective capacity of 30% and 25% traffic load. However, at their capacity load, if transmission power is updated every two voice packets, the coverage for the Kalman and G-SINR method reduces to 89.3% and

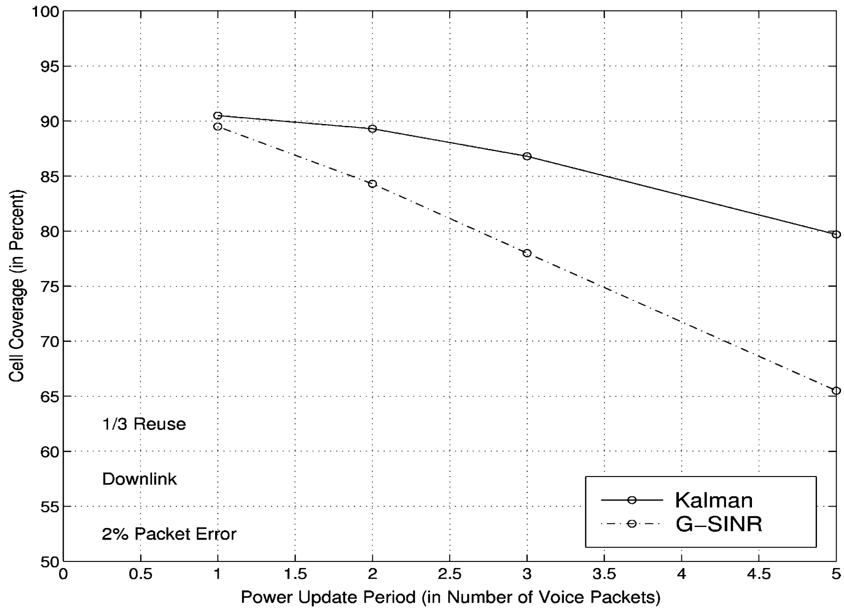

Fig. 4. Impact of power update period.

84.3%, respectively. As shown in Fig. 4, additional increase in the update period further degrades the coverage performance. Nevertheless, these results show that the Kalman power control is more robust than the G-SINR method.

The coverage degradation in Fig. 4 implies reduction of voice capacity if the 90% coverage requirement has to be maintained. Figure 5 presents how the voice capacity decreases as a function of update period. Note that the update period in terms of number of voice packets can be determined from the fact that one packet is generated per 20 ms in the EDGE system. Similar to the coverage performance in Fig. 4, Figure 5 reveals that the capacity gain by power control for cyclic frequency hopping strongly depends on the update period (and the current EDGE standards specifies a period of 480 ms). This is so because the talk spurt lasts for only one second on average. Thus, interference power fluctuates far more frequently in the packet environment than in the circuit-switching counterpart, especially for the 1/3 reuse with partial traffic loading. Thus, frequent power update is needed to keep up with the fluctuation and to realize the potential gain of power control. In particular, when the update period is reduced to 100 ms, the Kalman power control for the 1/3 reuse yields about 49% increase in voice capacity when compared to that for the 3/9 reuse without power control. The improvement for the Kalman power control is reduced when the update period decreases from 40 to 20 ms because of additional control overhead, which is assumed to be one burst (i.e., ¼ time slot) per update period.

3.3. Reason for Superior Performance of Kalman Method

Let us explain why the Kalman power control performs better than the G-SINR method in Figs. 3–5. Actually, these two methods are similar. To see that, using a fact that $\beta(n-1) = p(n-1)g(n-1)/i(n-1)$ where $i(n-1)$ is the actual interference power in mW for packet $n-1$, (10) becomes

$$p(n) = \min\{\gamma^* i(n-1)/g(n-1), p_{\max}\} \qquad (11)$$

If the path gain $g(n)$ does not change drastically from one packet to the next, (1) and (11) are similar, except that the Kalman method in (1) is based on interference prediction $\tilde{i}(n)$, while the G-SINR method uses the measured interference power for the last packet. In essence, the Kalman method provides some smoothing on interference measurements. If measurements contain errors, this smoothing effect can be important. Of course, similar smoothing can also be applied to the G-SINR method for performance improvement. Nevertheless, in the case of accurate measurements as assumed so far and path gains unchanged in time, the smoothing effect will not provide significant

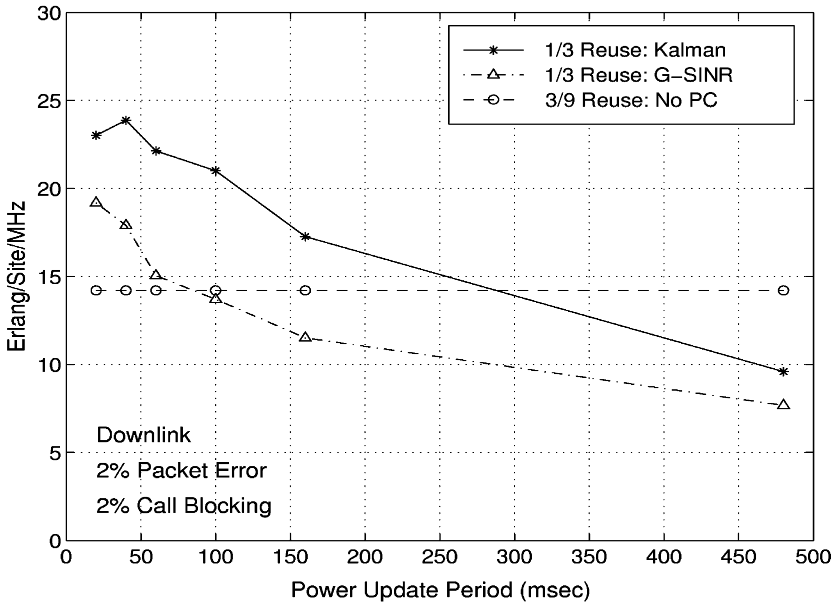

Fig. 5. Spectrum efficiency vs. update period for the EDGE system.

differences in performance. (We discuss impacts due to measurement errors later.)

On the other hand, with perfect measurements, the major difference between the Kalman and G-SINR method lie on the ways they choose power for the first packet of each talk spurt. For traditional circuit-switched voice, the selection of the first transmission power has little impacts on the overall performance because call holding time is much longer than the power update period to ensure "power convergence." However, for packet voice service with an average of 50 packets per talk spurt, the selection of the first transmission power becomes important. Since the Kalman method continuously tracks interference power, the power can be appropriately selected for the first packet according to (1). In contrast, the G-SINR method chooses the first power to fully compensate the signal path gain, which can be quite different from the appropriate power for combating interference. Thus, the G-SINR method does not perform as well as the Kalman method does for the packet voice service. The same comment applies to other power-control methods that are not based on the tracking of interference prior to the beginning of talk spurts.

3.4. Performance Impacts due to Measurement Errors

The Kalman power control can be further compared with another power-control method as follows. When a MS is not receiving any voice packet from its BS, it continuously measures the interference power and the power for the first packet of each talk spurt is determined based on the last interference measurement. However, once the talk spurt starts, the MS switches to measure SINR and adjusts power according to (10). Since power for the first and subsequent packets of each talk spurt is determined based on the interference (I) and SINR measurements, respectively, the power control is called the I-SINR method. As pointed out in the discussion for (11), transmission power during the talk-spurt period is basically adjusted using the interference measurement of the last packet. Thus, power is chosen according to signal path gain and interference power in the previous packet (frame), without filtering. This method may be simpler than the Kalman power control because it is possibly easier to measure SINR for the I-SINR scheme than to separate interference from signal during talk-spurt periods in the Kalman method. However, the smoothing function by the Kalman filter is helpful in case of measurement noise. By assuming Gaussian noise, we compare in Fig. 6 the coverage performance for the I-SINR and Kalman method as a function of interference-to-noise ratio (INR). For the reasons discussed above, the I-SINR and Kalman method yield very similar performance for high INR (e.g., above 20 dB). However, their performances differ as the

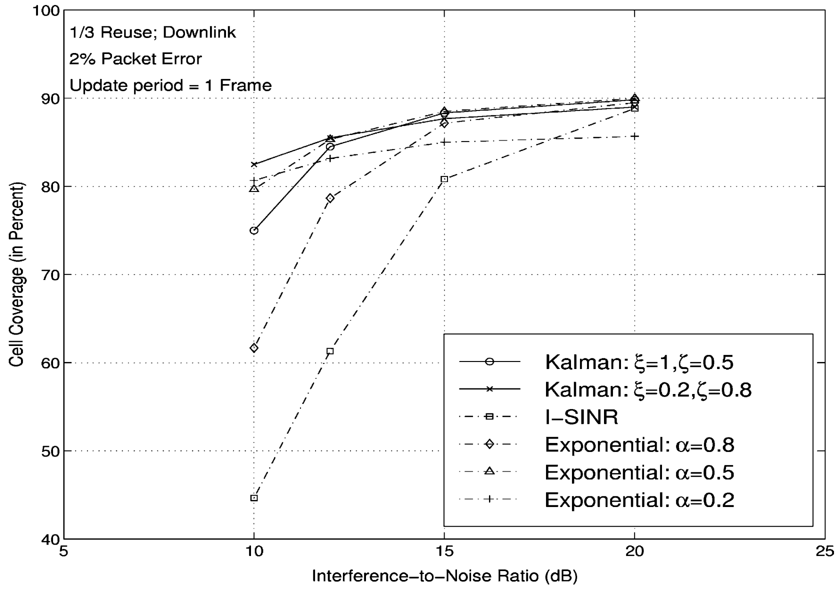

Fig. 6. Coverage performance vs. INR

INR decreases. Clearly, the Kalman filtering yields better performance than the I-SINR method for decreased INR. More significantly, these results show the importance of smoothing and tracking in presence of noise. Since noise also affects the accuracy of SINR measurements and since they are available only after talk spurts begin, the I-SINR method will not perform well at the beginning of a talk spurt when compared with the Kalman power control, which continuously tracks and smoothes interference power all the times. A similar comment also applies to the selection of appropriate power for the first packet of a talk spurt as the interference power is not tracked during talk spurts in the I-SINR method.

To further illustrate the need of filtering, we also include results for an exponential-smoothing method in Fig. 6. In this method, the interference power $\tilde{i}_e(n)$ is approximated by

$$\tilde{i}_e(n) = (1-\alpha)\tilde{i}_e(n-1) + \alpha Z_n \tag{12}$$

where Z_n is the measured interference power (noise included) for frame n and $\tilde{i}_e(n)$ is used in place of $\tilde{i}(n)$ in (1) to adjust power. In essence, the I-SINR method is a special case of the exponential-smoothing method with $\alpha = 1$. Figure 6 also presents the coverage performance for the exponential-smoothing method with selected α values. These results reveal that the performance of the exponential-smoothing technique varies, depending on the α parameter. The Kalman filter generally provides robust performance over a reasonable range of INR value, although the topic of comparing various filtering techniques for wireless packet networks will be an area of our future study.

4. CONCLUSION

Using the EDGE system with cyclic frequency hopping as an example, we have applied the Kalman-filter power control method [6] based on interference tracking to packet voice service. Our results reveal that the power-control method significantly improves the spectral efficiency by enabling the 1/3 frequency reuse while maintaining the stringent 2% packet loss probability and 90% coverage for voice service. Specifically, for allocated spectrum of 1.8, 3.6 and 5.4 MHz, the 1/3 reuse with the Kalman power control can yield 102.5%, 49.5% and 32.5% improvement in spectral efficiency, respectively, over the 3/9 reuse regardless of whether or not power control is used (because the capacity for the 3/9 reuse is blocking limited). We have also compared the capacity and coverage performance of the Kalman method with a traditional SIR method and a control method that is based on the last interference measurement. We found that appropriate selection of power for the first packet of each talk spurt and the filtering function for noisy measurements are crucial in providing high capacity for packet voice service. Specifically related to the EDGE system, our results reveal a need for shortening the power update period, which is 480 ms (i.e., one update per 24 voice packets) in the specifications.

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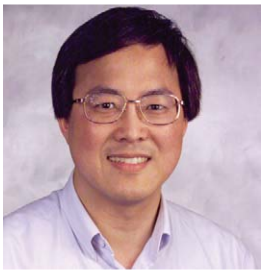

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