

# Quality of Service Assurance Model for AMR Voice Traffic in Downlink WCDMA System

Sung Hwan Jung<sup>1</sup> · Jung Wan Hong<sup>2†</sup> · Chang Hoon Lie<sup>1</sup>

<sup>1</sup>Department of Industrial Engineering, Seoul National University, Seoul 151-742, Korea

<sup>2</sup>Department of Industrial Systems Engineering, Hansung University, Seoul 136-792, Korea

## 순방향 WCDMA 채널에서 AMR 음성 트래픽의 품질 보증 모델

정성환<sup>1</sup> · 홍정완<sup>2</sup> · 이창훈<sup>1</sup>

<sup>1</sup>서울대학교 산업공학과 / <sup>2</sup>한성대학교 산업시스템공학과

We propose the QoS (Quality of Service) assurance model for AMR (Adaptive MultiRate) voice users considering the capacity and service quality jointly in downlink WCDMA system. For this purpose, we introduce a new system performance measure and the number-based AMR mode allocation scheme. The proposed number-based AMR mode allocation can be operated only with the information of total number of ongoing users. Therefore, it can be more simply implemented than the existing power-based allocation. The proposed system performance measure considers the stochastic variations of AMR modes of ongoing users and can be analytically obtained using CTMC (Continuous Time Markov Chain) modeling. In order to validate the proposed analytical model, a discrete event-based simulation model is also developed. The performance measure obtained from the analytical model is in agreement with the simulation results and is expected to be useful for parameter optimization.

**Keywords:** AMR, WCDMA, CTMC, System Performance, Event-based Simulation

## 1. Introduction

The telephone speech service will still remain one of the most important service in third generation system such as WCDMA (Wideband Code Division Multiple Access). Bit rate flexibility for the speech service is highly desirable because the speech service co-exists with other multimedia classes e.g. streaming and interactive bearer services.

The AMR (Adaptive Multi-Rate) codec, which is

mandatory speech codec in WCDMA system, can be used for providing adaptive service. The AMR codec provides 8 different modes with source bit rate ranging from 12.2 kbps down to 4.75 kbps. Introducing the AMR codec extends the service quality range for speech quality compared to a fixed source rate codec (Bruhn *et al.*, 1999; Corbun *et al.*, 1998).

Adaptation of WCDMA is typically a slow process with the time step of tens of second or more and the voice capacity approximately doubles when the bit rate is reduced from 12.2kbps to 4.75kbps

---

This research was financially supported by Hansung University in the year of 2006.

† Corresponding author : Professor Jung Wan Hong, Department of Industrial Systems Engineering, Hansung University, Seoul 136-792, Korea,  
Fax : +82-2-760-4490, E-mail : jwhong@hansung.ac.kr

Received June 2006; revision received September 2006; accepted October 2006.

in general situations (Holma and Toskala, 2004). Therefore, it is possible to achieve a trade-off between the network's capacity and speech quality according to the operator's requirements in WCDMA system with the AMR speech codec. For example, in the same system load, an operator may admit two users with lowest AMR bit rate rather than admit one user with highest AMR bit rate rejecting the service of the other user according to the network operation policy. However, in order to develop the policy for answering this question, the service quality and capacity have to be jointly evaluated and optimized.

The downlink capacity and performance studies for WCDMA system have been widely documented (Fu and Thompson, 2002; Hoz and Cordier, 1999; Jeon and Jeong, 2002; Kim *et al.*, 2003; Lee and Han, 2002; Malik and Zeghlache, 2002). Most of them are simulation-based studies and cannot give a clear answer to the optimal control parameters, which can be applied to real situations.

Karlsson *et al.*(2002) evaluate the different ways of allocating AMR rates to users on the downlink in a WCDMA system. System simulation shows that the AMR codec introduces a significant trade-off between capacity and quality for the speech service. However, this study adopts a simulation approach to analyze the performance of the AMR mode allocation and does not present the way of finding optimal control parameters in the aspect of QoS(Quality of Service) assurance.

In this paper, we propose the QoS assurance mo-

del for AMR voice users considering the capacity and service quality jointly. For this purpose, we introduce a new system performance measure and the number-based AMR mode allocation scheme. The proposed number-based AMR mode allocation can be operated only with the information of total number of ongoing users. Therefore, it can be more simply implemented than the existing power-based allocation. The proposed system performance measure considers the stochastic variations of AMR modes of ongoing users and can be analytically obtained in our proposed model. In order to validate the proposed analytical model, a discrete event-based simulation model is also developed. The performance measure obtained from the analytical model is in agreement with the simulation results and is expected to be useful for parameter optimization.

## 2. AMR in WCDMA

In the downlink of WCDMA system, the coverage depends more on the load than in the uplink, because the power is shared with the downlink users: the more users, the less power per user. In addition, the downlink capacity is more important than in the uplink because of its higher capacity requirements due to asymmetric service (Malik and Zeghlache, 2002). Therefore, this study focuses on the downlink AMR mode allocation considering the variation of the system loads.

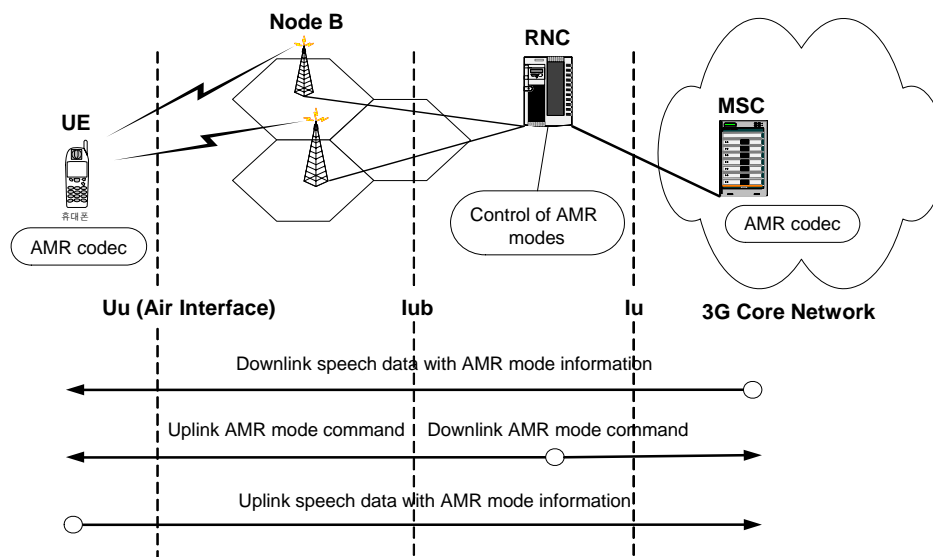


Figure 1. AMR control in WCDMA

Power is the most important radio resource in a WCDMA system in which different AMR modes require different amount of transmission power. The WCDMA air-interface has an in-built link quality control system compromising the fast power control and some kind of quality based outer loop control. The radio network dynamically controls the bit rate according to system loads in WCDMA. The admission control and rate control algorithms within the radio network generally govern AMR mode changes (3GPP TR 25.922 V3.1.0, 2000).

The control loop is shown in <Figure 1>. For a circuit switched telephony speech service, the AMR codec is located in the MSC (Mobile Switching Center) at the network side. In the downlink direction, AMR encoded speech frames are transferred over an Iu bearer service to the RNC (Radio Network Controller) and further over a RAB (Radio Access Bearer) service to reach the UE (User Equipment). It is up to the RNC to determine if the AMR RAB can be granted and what mode it should be initiated. To facilitate fast transparent AMR mode switches, both the Iu bearer and the RAB are configured to carry frame formats of different AMR modes. The RNC is in charge of allocating modes for new AMR RABs as well as adapting mode for established RABs.

For the downlink, the AMR mode command from RNC to TransCoder (TC) is realized as inband. The admission control functionality is located in RNC where the load information from several cells can be obtained.

### 3. Model Assumptions and System Descriptions

#### 3.1 Model Assumptions

In this paper, only the call level QoS is considered because the frame level QoS such as FER (Frame Erasure Rate) is assumed to be well maintained by the fast inner loop power control. To analyze the performance of AMR mode allocation in a voice class, other multimedia classes are not considered.

We adopt the homogeneous and isolated cell approach. This means that the new call arrival rate and the handoff call arrival rate into a cell can be

measured statistically and follow similar patterns with neighboring cells. New call requests and hand-off call arrivals are assumed to form a Poisson process with mean arrival rate  $\lambda^{NC}$  and  $\lambda^{HC}$  respectively. It is also assumed that the traffic is uniformly distributed between the cells.

The CHT (Call Holding Time) follows an exponential distribution with mean of  $1/\mu$ . The CRT (Cell Residence Time) is the amount of time that a mobile stays in a cell before handoff, is assumed to follow exponential distribution with mean of  $1/\eta$ . Then the cell departure rate of a call is  $\nu = \mu + \eta$ .

Let  $N = (n_1, \dots, n_I)$  be the system state vector that indicates the number of users in each AMR mode. The  $R^{(i)}$  means the bit rate of AMR mode  $i$  and it is greater than that of AMR mode  $i+1$ ,  $R^{(i+1)}$ . Mode 1 means the highest bitrate of 12.2 kbps. If it can be assumed that the average transmission power of the same AMR mode is same and the system state vector is known, the total mean transmission power of the system can be calculated by modifying the following downlink pole equation (Sipila *et al.*, 2000).

$$\overline{P_N} = \frac{P_{CCH} + P_{Noise} \sum_{i=1}^I \frac{\beta^{(i)} R^{(i)} \epsilon}{W} n_i \overline{L}}{1 - \sum_{i=1}^I \frac{\beta^{(i)} R^{(i)} \epsilon}{W} n_i (1 - \alpha + f_{DL})} \quad (1)$$

In equation (1),  $P_{CCH}$  is the common channel power,  $\beta^{(i)}$  is the  $E_b/N_o$  requirement for the AMR mode  $i$  user. Fast power control can obtain exactly the minimum average  $E_b/N_o$ .  $\epsilon$  is the effective channel activity factor for the voice user.  $\alpha$  is the orthogonal factor.  $P_{Noise}$  is the thermal noise power.  $\overline{L}$  is the average path loss from the serving Node B to a random user and can be obtained by monitoring the system.  $f_{DL}$  is the average other-to-own-cell interference ratio in downlink.

#### 3.2 Call Admission Control

Contrary to FDMA (Frequency-Division Multiple Access) and TDMA (Time-Division Multiple Access), there is not an absolute number of maximum available channels that can be allocated to potential users in CDMA (Sipila *et al.*, 2000).

The admission control algorithm estimates the load that the establishment of the bearer would cause in the radio network. This has to be estimated sepa-

rately for the uplink and downlink. In this paper, only downlink direction is considered.

We adopt a power based admission control (Knutsson *et al.*, 1998; Lee and Han, 2002). Handoff user has higher priority over new call requests. In order to support this purpose, handoff users are admitted if  $P_N < P_{Th}^{HC}$  and new call users are admitted if  $P_N < P_{Th}^{NC}$ . Here,  $Th$  denotes threshold and  $P_N$  is the transmitted output power from Node B when the system state is given as  $N$ .

### 3.3 Number-Based Mode Allocation Scheme

In (3GPP TR 26.975, 2002), the power-based dynamic mode allocation scheme is suggested. In this scheme, the code power of each user is measured in Node B and periodically reported to the RNC. The reported power value is compared to the threshold value of power to determine the AMR mode. However, this scheme requires an elaborate measurement mechanism to avoid the cost incurred by inaccurate monitoring. Estimating the probability distribution of power is difficult because the transmission powers are regulated by the threshold values of power. Therefore, it is not easy to determine the optimal power threshold values of power maximizing system performance while guaranteeing the certifiable QoS.

We newly propose a number-based dynamic mode allocation scheme considering the fact that the number of users in the system strongly influences system capacity or transmission power. The proposed number-based AMR mode allocation can be operated only with the information of total number of ongoing users. Therefore, this scheme can be more easily implemented and the optimal control parameters of this scheme can be easily found than of the existing power-based mode allocation. The objective of the proposed mode allocation is to guarantee the fairness, which is referred to be satisfied if all of the calls are assigned adjacent to the AMR mode while maximizing the power utilization. According to the definition of fairness, there are no calls that are assigned to more than one AMR mode difference. Theoretically, the AMR mode can be changed every 20ms. However, in practice, frequent mode changes invoke a cost in the aspect of system and degrade QoS for users. Therefore, the proposed mode allocation algorithm

is assumed to be triggered when the call acceptance and departure events occur.

The allocation algorithm allows calls to move only to an adjacent mode at each allocation event to prevent users from experiencing severe fluctuations of adaptation. Once a call moves to an adjacent mode, it can move again after all other calls from its mode have moved to the new mode.

Note that the system vector  $N$  can be uniquely calculated according to our mode allocation scheme when the total number of voice users is determined.

Let  $\bar{p}_{N,i}$  be the average power of transmitting one user of AMR mode  $i$  when the system state is  $N$ .  $i^*$  and  $i^* + 1$  for  $i^* \in \{1, \dots, I-1\}$  be the AMR modes with calls. When the total voice user is fixed as  $n$ , the distribution of number of AMR mode users can be determined by simultaneously finding minimum integer  $i^*$  and maximum integer  $x^*$  satisfying the following equation.

$$\begin{aligned} \bar{P}_N &= \bar{p}_{N,i^*} \cdot x^* + \bar{p}_{N,i^*+1} \cdot (n - x^*) + P_\delta \\ (0 \leq P_\delta < \bar{p}_{N,i^*}), \quad i^* \in \{1, \dots, I-1\}, \quad x^* \in \{0, \dots, n\} \end{aligned} \quad (2)$$

In equation (2),  $N$  is the vector of which  $i$ th element is  $x$  and  $(i+1)$ st element is  $n-x$  and the others are all zeros. Here  $\bar{P}_N$ ,  $\bar{p}_{N,i^*}$ ,  $\bar{p}_{N,i^*+1}$  can be obtained by using equation (1). As a result, the elements of system vector can be obtained as follows

$$n_i = \begin{cases} n - x^*, & i = i^* + 1 \\ x^*, & i = i^* \\ 0, & i \neq i^*, i^* + 1 \end{cases} \quad (3)$$

## 4. System Performance Measure

As mentioned in the introduction, AMR speech quality and cellular system capacity are dependent and thus have to be optimized jointly. While we can easily estimate the system capacity in a statistical framework, this is not the case with speech service quality because the service quality is the subjective term based on the user perception. In this paper, we assume that NBP (New call Blocking Probability) and HDP (Handoff Dropping Probability) are the major sources of the user dissatisfaction in call level service quality. It is also assumed

that the US(User Satisfaction) of ongoing call depends on the assigned AMR codec mode and FER level, of which functional relation is derived from the test results of AMR codec characterization (3GPP TR 26.975, 2002). Fixing the FER value as a constant, the relative speech quality of AMR codec can be represented as a function of AMR mode  $i$ ,  $Q_{rel}(i)$ . A dropped call has higher impact on the user satisfaction than unacceptable speech quality. Blocking is assumed to be less irritating than a dropped call.

Using the QoS measures mentioned above,  $SP$  (System Performance), which can be derived by analytic method, is proposed as follows.

$$SP = US - c_B \cdot [w \cdot NBP + (1 - w) \cdot HDP] \quad (4)$$

Here,  $US$  is the average user satisfaction experienced by a tagged user during the conversation.  $c_B$  is the relative cost of call blocking to the revenue of the user satisfaction and  $w$  represents the portion of new calls in the total cost of call blocking.

## 5. Performance Analysis

### 5.1 Calculation of NBP and HDP

In the assumption of number-based mode allocation and stationary traffic load situations, the transmission power of each state follows independent and identical probability distribution. By approximating the distribution using the central limit theorem, we have the probability distribution of each state power as  $P_N \sim N(\overline{P}_N, \sigma_N^2)$ . Here,  $\overline{P}_N$  and  $\sigma_N^2$  can be estimated by monitoring the system in real situations.

Under the proposed mode allocation scheme, the system state follows a pure birth and death process with input rate  $\lambda(n)$  and output rate  $n \cdot \nu$ . Note that state vector  $N$  is uniquely determined when the total number of users  $n$  is assigned according to the proposed mode allocation scheme.  $\lambda(n)$  can be obtained using the following equation.

$$\lambda(n) = A^{NC}(N) \cdot \lambda^{NC} + A^{HC}(N) \cdot \lambda^{HC} \quad (5)$$

In (5),  $A^{NC}(N)$  is the probability, which a sys-

tem admit a new call user when the system state is  $N$ .  $A^{HC}(N)$  is the analogous probability for the handoff case. These probabilities are obtained as follows:

$$A^{NC}(N) = \Pr[P_N < P_{Th}^{NC}] = Q\left[\frac{P_{Th}^{NC} - \overline{P}_N}{\sigma_N}\right] \quad (6)$$

$$A^{HC}(N) = \Pr[P_N < P_{Th}^{HC}] = Q\left[\frac{P_{Th}^{HC} - \overline{P}_N}{\sigma_N}\right] \quad (7)$$

$$\text{Here, } Q[z] = \int_{-\infty}^z \frac{1}{\sqrt{2\pi}} e^{-w^2/2} dw.$$

The stationary probabilities are calculated as

$$\pi(n) = \pi(0) \cdot \prod_{k=0}^{n-1} \frac{\lambda(k)}{(k+1) \cdot \nu}, \quad n=1, \dots, M \quad (8)$$

$$\pi(0) = \frac{1}{1 + \sum_{n=1}^M \prod_{k=0}^{n-1} \frac{\lambda(k)}{(k+1) \nu}} \quad (9)$$

In equation (8),  $M$  represent the maximum allowable number of users. It can be estimated by using downlink pole equation of a multi-cell case as in equation (10) or can be obtained by monitoring the real system.

$$M = \left\lfloor \frac{P^S - P_{CCH}}{\frac{\beta^{(l)} R^{(l)} \epsilon}{W} [P_{Noise} \cdot \overline{L} + (1 - \alpha + f_{DL}) \cdot P^S]} \right\rfloor \quad (10)$$

In equation (10),  $P^S$  denotes the transmission power capacity for voice users. Then NBP and HDP can be calculated as follows:

$$NBP = \sum_{n=0}^M \{1 - A^{NC}(n)\} \cdot \pi(n) \quad (11)$$

$$HDP = \sum_{n=0}^M \{1 - A^{HC}(n)\} \cdot \pi(n) \quad (12)$$

### 5.2 Calculation of US

In order to estimate the  $US$  strictly from the service user's perspective, we should estimate the dwelling times with a certain AMR mode. Therefore, one incoming call is chosen at random time and its state transitions are observed until it departs

from a cell. Hereafter we call this a tagged call. The system state variable with the tagged call,  $n$ , changes according to the CTMC (Continuous Time Markov Chain) with one absorbing state. The system enters the absorbing state when a tagged call departs from the cell.

The  $US$  value is dependent on the ratio of the time length a tagged call stayed at each AMR mode during its lifetime. Dwelling time of a tagged call in a certain AMR mode is dependent on the initial number of ongoing calls. The expected  $US$  can be expressed as follows.

$$US = \sum_{n=1}^M US(n) \cdot Pr(n) \tag{13}$$

In equation (13),  $Pr(n)$  means the probability that the number of voice user is  $n$  including the tagged call and can be calculated as equation (14).

$$Pr(n) = \frac{\pi(n)}{1 - \pi(0)} \tag{14}$$

In equation (13),  $US(n)$  is the  $US$  value on condition that the initial number of ongoing calls is  $n$  and is obtained as follows:

$$US(n) = \frac{1}{\tau} \sum_{i=1}^I Qrel(i) \cdot \tau_{i|n} \tag{15}$$

Here,  $\tau$  is a call's lifetime in a cell and it is certain that  $\tau = \sum_{i=1}^I \tau_{i|n} = 1/\nu$  for any  $n$ .

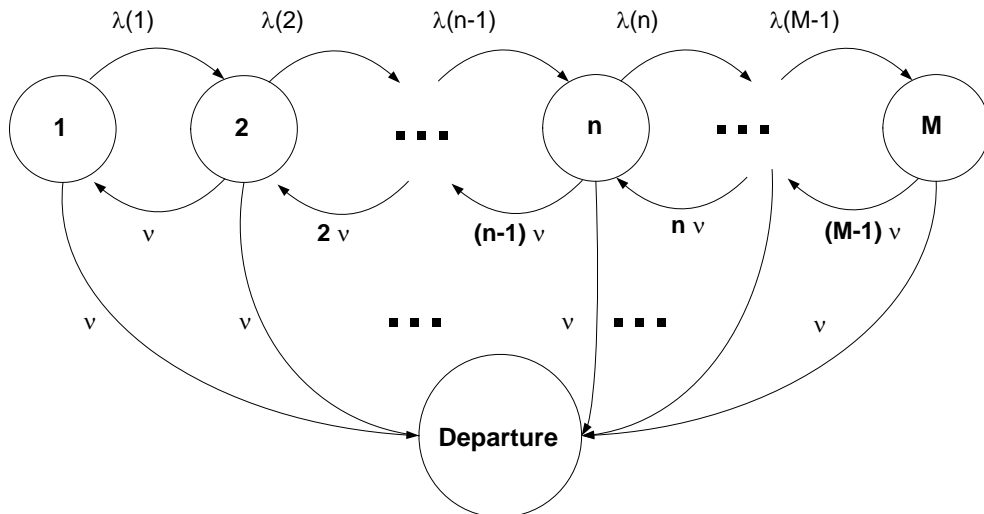


Figure 2. State transition diagram including a tagged call of AMR voice

$\tau_{i|n}$  is the conditional expected time the tagged call spends at AMR mode  $i$  until departure when the initial state is given as  $n$ . It is calculated as

$$\tau_{i|n} = \sum_{n^*=1}^M P_i(n^*) \cdot y_{n^*|n} \tag{16}$$

In equation (16),  $y_{n^*|n}$  is the conditional expected time the CTMC spends in state  $n^*$  until the tagged call's departure when an initial state is  $n$ . It can be estimated by the following procedures.

Let  $Q$  denote the infinitesimal generator matrix of CTMC of <Figure 2> and  $\hat{Q}$  denote the sub matrix of  $Q$  excluding the one column of absorbing state. Kolmogorov's equations in matrix form can be written as follows (Trivedi, 2002).

$$\frac{d \hat{\pi}_t}{dt} = \hat{\pi}_t \cdot \hat{Q} \tag{17}$$

$\pi_t$  is the probability vector and  $\hat{\pi}_t$  is the sub vector of  $\pi_t$  pertaining to only transient states at time  $t$ . Integrating equation (15) by  $t$  and letting  $t \rightarrow \infty$ , we have equation (18).

$$Y \cdot \hat{Q} = -\hat{\pi}_0 \tag{18}$$

In equation (18),  $Y = \{y_{n^*|n}\}$  and  $\hat{\pi}_0$  is the probability vector representing the initial state at  $t = 0$ . By solving equation (18) when an initial state is fixed as  $n$ ,  $Y$  can be obtained.

In equation (16),  $P_i(n^*)$  is the probability that the tagged user is at AMR mode  $i$  when the number of users is  $n^*$  and is calculated simply as

$$P_i(n^*) = \frac{n_i^*}{n_i} \quad (19)$$

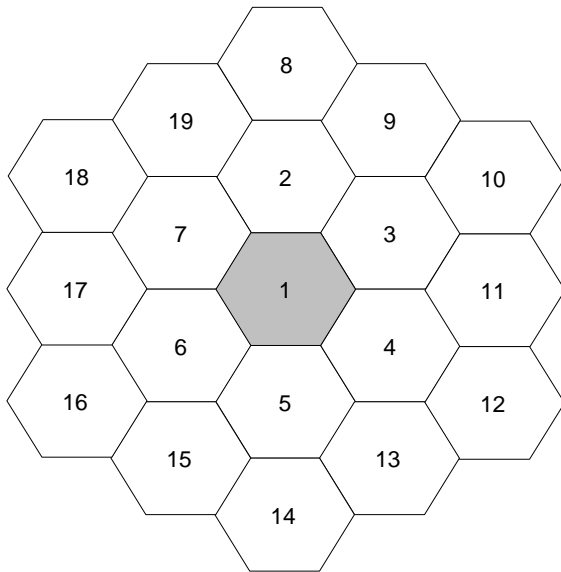
Then  $\tau_{in}$  can be finally obtained using equation (16), equation (18) and equation (19).

## 6. Numerical Results

### 6.1 Simulation Model

To verify the analytic results, a discrete event-based simulation model is developed. The simulation platform models a population of users evolving in a hexagonal cellular structure. The network system model consists of second tiers hexagonal cells (omni-directional antenna) with the tagged cell at their center as in <Figure 3>.

Initially, the population is uniformly generated within cell and in each observing time user's profile parameters are re-initialized. Mobility is not considered but the locations of users are generated randomly in the cell at each event time.



**Figure 3.** Cellular network structure

<Table 1> gives the system parameters used throughout this work. While the Node B maximum transmit power is typically 20W (43dBm), this study

**Table 1.** System parameters in simulations

System Parameter	Values
Chip Rate( $W$ )	3.84 Mcps
Orthogonal Factor( $\alpha$ )	0.5
Noise Power( $P_{Noise}$ )	-100 dBm
Maximum Transmission Power( $P^S$ )	10 W
Common Channel Power( $P_{CCH}$ )	30%
Distance Between Node B	1.2 Km
$c_B$	3
$w$	0.2

assumes 10W for the reduction of time complexity in simulation.  $C_B$  and  $w$  which are proposed in our modeling can be determined by the system's operation policy. In <Table 1>,  $C_B = 3$  means that the relative cost of call blocking to the revenue of the user satisfaction is triple.  $w = 0.2$  represents that the portion of new calls in the total cost of call blocking is 0.2, i.e. the cost of a handoff dropping is four times that of a new call blocking. Refer to (Holma and Toskala, 2004) for more information about the parameters presented in Table 1. To emulate a network pattern as realistic as possible, the wrap around technique is implemented. Full speech codec activity, i.e. no DTX (Discontinuous Transmission) is assumed.

The channel is characterized by a propagation model that follows the 3GPP specifications described in (Sallabi *et al.*, 2005). The base station antenna height is fixed at 15 meters above the average rooftop. By considering a carrier frequency of 2 GHz, the propagation model formula is

$$L_{j,i} = 128.1 + 37.6 \text{Log}_{10}(D_{j,i}) \quad (20)$$

In equation (20),  $D_{j,i}$  (kilometers) is the distance between  $j$ th base station-mobile station and  $i$ th UE of tagged cell and  $L_{j,i}$  is the path loss. After  $L$  is calculated, log-normal distributed shadowing with mean of 0dB and standard deviation of 10dB is added.

New incoming calls are accepted with the lowest mode regardless of the call types or handoff call and new incoming call.

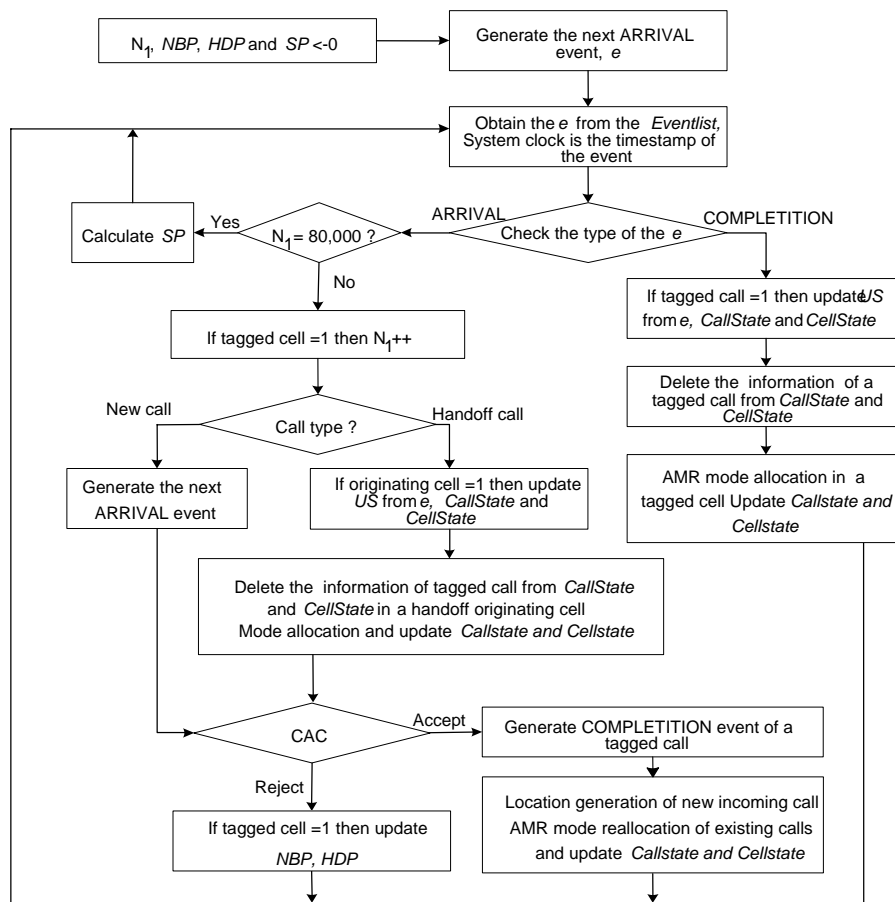
In <Table 2>,  $E_b/N_0$  is the measure of the signal to noise ratio and is used as the basic measure of how strong the signal is. It is taken into account that the physical layer overhead increases for the

lower AMR mode and that is modeled in <Table 2> as a higher  $E_b/N_0$ . The relative quality levels of all AMR modes (Qrel(i)) can be expressed as a relative metric with respect to the highest speech quality of AMR 12.2kbps, which MOS (Mean Opinion Score) is 4.01 in clean speech and error free condition. The experimental MOS data is available in (3GPP TR 26.975, 2002). In this paper, the outer loop power control is assumed to set to maintain a FER-target of 0.5% for all AMR modes.

**Table 2.** Relative QoS values (Qrel)

Mode : i	1	2	3
Bitrate	12.2 kbps	7.4 kbps	4.75 kbps
Required $E_b/N_0$	8.0 dB	7.5 dB	7.0 dB
( $\Delta$ MOS) FER(0.5%)	0.12	0.25	0.66
(MOS) FER(0.5%)	3.89	3.76	3.35
Qrel(i)	0.97	0.937	0.835
Quality level	Good	Good	acceptable

There are two types of event in this simulation: the *ARRIVAL* event for calls' arrivals and the *COMPLETION* event for calls' departures from the cell. These events are inserted into *eventlist* and deleted from the *eventlist* in the non-decreasing *timestamp* order. A *SYSTEM CLOCK* is maintained to indicate the progress of simulation. The *SYSTEM CLOCK* value is the *timestamp* of the event being processed. The simulation counts  $N_1$ , the total number of call arrivals. Simulation iteration terminates when  $N_1 = 80,000$ . For an *ARRIVAL* event, the arrival type (either *NEW* or *HAND-OFF*) is determined by using a uniform random number generation in the interval [0,1]. If the value is less than  $\eta/(\mu + \eta)$  then the arrival type is *HAND-OFF*, otherwise the type is *NEW*. The system states are maintained by three vectors; *CellState* of each cell, *CallState* of each call and the *eventlist*. The *CellState* has the information on the total number of user in its cell, each number of AMR mode and the total transmission power.



**Figure 4.** Simulation flow chart



The *CallState* has the information of each call, the location of a call, the connection power, the AMR mode and the accumulated time spent in each AMR mode etc. When the simulation iteration terminates, the *SP* is calculated using the obtained *NBP*, *HDP* and *US* values.

## 6.2 Numerical Examples

In this section, the performance of the proposed AMR mode allocation scheme is compared with the fixed AMR mode schemes by simulation experiments. The QoS measures derived from proposed analytical model are illustrated compared with those from the simulation model.

<Figure 5> shows the *SP* versus offered load for fixed mode and our proposed number-based mode allocation. Whereas the static modes perform well in certain load intervals, high *SP* is achieved in all load regions with the proposed scheme.

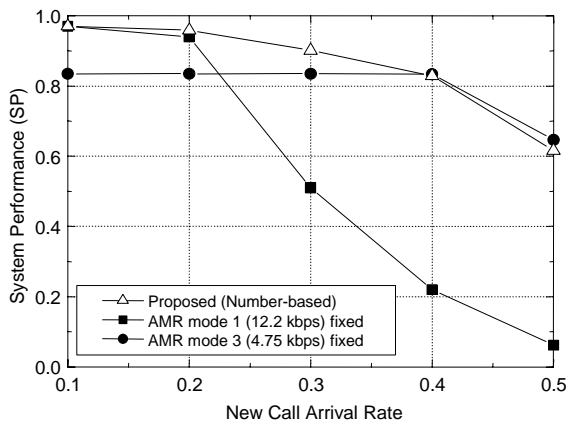


Figure 5. System performance versus offered load

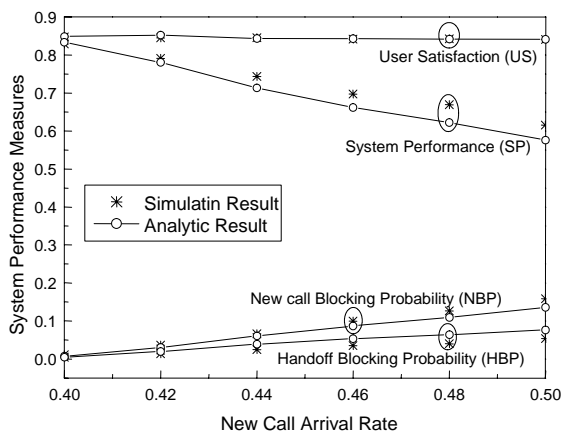


Figure 6. System performance measures of simulation and analytic model

Simulation and analytical results are compared in <Figure 6>. The parameters  $\lambda^{HC}$  and  $P_N$  are obtained from simulation results and then used as input data in the analytical model. The QoS measures of the proposed analytical method are in agreement with the simulation results in the normal traffic load situation.

Using the proposed number-based AMR allocation scheme and system performance measure, we can find the optimal call admission control parameter such as new call threshold power. <Figure 7> shows the variation of the system performance with the new call threshold power when the new call arrival rate is 0.5. The optimal new call threshold power can be found as 8.9W.

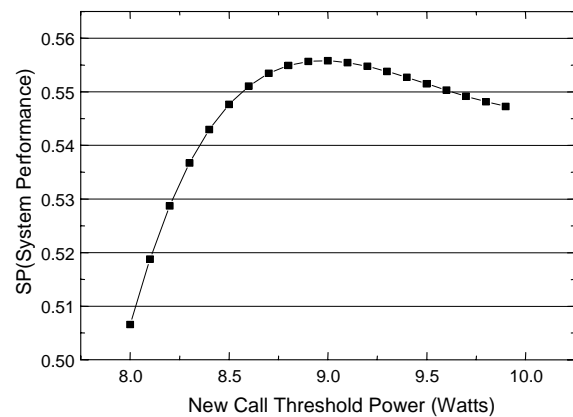


Figure 7. System performance vs. new call threshold power

## 7. Conclusions

We propose QoS assurance schemes for AMR voice users considering the capacity and service quality jointly. In specific, new system performance measure and the number-based AMR mode allocation scheme are presented. The proposed system performance measure considers the stochastic variations of AMR modes of ongoing users. The proposed number-based AMR mode allocation can be operated only with the information of total number of ongoing users. Therefore, it can be more simply implemented than the existing power based mode allocation. Simulation model is developed to compare the exactness of the analytic method. The accuracy of the analysis by the proposed Markov model is corroborated by the means of the simu-

lation results. A procedure for defining the optimal control parameters is also presented in the numerical results.

## References

- 3GPP TR 25.922 V3.1.0: Radio Resource Management Strategies, (2000).
- 3GPP TR 26.975: Performance Characterization of the AMR Speech Codec, (2002).
- Bruhn, S., Blocher, P., Hellwig, K., and Sjoberg, J. (1999), Concepts and solutions for link adaptation and inband signaling for the GSM AMR speech coding standard, *IEEE 49th Vehicular Technology Conference*, **3**, 2451-2455.
- Corbun, O., Almgren, M., and Svanbro, K. (1998), Capacity and speech quality aspects using adaptive multi-rate(AMR), *The 9th IEEE International Symposium on Personal, Indoor and Mobile Radio Communications*, **3**, 1535-1539.
- Dimitriou, N., Tafazolli, R. and Sfikas, G. (2000), Quality of service for multimedia CDMA, *IEEE Communications Magazine*, **38**(7), 88-94.
- Fu, H. and Thompson, J. S. (2002), Downlink capacity analysis in 3GPP WCDMA networks system, *Third International Conference on 3G Mobile Communication Technologies*, 534-538.
- Holma, H. and Toskala, A. (2004), *WCDMA for UMTS, 3rd Ed.*, John Wiley & Sons, New York.
- Hoz, D. and Cordier, C. (1999), W-CDMA downlink performance analysis, *IEEE 50th Vehicular Technology Conference*, **2**, 68-972.
- Jeon, W. S. and Jeong, D. G. (2002), Call admission control for CDMA mobile communications systems supporting multimedia services, *IEEE Transactions on Wireless Communications*, **1**(4), 649-659.
- Karlsson, M., Almgren, M., Bruhn, S., Larsson, K., and Sundelin, M. (2002), Joint Capacity and Quality Evaluation for AMR Telephony Speech in WCDMA Systems, *IEEE 56th Vehicular Technology Conference*, **4**, 24-28, 2046-2050.
- Kim, D. I., Hossain, E. and Bhargava, V. K. (2003), Downlink joint rate and power allocation in cellular multirate WCDMA systems, *IEEE Transactions on Wireless Communications*, **2**(1), 69-80.
- Kleinrock, L. (1975), *Queueing Systems, vol I*, John Wiley & Sons.
- Knutsson, J., Butovitsch, P. Persson, M., and Yates, R. D. (1998), Downlink Admission Control Strategies for CDMA Systems in a Manhattan Environment, *IEEE Vehicular Technology Conference*, Ottawa, Canada, **2**, 1453-1457.
- Lee, J. and Han, Y. (2002), Downlink admission control for multimedia services in WCDMA, *The 13th IEEE International Symposium on Indoor and Mobile Radio Communications*, **5**, 2234-2238.
- Malik, S. A. and Zeghlache, D. (2002), Resource allocation for multimedia services on the UMTS downlink, *IEEE International Conference on Communication*, **5**, 3076-3080.
- Sallabi, F., Lakas, A., Shuaib, K. and M. (2005), Boulmalf, WCDMA downlink simulator with efficient wrap-around technique, *WOCN 2005, Wireless and Optical Communications Networks*, 472-479.
- Sipila, K., Honkasalo, K. C., Laiho-Steffens, J., and Wacker, A. (2000), Estimation of Capacity and Required Transmission Power of WCDMA Downlink Based on a Downlink Pole Equation, *IEEE Vehicular Technology Conference*, Tokyo, **2**, 15-18, 1002-1005.
- Trivedi, K. S. (2002), *Probability and Statistics with Reliability, Queuing and Computer Science Applications, 2nd. Ed.* John Wiley & Sons, New York.