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CALL ADMISSION CONTROL FOR INTEGRATED ON/OFF VOICE AND BEST-EFFORT DATA SERVICES IN MOBILE CELLULAR COMMUNICATIONS

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BONAFIDE CERTIFICATE

Certified that this project report titled "**CALL ADMISSION CONTROL FOR INTEGRATED ON/OFF VOICE AND BEST-EFFORT DATA SERVICES IN MOBILE CELLULAR COMMUNICATIONS**" is the bonafide work of **Mr.P.Bhagathsingh [Reg. No.0920107003]** who carried out the research under my supervision. Certified further, that to the best of my knowledge the work reported herein does not form part of any other project or dissertation on the basis of which a degree or award was conferred on an earlier occasion on this or any other candidate.

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ABSTRACT

Call Admission Control (CAC) is a key for ensuring the quality of service in wireless multiservice network. With the advances in wireless communication technology and the growing interest in deploying multimedia services in wireless networks, the issue of providing an efficient CAC has come to the force. A suitable CAC for the multimedia service networks is expected to make efficient use of the scarce wireless resource while supporting different services with different QoS metrics. In this project, a CAC model is proposed with four priority levels supporting five classes of service (e.g. UGS, ErtPS, rtPS, nrtPS and BE) in a wireless multiservice network: IEEE 802.16e. The idea is to design an efficient CAC that deals with new and handover (HO) calls. The proposed model manages to establish priority between new and handover calls. It is assumed that the system operates under a reservation channel scheme and a queuing strategy in order to maintain the HO priority. This model must maintain a balance between two conflicting requirements: maximize the resource utilization and minimize the forced handover call dropping rate. In order to maintain the maximum resource utilization, the maximum number of calls should be admitted into a network which may result in unacceptably high HO call dropping rates due to insufficient resources for HO calls. It is very important to propose a CAC model with reserves the minimum amount of necessary resources to maintain an acceptable HO call dropping and provide high resource utilization.

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LIST OF ABBREVATIONS

CAC	**Call admission control**
QoS	**Quality of service**
UGS	**Unsolicited Grant Service**
ertPS	**Extended Real-time Polling Service**
rtPS	**Real-time Polling Service**
nrtPS	**Non-real-time Polling Service**
BE	**Best Effort**
HO	**Handover Calls**
CDMA	**Code Division Multiple Access**
GC	**Guard Channel**
FGC	**Fractional Guard Channel**
LFGC	**Limited Fractional Guard Channel**
UFGC	**Uniform Fractional Guard Channel**
WiMAX	**Worldwide Interoperability for Microwave Access**
FIFO	**First In First Out**
WFQ	**Weighted Fair Queuing**

CHAPTER 1
INTRODUCTION

With the constant improvement of wireless technology and the explosive growth of wireless communication market, the demand for newer multimedia applications is increasing rapidly. The convergence of wireless technology and multimedia applications presents network operators with enormous opportunities as well as great challenges. Qualities of Service (QoS) provisioning and mobility management are two key challenging issues that must be addressed in wireless multi services networks. QoS provisioning in wireless networks supporting multimedia applications have to meet the expectations of users while maintaining reasonably high utilization of radio resources.

The QoS provisioning problem is more challenging than in fixed networks for two main reasons. First, the link bandwidth resource is limited in a wireless environment. Second the changing environment in wireless networks due to the user's mobility and interference results in varying bandwidth. Thus, how to allocate and how to use the limited wireless resources efficiently are to be studied. On the other hand, the mobility management is another important issue to be addressed when studying the wireless multi services networks. A mobile user will be able to freely move across the networks while maintaining its current communications. An interrupted communication is a very frustrating phenomenon that may happen to a user. Thus, an efficient Call admission control (CAC) protocol must manage to avoid the forced termination of an ongoing call.

CAC schemes are critical to the success of future generations of wireless networks. On one hand, CAC schemes provide the users with access to a wireless network for services. On the other hand, they are the decision making part of the network carriers with the objectives of providing services to users with guaranteed quality and at the same time, achieving as much as possible resource utilization. It is therefore conceivable that CAC policy is one of the critical design considerations in any wireless networks. The design of modern wireless networks is based on a cellular architecture that allows efficient use of the limited available spectrum. The cellular architecture consists of a backbone network with fixed base stations interconnected through a fixed network (usually wired) and of mobile units that communicate with the base stations via wireless links. The geographic area within which mobile units can communicate with a particular base station is referred to as a cell. Neighboring cells overlap with each other, thus ensuring continuity of communications when the users move from one cell to another. The mobile units communicate with each other, as well as with other networks, through the base stations and the backbone network. A set of channels (frequency bands or codes) is allocated to each base station.

When a mobile user wants to communicate with another user or a base station, it must first obtain a channel (or code) from one of the base stations that hear it (the best). If a channel is available, it is granted to the user. In the case that all the channels are busy, the new call is blocked. This kind of blocking is called new call blocking. The user releases the channel under either of the following scenarios: (i) The user completes the call; (ii) The user moves to another cell before the call is completed. The procedure of moving from one cell to another, while a call is in progress, is called handoff. While performing handoff, the mobile unit requires that the base station in the cell that it moves into will allocate it a channel. If no channel is available in the new cell, the handoff call is blocked. This kind of blocking is called handoff blocking. The motivation for many studies on the new call and handoff blocking is that the QoS in cellular networks is mainly determined by these two blocking probabilities. The first determines the fraction of new calls that are blocked, while the second is closely related to the fraction of admitted calls that terminate prematurely due to dropout.

CAC is a key element in the provision of guaranteed quality of service in wireless networks. The design of call admission control algorithms for mobile cellular networks is especially challenging given the limited and highly variable resources, and the mobility of users encountered in such networks. The Goal is to provide a broad classification and thorough discussion of existing call admission control schemes. In addition to this, it is presented some modeling and analysis basics to help in better understanding the performance and efficiency of admission control schemes in cellular networks.. Handoff prioritization is the common characteristic of admission schemes. Moreover, optimal and near-optimal reservation schemes are presented.

QoS provisioning in wireless networks is a challenging problem due to the scarcity of wireless resources, i.e. radio channels, and the mobility of users.CAC is a fundamental mechanism used for QoS provisioning in a network. It restricts the access to the network based on resource availability in order to prevent network congestion and service degradation for already supported users. A new call request is accepted if there are enough idle resources to meet the QoS requirements of the new call without violating the QoS for already accepted calls. With respect to the layered network architecture, different quality of service parameters is involved at different layers. At physical layer, bit-level QoS parameters such as bit energy-to-noise density describe the quality of service a mobile user receives. In packet-based communication systems, packet-level QoS parameters such as packet loss, delay and jitter characterize the perceived QoS. However, most of the existing research on CAC in cellular networks has focused on an abstract representation of the network in which only call-level QoS parameters, namely, call blocking and dropping probabilities are considered.

The next generation mobile communications networks will provide multimedia services, e.g., voice and video telephony, high-speed Internet access, mobile computing, etc. A representative example for such a system is International Mobile Telecommunications-2000 under the standardization process of the International Telecommunication Union. Mobility management for providing seamless multimedia communication is one of the most important engineering issues in such a next generation mobile network. The concept of mobility management includes both handoff and location management. Location management is a basic function to deliver incoming calls appropriately to the called mobile roaming from place to place. The handoff is an essential function for permitting users to move from cell to cell with an ongoing call.

There are two major engineering issues concerning handoff. The first is the handoff initiation process. Usually, two types of initiation processes are considered: that based on signal strength and that based on carrier-to-interference ratio. That is, in general, the handoff initiation process is related heavily to radio propagation effects. In this project, the radio propagation effects on the handoff initiation process are taken into account in the call statistics (e.g., the distribution of channel holding time). The second is CAC, which is related to the network resource management. There are two kinds of call request: new call and handoff call. In resource sharing between the call requests, since premature termination of connected calls is usually more undesirable than rejection of a new call request, it is widely accepted that a system should give higher priority to handoff call requests as compared to new call requests. This project concentrates on attention on the handoff mechanism from the aspect of resource management and CAC. Specifically, an advanced CAC scheme for mobile multimedia communications is proposed. The special concerns in designing the scheme are:

1) The priority mechanism between new and handoff calls.
2) The situation of traffic (load) asymmetry between uplink (from mobile to base) and downlink (from base to mobile) in mobile multimedia communications, which will be discussed in detail in the next section.
3) The easy implementation process is simpler.

1.1 SOFTWARE USED

This project is implemented in MATLAB 7.9 or the above version.

1.2 ORGANIZATION OF THE REPORT

- **Chapter 2** discusses about the parameter the wireless network parameters.
- **Chapter 3** discusses about the basics and types of Call Admission Control.
- **Chapter 4** discusses about Fractional Guard Channel and its types.
- **Chapter 5** discusses about Quality of Services.
- **Chapter 6** discusses about WiMAX and its standards.
- **Chapter 7** discusses about Queuing Algorithms.
- **Chapter 8** discusses about the problem involved in WiMAX communication
- **Chapter 9** discusses about the proposed model.
- **Chapter 10** discusses about the CAC model implementation.
- **Chapter 11** discusses about the results.
- **Chapter 12** discusses about the conclusion and future scope of the project.

CHAPTER 2

WIRELESS NETWORK PARAMETERS

The following terms are used in the literature to be used throughout this project.

2.1 CALL HOLDING TIME

The duration of the requested call connection. This is a random variable which depends on the user behavior (call characteristics).

2.2 CELL RESIDENCY TIME

The amount of time a mobile user spends in a cell. Cell residency is a random variable which depends on the user behavior and system parameters, e.g. cell geometry.

2.3 CHANNEL HOLDING TIME

How long a call which is accepted in a cell and is assigned a channel will use this channel before completion or handoff to another cell. This is a random variable which can be computed from the call holding time and cell residency time and generally is different for new calls and handoff calls.

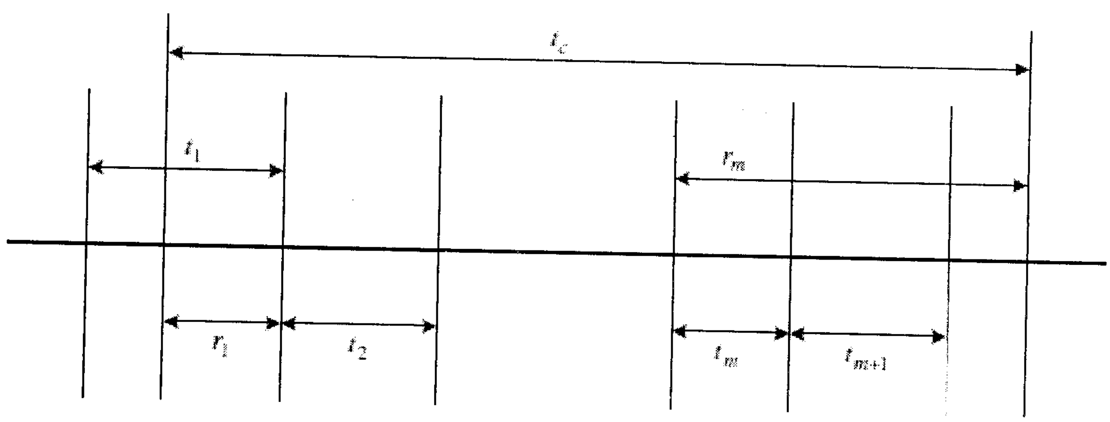

Fig 2.1 Diagram for Call Holding Time and Cell Residence Time

The above diagram shows a time diagram for call holding and cell residency times. Let t_c be the call holding time for a typical new call, tm be the cell residency time, r_1 be time between the instant the new call is initiated at and the instant the new call moves out of the cell if the new call is not completed, and r_m ($m > 1$) be the residual life of call holding time when the call finishes the m-th handoff successfully.

2.4 CALL DROPPING AND HANDOFF FAILURE

When a mobile terminal (mobile user) requests service, it may either be granted or denied Service. This denial of service is known as call blocking, and its probability as call blocking probability (p_b). An active terminal in a cellular network may move from one cell to another. The continuity of service to the mobile terminal in the new cell requires a successful handoff from the previous cell to the new cell. A handoff is successful if the required resources are available and allocated for the mobile terminal. The probability of a handoff failure is called handoff failure probability (p_f). During the life of a call, a mobile user may cross several cell boundaries and hence may require several successful handoffs. Failure to get a successful handoff at any cell in the path forces the network to discontinue service to the user. This is known as call dropping or forced termination of the call and the probability of such an event are known as call dropping probability (p_d). In general, dropping a call in progress is considered to have a more negative impact from the user's perspective than blocking a newly requested call.

2.4.1 HANDOFF SCHEMES

The handoff schemes can be classified according to the way the new channel is set up and the method with which the call is handed off from the old base station to the new one. At call-level, there are two classes of handoff schemes, namely hard handoff and soft handoff.

1) Hard handoff:

In hard handoff, the old radio link is broken before the new radio link is established and a mobile terminal communicates at most with one base station at a time. The mobile terminal changes the communication channel to the new base station with the possibility of a short interruption of the call in progress. If the old radio link is disconnected before the network completes the transfer, the call is forced to terminate. Thus, even if idle channels are available in the new cell, a handoff call may fail if the network response time for link transfer is too long. Second generation mobile communication systems based on GSM fall in this category.

2) Soft handoff:

In soft handoff, a mobile terminal may communicate with the network using multiple radio links through different base stations at the same time. The handoff process is initiated in the overlapping area between cells some short time before the actual handoff takes place. When the new channel is successfully assigned to the mobile terminal, the old channel is released. Thus, the handoff procedure is not sensitive to link transfer time. The second and third generation Code Division Multiple Access (CDMA)-based mobile communication systems fall in this category.

CHAPTER 3

CALL ADMISSION CONTROL

3.1 INTRODUCTION

CAC is a fundamental mechanism used for QoS provisioning in a network. Call admission control is a key element in the provision of guaranteed quality of service in wireless networks. Admission control decision is made using a traffic descriptor that specifies traffic characteristics and QOS requirements. The main traffic characteristics are

i. Sustained cell rate

ii. Maximum burst size

iii. Peak cell rate

The design of call admission control algorithms for mobile cellular networks is especially challenging given the limited and highly variable resources, and the mobility of users encountered in such networks. QoS provisioning in wireless networks is a challenging problem. CAC is a fundamental mechanism used for QoS provisioning in a network. It restricts the access to the network based on resource availability in order to prevent network congestion and service degradation for already supported users. This is due to the scarcity of wireless resources i.e. radios channels, and the mobility of users. It restricts the access to the network based on resource availability in order to prevent network congestion and service degradation for already supported users.

In general there are two categories of CAC schemes in cellular networks. They are

i) Deterministic CAC

ii) Stochastic CAC

3.1.1 DETERMINISTIC CAC

QoS parameters are guaranteed with 100% confidence. Typically, these schemes require extensive knowledge of the system parameters such as user mobility which is not practical, or sacrifice the scarce radio resources to satisfy the Deterministic QoS bounds.

3.1.2 STOCHASTIC CAC

QoS parameters are guaranteed with some probabilistic confidence. Its classification is given in Fig 3.1. In that reservation scheme is used.

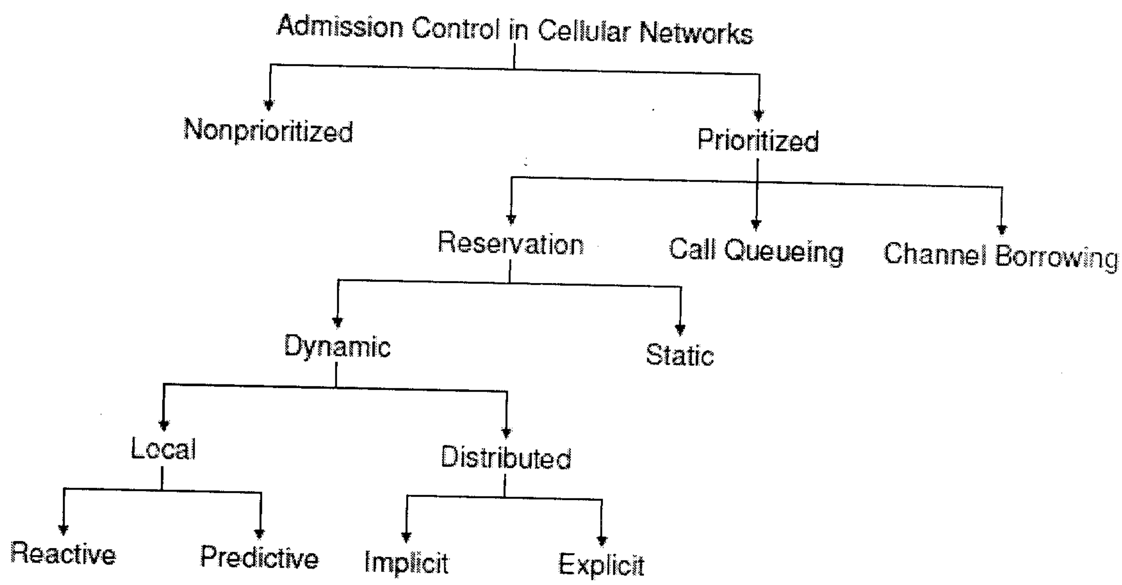

Fig 3.1 Stochastic Call Admission Control Schemes in Cellular Networks

3.1.2.1 RESERVATION SCHEMES

The notion of guard channels was introduced in the mid 80s as a call admission control mechanism to give priority to handoff calls over new calls. In this policy, a set of channels called the guard channels are permanently reserved for handoff calls. Hong and Rappport showed that this scheme reduces handoff dropping probability significantly compared to the no prioritized case. They found that call dropping probability p_d decreases by a significantly larger order of magnitude compared to the increase of call blocking probability p_b when more priority is given to handoff calls by increasing the number of handoff channels.

Consider a cellular network with C channels in a given cell. The Guard Channel scheme (GC) reserves a subset of these channels, say $C - T$, for handoff calls. Whenever the channel occupancy exceeds a certain threshold T, GC rejects new calls until the channel occupancy goes below the threshold. Assume that the arrival process of new and handoff calls is Poisson within adjusted dynamically based on the observed traffic load and dropping rate in a control time window. If the observed dropping rate is above the guaranteed p_d then the number of reserved channels is increased. On the other hand, if the current dropping rate is far below the target p_d then the number of reserved channels is decreased.

CHAPTER 4

FRACTIONAL GUARD CHANNEL

4.1 FRACTIONAL GUARD CHANNEL

A different variation of the basic GC scheme is known as Fractional Guard Channel (FGC). Whenever the channel occupancy exceeds the threshold T, the GC policy is to reject new calls until the channel occupancy goes below the threshold. In the fractional GC policy, new calls are accepted with a certain probability that depends on the current channel occupancy. A randomization parameter which determines the probability of acceptance of a new call is used. Both GC and FGC policies accept handoff calls as long as there are some free channels. One advantage of FGC over GC is that it distributes the newly accepted calls evenly over time which leads to a more stable control .The behavior of FGC in a cell with C channels is depicted in diagram. Note that in state n, the acceptance ratio is an. Using balance equations, the steady-state probability of having n channels busy is given by equation 4.1

$$P_n = \frac{\prod_{i=0}^{n-1} (\upsilon + a_i \lambda)}{(\mu + \eta)^n} P_o, 1 \le n \le C \qquad (4.1)$$

Where

$$P_0 = \left[1 + \sum_{n=1}^{C} \frac{\prod_{i=0}^{n-1} (\upsilon + a_i \lambda)}{(\mu + \eta)^n} \right]^{-1}$$

It has been found that due to advance reservation in reservation schemes the efficiency of cellular systems has an upper bound even if no constraint is specified on the call blocking probability. This upper bound is related to call and mobility characteristics through the mean number of handoffs per call. Moreover, the achievable efficiency decreases with decreasing cell size and with increasing call holding time.

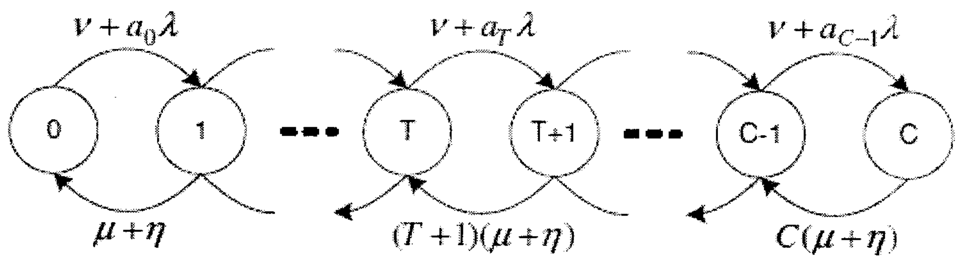

Fig4.1.State transition diagram of the fractional guard channel scheme

There are two types of fractional guard channel. They are,

1) Limited Fractional Guard Channel scheme (LFGC)

2) Uniform Fractional Guard Channel scheme (UFGC)

4.1.1 LFGC

It is one of the type of fractional guard scheme.LFGC finely controls communication service quality by effectively varying the average number of reserved channels by a fraction of one. It is better than UFGC and GC for any mobile condition.

4.1.2 UFGC

UFGC accepts new calls within admission probability independent of channel occupancy.

CHAPTER 5

QUALITY OF SERVICE

5.1 QoS PROVISIONING

QoS is a technology which uses various mechanisms to decrease the negative effects of congestion in packet-switched networks. Even though the gain obtained through statistical multiplexing is more than enough, the task of providing QoS is a difficult task. Some of the problems that QoS faces are mobility, handoff, and limited bandwidth. There are different types of QoS they are

1) Packet – Level QoS

At this level, users need guarantees on

1. Packet dropping probability
2. Maximum packet delay
3. Maximum jitter

2) Call – Level QOS

In the case of wireless networks, a new call blocking probability and the handoff dropping probability be small. Handoff occurs when a user, in the middle of a call, moves to an adjacent all.

3) Class – Level QOS

Some of the class level QOS are

1. Blocking
2. Call blocking portability of all classes

In future developments, call admission schemes like require to meet class level.

CHAPTER 6

WiMAX

WiMAX is defined as Worldwide Interoperability for Microwave Access by the WiMAX Forum, formed in June 2001 to promote conformance and interoperability of the IEEE 802.16 standard, officially known as Wireless MAN. The Forum describes WiMAX as "a standards-based technology enabling the delivery of last mile wireless broadband access as an alternative to cable and DSL".

The 802.16 specification applies across a wide swath of the RF spectrum. However, specification is not the same as permission to use. There is no uniform global licensed spectrum for WiMAX. In the US, the biggest segment available is around 2.5 GHz, and is already assigned, primarily to Sprint Nextel and Clear wire. Elsewhere in the world, the most likely bands used will be around 3.5 GHz, 2.3/2.5 GHz, or 5 GHz, with 2.3/2.5 GHz probably being most important in Asia. In addition, several companies have announced plans to utilize the WiMAX standard in the 1.7/2.1 GHz spectrum band recently auctioned by the FCC, for deployment of "Advanced Wireless Services"(AWS). There is some prospect in the U. S. that some of a 700 MHz band might be made available for WiMAX use, but it is currently assigned to analog TV and awaits the complete rollout of digital TV before it can become available, likely by 2009. In any case, there will be other uses suggested for that spectrum if and when it actually becomes open.It seems likely that there will be several variants of 802.16, depending on local regulatory conditions and thus on which spectrum is used, even if everything but the underlying radio frequencies is the same. WiMAX equipment will not, therefore, be as portable as it might have been - perhaps even less so than WiFi, whose assigned channels in unlicensed spectrum varies little from jurisdiction to jurisdiction. The actual radio bandwidth of spectrum allocations is also likely to vary. Typical allocations are likely to provide channels of 5 MHz or 7 MHz. In principle the larger the bandwidth allocation of the spectrum, the higher the bandwidth that WiMAX can support for user traffic.

6.1 STANDARDS

The current 802.16 standard is IEEE Std 802.16e-2005, approved in December 2005. It followed on from IEEE Std 802.16-2004, which replaced IEEE Standards 802.16-2001, 802.16c-2002, and 802.16a-2003.IEEE Std 802.16-2004 (802.16d) addresses only fixed systems. 802.16e adds mobility components to the standard.

6.1.1 IEEE 802.16e

IEEE 802.16-2005 (formerly named, but still best known as, 802.16e or Mobile WiMAX) provides an improvement on the modulation schemes stipulated in the original (fixed) WiMAX standard. It allows for fixed wireless and mobile Non Line of Sight (NLOS) applications primarily by enhancing the Orthogonal Frequency Division Multiple Access (OFDMA) .Scalable Orthogonal Frequency Division Multiple Access (SOFDMA) improve upon OFDM256 for Non Line of Sight applications by:

* Improving NLOS coverage by utilizing advanced antenna diversity schemes, and hybrid-Automatic Retransmission Request.
* Increasing system gain by use of denser sub-channelization, thereby improving indoor penetration.
* Introducing high-performance coding techniques such as Turbo Coding and Low-Density Parity Check (LDPC), enhancing security and NLOS performance.
* Introducing downlink sub-channelization, allowing administrators to trade coverage for capacity or vice versa.
* Improving coverage by introducing Adaptive Antenna Systems and Multiple Input Multiple Output technology.
* Eliminating channel bandwidth dependencies on sub-carrier spacing, allowing for equal performance under any RF channel spacing (1.25-14 MHz).
* Enhanced Fast Fourier transform algorithm can tolerate larger delays spreads, increasing resistance to multipath interference.

On the other hand, 802.16-2004 (fixed WiMAX) offers the benefit of available commercial products and implementations optimized for fixed access. Fixed WiMAX is popular standard among alternative service providers and operators in developing areas due to its low cost of deployment and advanced performance in a fixed environment. Fixed WiMAX is also seen as a potential standard for backhaul of wireless base stations such as cellular, WiFi of even mobile WiMAX. SOFDMA and OFDMA256 are not compatible so most equipment will have to be replaced. However, some manufacturers are planning to provide a migration path for older equipment to SOFDMA compatibility which would ease the transition for those networks which have already made the OFDMA256 investment.

6.1.2 QoS

QoS in 802.16e is supported by allocating each connection between the Security Sub layer(SS) and the Base Station(BS) (called a *service flow* in 802.16 terminology) to a specific *QoS class*. In 802.16e, there are 5 QoS classes.

Table6.1. 802.16e-2005 QoS Classes

802.16e-2005 QoS classes			
Service	**Abbrev**	**Definition**	**Typical Applications**
Unsolicited Grant Service	UGS	Real-time data streams comprising fixed-size data packets issued at periodic intervals	T1/E1 transport
Extended Real-time Polling Service	ErtPS	Real-time service flows that generate variable-sized data packets on a periodic basis	VoIP
Real-time Polling Service	rtPS	Real-time data streams comprising variable-sized data packets that are issued at periodic intervals	MPEG Video
Non-real-time Polling Service	nrtPS	Delay-tolerant data streams comprising variable-sized data packets for which a minimum data rate is required	FTP with guaranteed minimum throughput
Best Effort	BE	Data streams for which no minimum service level is required and therefore may be handled on a space-available basis	HTTP

CHAPTER 7

QUEUING ALGORITHMS

Queue management is needed for operation during periods of congestion, when network devices cannot cope with the task of transmitting packets to the output interface at the same rate at which they arrive. If this overload is caused by insufficient performance of the processor unit of the network device, then the input queue of the respective input interface is used for temporary storage of unprocessed packets. If the request is differentiated into several classes, there might be several input queues in the same interface. If the overload is caused by insufficient bandwidth of the output interface, then packets are temporarily stored in the output queue or queues of this interface.

7.1 FIFO QUEUE

The essence of traditional **FIFO algorithm** lies in that if overload occurs, all packets are placed into a single common queue and retrieved from it according to the order in which they arrived — that is, first in, first out. In all packet-switching devices, the FIFO algorithm is used by default. Its advantages include the ease of implementation and the lack of need to configure it. However, it has rather a serious drawback — the impossibility of differentiated processing of the packets belonging to different flows. All packets are placed into the common queue and have the same priority. This relates to packets of delay-sensitive voice traffic as well as to packets of backup traffic, which is insensitive to delays but rather intensive and whose long bursts are capable of delaying voice traffic for quite a long time.

7.2 PRIORITY QUEUING

Priority queuing algorithms are popular in many areas of computing — for example, in multitasking operating systems, where certain applications must have priority over others. These algorithms are also used for priority queuing, when some classes of traffic must have priority over others. The priority-queuing mechanism is based on dividing all network traffic into a small number of classes and assigning some numeric characteristic, known as **priority**, to each class.

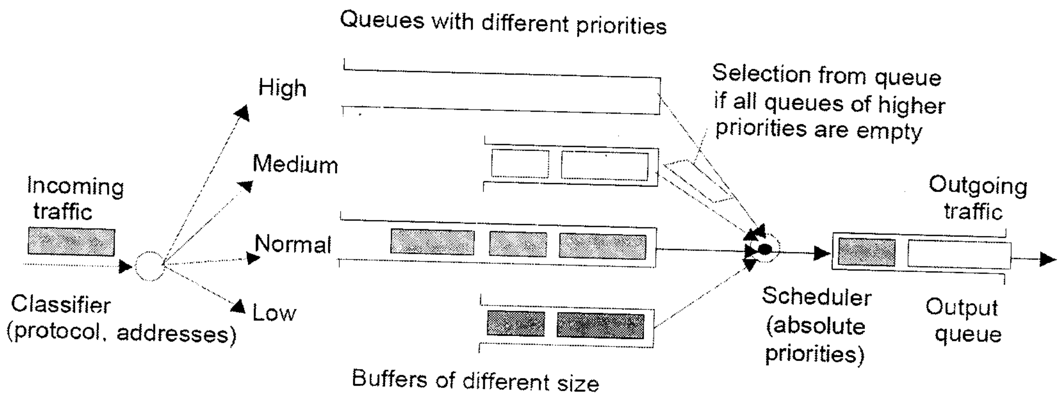

Fig 7.1 Priority Queues

Usually, buffers of the same size are assigned to all priority queues by default. However, many devices allow administrators to individually configure the buffer size for each queue. Buffer size determines the maximum number of packets that can be stored in the queue of a specific priority. If the packet arrives when the buffer is full, it is discarded. As a rule, buffer size is defined to be on the safe side when handling queues of average length. However, it is rather difficult to evaluate this value, since it depends on the network load. Because of this, to achieve this goal, it is necessary to continuously monitor network operation for long periods of time. In general, the higher the importance of the traffic for the user, the higher its rate and bursts, and the larger the buffer size required for it. In the example presented in Fig 7.1, large buffers are assigned for traffic of high and normal priorities; for other two classes, smaller buffers were allocated. For high-priority traffic, the motives of this solution are obvious. As for traffic with normal priority, it is expected to be rather intense and bursty. Priority queuing ensures high QoS for the packets from the queue with the highest priority. If the average rate at which these packets arrive to the device does not exceed the bandwidth of the output interface (and the processor performance of the device), then packets with the highest priority always get the bandwidth that they require. The delay level for such packets is also at a minimum. However, it is not 0 and depends mainly on characteristics of the flow of such packets. The higher the flow burst and information rate, the higher the probability of generation of the queue formed by packets of this high-priority traffic. Traffic of all other priority classes is nearly transparent for the packets with high priority. Since such situations are possible in which the packet with higher priority has to wait until the device accomplishes processing of the packet with lower priority. This happens when the arrival of the packet with higher priority coincides with when the device started to process the packet of lower priority.

As relates to other priority classes, the QoS provided to them will be lower than for the packets with the highest priority. Note that the degree of this quality decrease is difficult to predict. This decrease can be very significant if the traffic of highest priority is intense. If the utilization coefficient of the device, determined only by the traffic of highest priority, rises close to 1 for specific time instances, then the traffic of all lower priority classes becomes practically frozen for such intervals. Because of this, priority queuing is used when there is one class of real-time traffic in the network but its intensity is not high; consequently, serving this class does not impair serving all other traffic. For example, voice traffic is rather sensitive to delays, though its rate rarely exceeds 8 –64 Kbps. Therefore, if this traffic is assigned the highest priority, service provided to all the other classes of traffic would not be significantly infringed upon. However, other situations are possible. For example, consider a network where it is necessary to transmit video traffic, which also requires priority serving but has a significantly higher rate. For such cases, special queuing algorithms are developed, providing some guarantees that low-priority traffic will also be served even when the rate of the higher-priority classes significantly rises.

To combine several flows into an aggregate, it is necessary to ensure that they impose the same QoS requirements and have common input and output points to and from the network.

7.3 WEIGHTED QUEUING

The **Weighted Queuing Algorithm** is developed to provide a certain minimum of bandwidth to all classes of traffic, or at least to guarantee the observance of some requirements to delays. Class weight is the percentage of the total bandwidth of the resource (e.g., processor or output interface of a switch) that is guaranteed to this class of traffic. Like priority queuing, weighted queuing requires the division of traffic into several classes. For each class, a separate packet queue is created. However, in weighted queuing, each queue is assigned the percentage of the resource bandwidth guaranteed to this class under conditions of resource overload rather than a specific priority. For input flow, the role of resource is played by the processor, and for the output flow (after accomplishing the switching), this role is played by the output interface. As a result, each traffic class will get its guaranteed minimum bandwidth. In most cases, this result is more desirable than the suppressing of the low-priority traffic classes by classes of higher priorities.

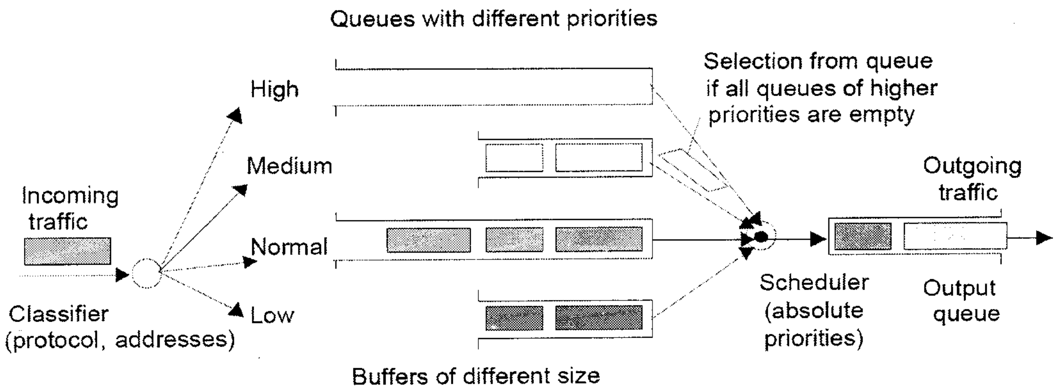

Fig 7.2 Weighted Queues

There exists another type of weighted queuing — **Weighted Fair Queuing** (WFQ). In this case, the resource bandwidth is divided between all flows equally (i.e., fairly).

It is observed that, Weighted queuing ensures the required relations between the rates of traffic from different queues only during periods of congestion, when each queue is constantly filled. If any queue is empty (which means that for the traffic of this class, the current period is not the period of congestion), then this queue is omitted during the current round-robin lookup, and the time allocated for serving this queue is distributed between all other queues according to their weights. Thus, during specific time periods, traffic of a specific class may have a higher rate than the appropriate percentage of the output interface bandwidth.

7.4 HYBRID ALGORITHMS OF QUEUING

Each of the two approaches described previously has specific advantages and drawbacks. Priority queuing ensures minimum delay levels, at least for the traffic of the highest priority. This algorithm serves all traffic from the queue of the highest priority, no matter how high its rate might be, and does not provide any guarantees in relation to the average bandwidth for the traffic from queues of lower priorities. Weighted queuing guarantees average traffic rate but does not provide any guarantee with regard to delays.

There are also hybrid algorithms of queuing that try to find a compromise between these methods. The most popular type of such an algorithm uses one priority queue and serves all other queues according to the weighted algorithm. Usually, priority queues are used for real-time traffic, and other queues are used for elastic traffic of several classes. Each class of elastic traffic gets some guaranteed minimum of the bandwidth during congestion periods. This minimum is calculated as a percentage of the bandwidth remaining after serving the priority traffic. Obviously, it is necessary to limit the priority traffic somehow to prevent it from consuming the entire bandwidth of the resource.

CHAPTER 8

PROBLEMATIC

The 802.16 MAC protocol is oriented to connection. Security Sub layer (SS) shall establish a connection to the Base Station (BS) to transmit data. The sending process data is presented in Figure 8.1. It begins with a connection establishment phase which includes the QoS negotiation parameters and control admission. If sufficient resources are available, the connection is accepted. The SS shall send a request for bandwidth allocation. This step requires an effective scheduling of flows in both parties BS and SS. Blocks dotted traces show the parts defined by the standard, they are left to constructors in order to be implemented. Several researches focused on defining optimal mechanisms for those parts not covered by the standard.

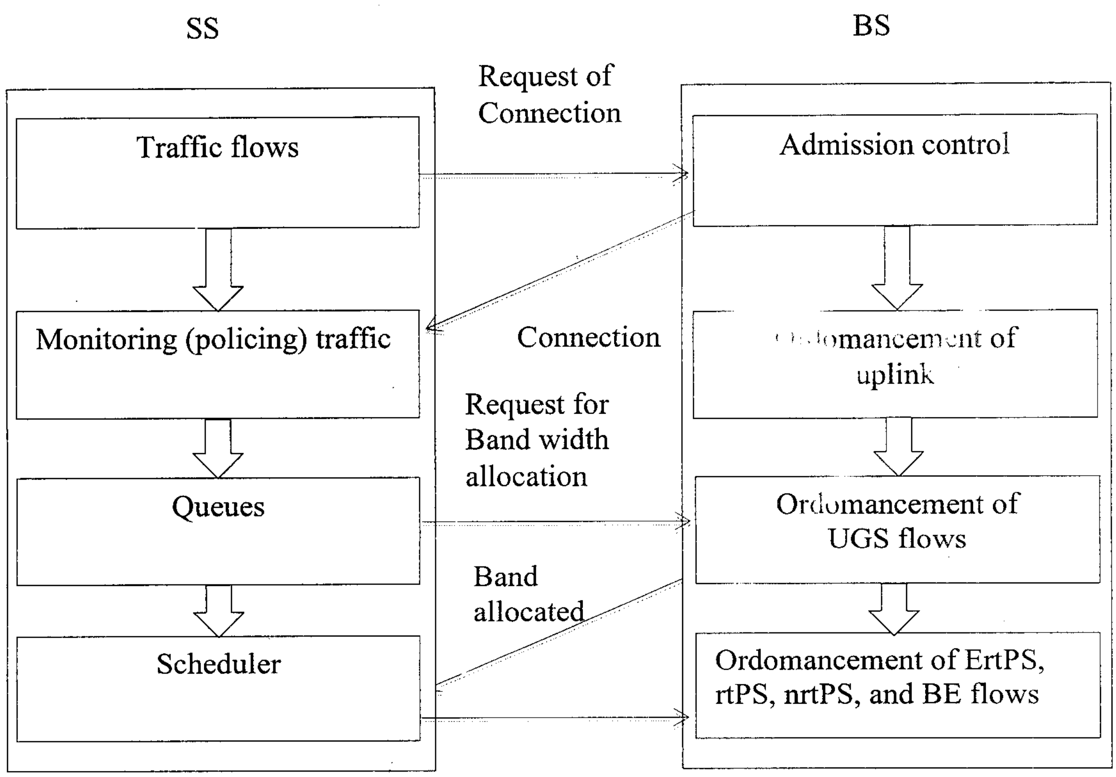

Fig 8.1 Data Sending Process and Problematic

Several researches focused on defining optimal mechanisms for those parts not covered by the standard. In literature, the algorithms proposed to allocate bandwidth to different classes of service using a hierarchical architecture. Indeed, the bandwidth assigned to connections beginning with the higher priority class (e.g. UGS) and then moving on to the lower priority classes. Then, the connections of each class are arranged according to different scheduling mechanisms. RtPS connections are provided by EDF discipline, connections nrtPS follow Weighted Fair Queuing (WFQ) discipline and BE connections are provided by the First in First Out (FIFO) discipline.

In following, a CAC model is introduced for improving the QoS in 802.16e networks. The CAC mechanism in this network that can satisfy both of the following purposes:

- The QoS satisfaction constraints of five services classes: UGS, ErtPS, rtPS, nrtPS and BE.
- And call management HO to be higher priority than new calls.

CHAPTER 9

PROPOSED MODEL

9.1 CAC PROPOSED MODEL

In the CAC model of Fig 9.1, a single cell in a homogeneous network consisting of several cells. The cell has a constant amount of bandwidth B divided in two proportions, first for the uplink and the second for the down link. The cell serves heterogeneous users who require different services (e.g. voice, data, video, etc.) in the form of new (N) or handover (H) calls coming from the adjacent cells.

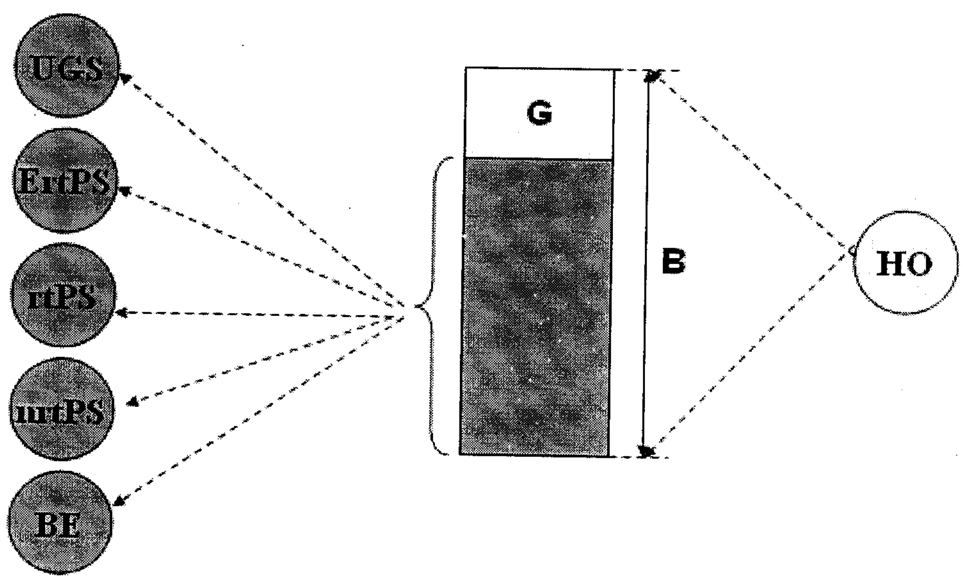

Fig 9.1. CAC Proposed Model within Wireless Multiservice Network

According to the requirements of each service class in terms of bandwidth and delay, it is intuitive to consider the following priority system as indicated by equation (9.1)

$$P(UGS)>P(ErtPS)>P(rtPS)>P(nrtPS)>P(BE) \tag{9.1}$$

With P(x) denotes the priority of the x class. Thus, the bandwidth is allocated to connections with highest priority class before the lowest priority class. The architecture (2) includes 5 priorities, and if we want to take into account HO calls, the architecture (3) includes 10 priorities as indicated by equation (9.2)

$$P(UGS,H)>P(ErtPS,H)>P(rtPS,H)>P(nrtPS,H)>P(BE,H)>P(UGS,N)>P(ErtPS,N)>P(rtPS,N)>P(nrtPS,N)>P(BE,N) \quad (9.2)$$

However, this architecture presents an enormous complexity in terms of call management because each defining priority queue is served according to a scheduling algorithm suitable for the service class involved. In the model and to reduce this complexity, the service classes are grouped according to their time requirements i.e. in real-time RT connections and non-real time NRT connections can be used. The Architecture (4) will include the four priorities as specified by equation (9.3)

$$P\ (RT,\ H)>P\ (NRT,\ H)>P\ (RT,\ N)>P\ (NRT,\ N) \quad (9.3)$$

- P(RT, H): priority assigned to real-time HO calls(includes HO calls belonging to UGS, rtPS and ErtPS).
- P(NRC, H) : priority assigned to non-real time HO calls (includes HO calls belonging to nrtPS and BE).
- P(RT, N): priority assigned to the new real-time connections (includes new connections belonging to UGS, rtPS and ErtPS).
- P(NRT, N): priority assigned to the new non-real time connections (includes new connections belonging to nrtPS and BE).

Since HO calls have higher priority than new connections, in the model, a bandwidth proportion will be restricted to HO calls. Real-time calls are characterized by a maximum tolerance to delay. Upon arrival of each real time call, to calculate a deadline after which if the call is not served, it will be rejected. As for non-real time calls, they will be served according to their order of arrival.

8.2 FLOWCHART

The treatment of each call type is described by the following procedures:

1) A real-time new call treatment

A real time new call is characterized by (Fig 9.2):

- b: bandwidth requested
- d_{max}: maximum delay tolerance
- d: time that the call spent in the queue until the instant t.

The treatment of this type of call is as follows:

- Call arrival:
  - If $U+b \leq B-G$; the call is admitted: $N_a=N_a+1$ and $U=U+b$ (requested bandwidth reservation)
  - If not, d=d+frame duration and as long as $d \leq d_{max}$
    - If $U+b \leq B-G$; the call is admitted: $N_a=N_a+1$ and $U=U+b$
    - If not d=d+frame duration
  - If $d>d_{max}$, $N_r=N_r+1$

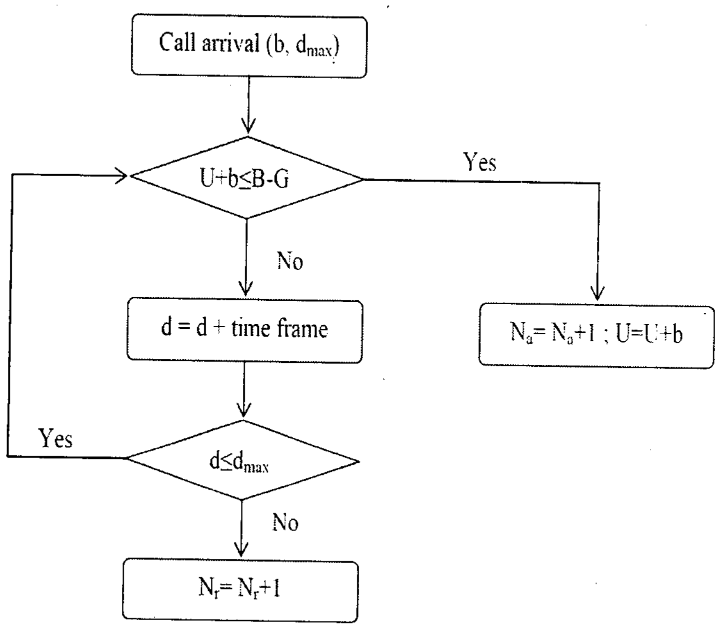

Fig 9.2 New Real-Time Call Treatment

2) A non-real time new call treatment

A non-real time new call is characterized by (Fig 9.3):

- b: Bandwidth requested.
- b_{min}: Reserved minimum rate.
- d: Time that the call spent in the queue until the instant t.

Other parameter is d_{max}: maximum delay tolerance during which the call will wait in the queue. It is necessary to be specified in order not to remain indefinitely waiting in the queue.

- Call arrival:
  - If $U+b \leq B-G$; the call is admitted: $N_a=N_a+1$ and $U=U+b$
  - If not, if $U+b_{min} \leq B-G$; the call is admitted: $N_a=N_a+1$ and $U=U+b_{min}$
  - If not, d=d+frame duration and as long as $d \leq d_{max}$.
    - If $U+b \leq B-G$; the call is admitted: $N_a=N_a+1$ and $U = U+b$
    - If not, if $U+b_{min} \leq B-G$; the call is admitted $N_a=N_a+1$and $U=U+b_{min}$
    - If not d=d+duration of frame
  - If $d>d_{max}$, $N_r=N_r+1$

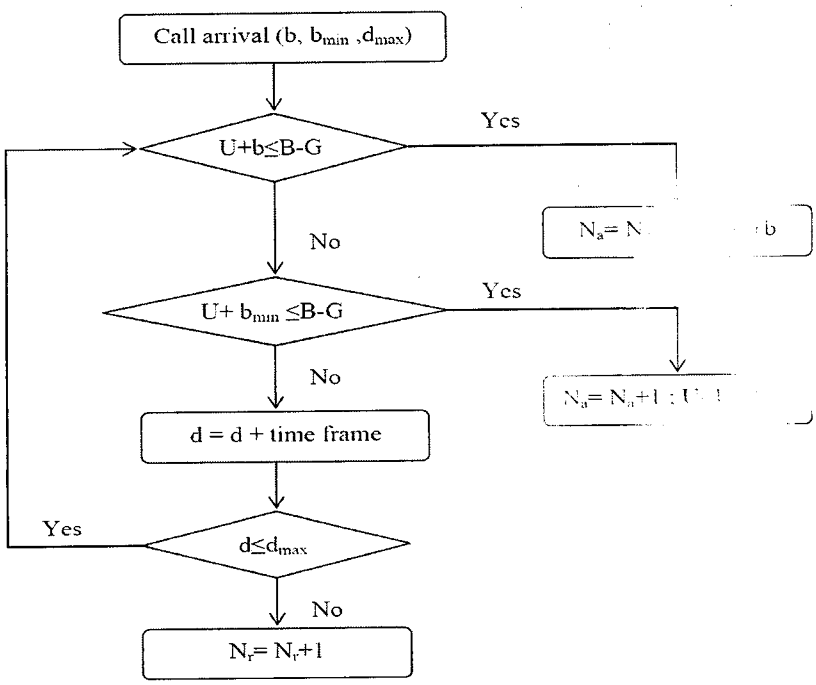

Fig 9.3 Non-Real Time New Call Treatment

3) Real-time HO call Treatment

A real-time HO call is characterized by (Fig 9.4):

- b_{ho} : Bandwidth required
- d_{max}: Maximum delay tolerance
- d: Time that the call spent in the queue until the instant t.

The treatment of this type of call is as follows:

- Call arrival:
  - If $U+b_{bo}\leq B$, the call is admitted:$H_a=H_a+1$ and $U=U+b_{ho}$
    - If not, d=d+frame duration and as long as $d\leq d_{max}$.
    - If $U+ b_{ho}\leq B$, the call is admitted: $H_a=H_a+1$ and
    - $U=U+b_{ho}$

This procedure is presented by the fig 9.4

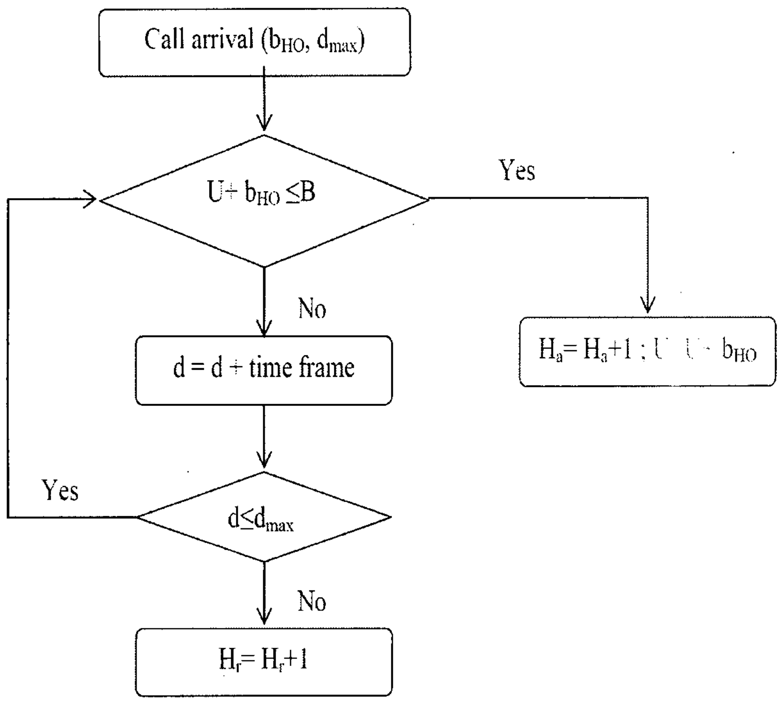

Fig 9.4 Real-Time HO Call Treatment

4) Non-real time HO call treatment

A non-real-time HO call is characterized by (Fig 9.5):

- b_{ho} : Maximum bandwidth permissible
- $b_{ho,min}$: Reserved minimum rate
- d: Time that the call spent in the queue until the instant t

Other parameter is d_{max}: Maximum delay tolerance during which the call will wait in the queue. It is necessary to be specified in order not to remain indefinitely waiting in the queue. The treatment of this type of call is as follows:

- Call arrival:
  - If $U+b_{ho} \leq B$, the call is admitted: $H_a=H_a+1$ and $U=U+b_{ho}$
  - If not, if $U+b_{ho,min} \leq B$, the call is admitted: $H_a=H_a+1$and $U=U+b_{ho,min}$
  - If not, d=d+frame duration and as long as $d \leq d_{max}$,
    - If $U+b_{ho} \leq B$, the call is admitted: $H_a=H_a+1$ and $U=U+ b_{ho}$
    - If not, if $U+b_{ho,min} \leq B$, the call is accepted: $H_a=H_a+1$ and $U=U+ b_{ho,min}$
    - if not d=d+frame duration
  - If $d>d_{max}$, $H_r=H_r+1$

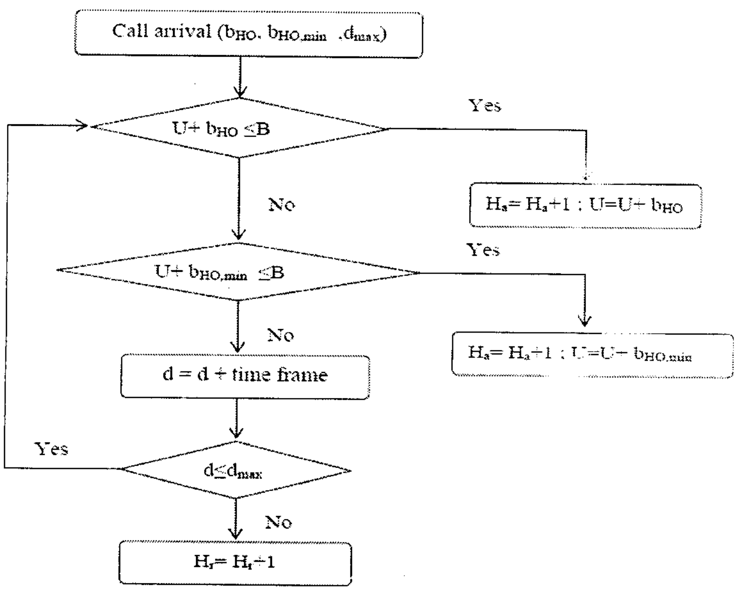

Fig 9.5 Non-Real Time HO Call Treatment

9.3 ADAPTIVE CAC ALGORITHM

The number of guard channels has been considered to be one of the key design parameters which have tremendous effects on the performance of wireless networks. In the approach, the number of guard channels of a wireless network at each base station will be determined through optimizing certain performance goal with service quality constraints. When a base station experiences high handoff call blocking rate, the number of guard channels is increased until the handoff blocking rate drops to below its threshold. When a base station does not get to use a significant portion of the guard channels over a period of time, the number of guard channels can be gradually decreased until most of the guard channels are used frequently. By doing this, the handoff blocking rate get close to its threshold. However, it is necessary to keep below the threshold all the time. Algorithm for determining adaptively the number of guard channel is given by

If a handoff call is dropped and

$D_H / H \geq \alpha_\mu T_H$ then

$C_H = \min\{ C_H + 1,\ C_{\max} \}$

Where α_μ is a threshold chosen as e.g. 0.9

If $D_H / H \geq \alpha_d T_H$ for N consecutive handoff calls, then

$C_H = \min\{ C_H - 1,\ C_{\max} \}$ where α_d is

Another threshold chosen as e.g. 0.6 and N is an integer chosen as e.g. 10

This algorithm has the following important features.

- It adjusts the number of guard channels C_H adaptively according to the dropping rate of handoff calls in time period τ and.
- It tries to make sure that the handoff call blocking rate is below the given threshold T_H and it also tries to reduce the new call blocking rate by decrementing C_H when it is observed to be more than needed.

The present algorithm will only increase the number of guard channels when a handoff call is dropped under the condition that $D_H / H \geq \alpha_\mu T_H$ and we will only decrease the number of guard channels after a number of consecutive handoff calls under the condition $D_H / H \geq \alpha_d T_H$ that are usually chosen to be less than 1. By choosing the algorithm will most likely keep the handoff call blocking rate below its given threshold. An algorithm in a similar spirit has been applied in for adaptive bandwidth reservation where the increase and the decrease in the reserved bandwidth are both done as soon as the threshold conditions for the monitored dropping probabilities are satisfied. The present algorithm will wait for $D_H / H \geq \alpha_d T_H$ consecutive handoff calls under the condition that before increasing the number of guard channels, where N is larger than 1. Same idea can be applied to the algorithm in to improve its performance. In particular, by waiting for N consecutive handoff calls under certain condition before increasing the number of guard channels will keep the system performance from oscillating.

One design parameter left untouched thus far is the time period τ which indicates the total time for updating all the measurements used in the algorithm. It must be long enough in order to have a meaningful evaluation and decision making. If τ is too small, the system will response to changes too often and certain measurements (e.g., call blocking rates) may not be accurate, which may result in oscillations in system performance. On the other hand, if is too large, the system may not response fast enough and thus may not perform to its best possible. Generally speaking, can be chose τ to be proportional to $1/T_H$ (T_H is the threshold for handoff blocking probability) or proportional to the inverse of any other QoS parameters.

CHAPTER 10

CAC MODEL IMPLEMENTATION

10.1 SIMULATION ENVIRONMENT

In developing the algorithm, Matlab version 7.9 is used .This software is chosen because it has a basis of functions already implanted and necessary for modeling different traffic model laws and modeling queues. Indeed, functions such as:poissrnd or exprnd were very useful for the generation of traffic models based on random processes (poissoniens, exponential,etc.).Other functions like plot were also very useful for the results representation in curves form.

10.2 SIMULATION PARAMETERS

- In the simulation design, the following assumptions and parameters are used.
- There is a single cell in a homogeneous network consisting of several cells.
- Bandwidth available in the BS: 10Mbps divided equally between the uplink and the downlink.
- Frame duration: 5ms.
- The arrival process of new and HO calls are poissonni[illegible]pective parameters λ and μ with $\lambda = \mu *5$.
- The observation duration is equal to 200frames.

Two types of traffic are considered .They are.

Video traffic using MPEG-4 coding with:

- An average flow of 180Kbps.
- The inter-arrival packets duration is constant and equal to 40ms.
- The session duration is equal to 5s.
- The packet average size is equal to 900 bytes.
- The maximum waiting time for a new connection is equal to 40ms.

- FTP traffic: the flow varies between 16 and 256Kbps.
- The reserved minimum flow is equal to 5Kbps.

CHAPTER 11
RESULTS

1) The guard band Influence on the new calls blocking probability

The behavior of the BS for the arrival rates of new calls λ have been simulated. These rates vary from 25 to 45calls/s (with arrival rate of HO calls: $\mu = \lambda/5$) in both cases: G=0 and G=B/10 with B=5Mbps. The calculation of new calls blocking probability gave the two following curves (see Fig11.1): According to Fig 11.1, we see that the increasing of λ (and therefore μ) leads the increase of new connections blocking probability. For λ varies between 25 and 35calls/s, we have an acceptable blocking rate (between 0 and 4%), beyond these values, the blockage rate becomes too high, it exceeds 10% when λ exceeds 45calls/s. Comparing these two cases where G=0 and G=B/10, The first case where G=0 offers the lowest blocking rate, this is well justified since in this case, the new and HO calls are treated similarly: as long as the bandwidth is available, they are admitted. In the second case where G=B/10=500kbps, a part of the bandwidth is reserved exclusively for HO calls therefore the new calls have less chance to be admitted which causes the increased of calls blocking probability.

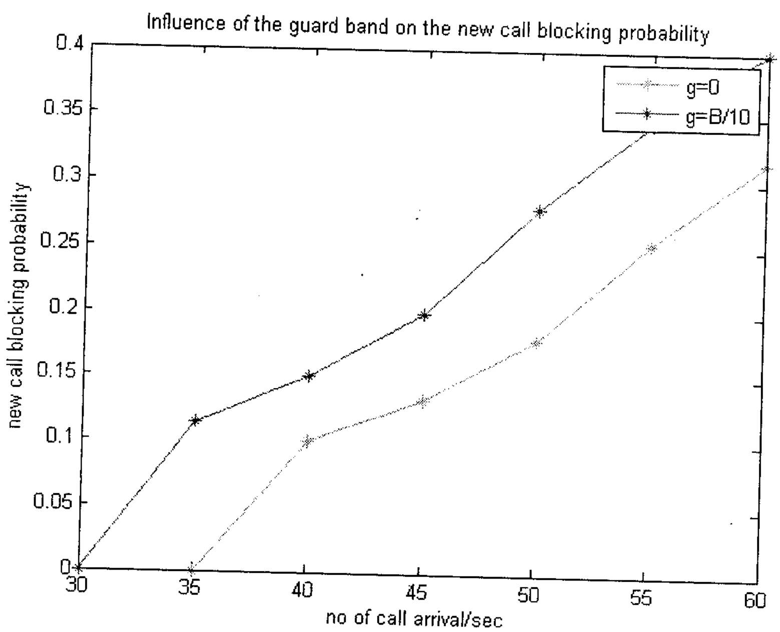

Fig 11.1 Influence of the Guard Band on the New Call Blocking Probability

2) Influence of the guard band on the HO calls dropping probability

In order to study the influence of the guard band on the HO calls dropping probability, the BS behavior have been simulated with the same parameters used in the precedent paragraph. Simulation results are presented in Fig 11.2.From the Fig 11.2 admission control policy with guard band offer a HO calls dropping rate much lower than without guard band that varies from 2 to 20times for an arrival rate λ varying between 30 and 55calls/s. Indeed, till 35calls/s, all HO calls are admitted but over 45calls/s, P_{ch} becomes too high and exceeds.

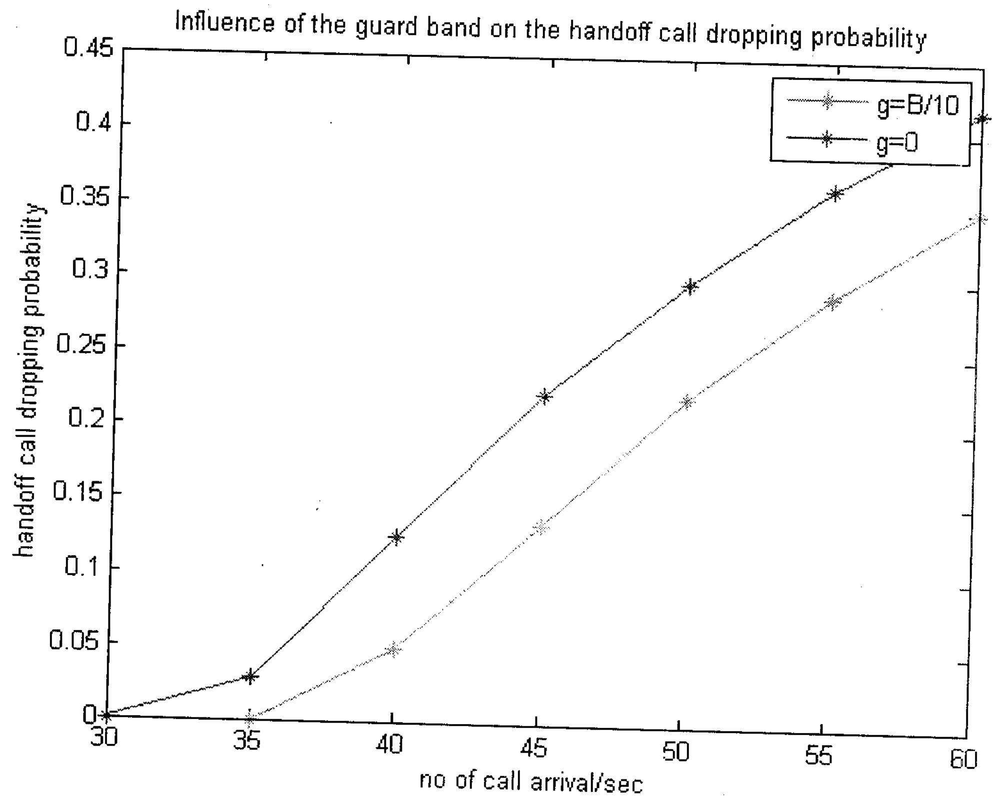

Fig 11.2 Influence of the Guard Band on the HO Calls Dropping Probability

3) New Handoff call blocking rate

The results of Fig 11.3 show that the present algorithm can adapt to changes in traffic conditions such as changes in the call arrival rate and can achieve optimal performance in terms of guaranteeing handoff call blocking threshold and minimizing the new call blocking rate at the same time.

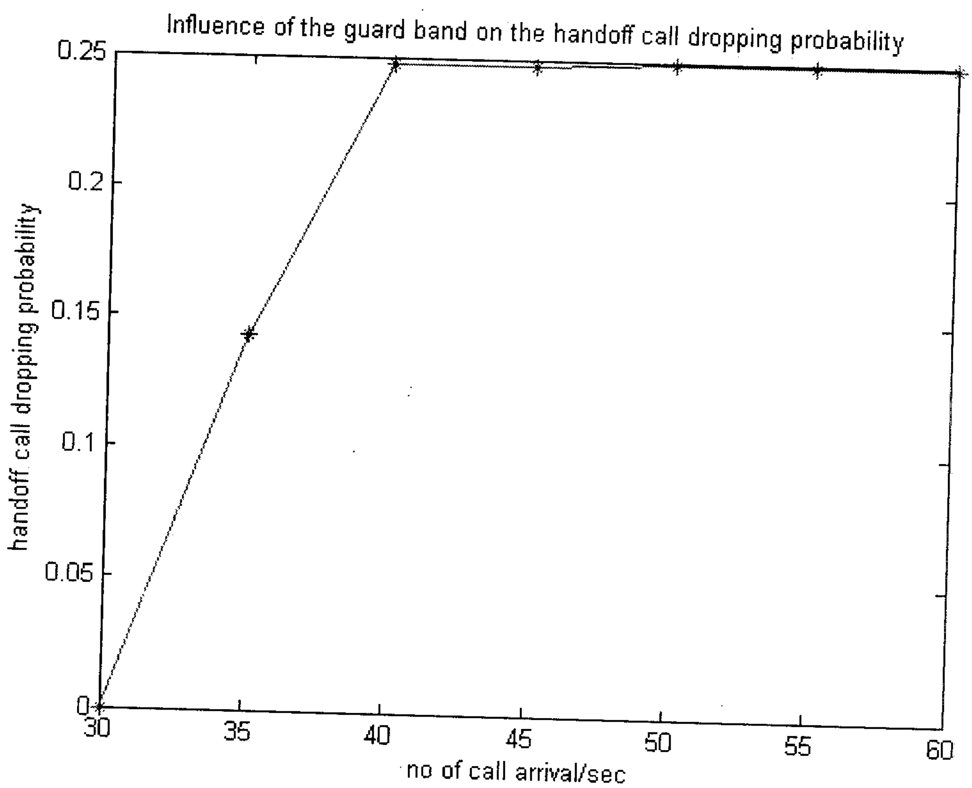

Fig 11.3 Handoff Call Blocking Rate

CHAPTER 13
CONCLUSION AND FUTURE SCOPE

In this project, an approach based on admission control mechanism that takes into account the HO management and different service classes defined by the 802.16e standard is presented . This approach is based on the guard band principle which reserves a bandwidth to meet and give more priority to HO calls. Compared with the admission control policy without guard band, the simulation results show that this policy provides better performance regarding the management of HO calls. Indeed, it offers a lower dropping probability and a lower average waiting time for real time HO calls. However this policy has a higher rate of blocking new calls.

FUTURE SCOPE

This proposed approach can be applied for the fourth generation networks. Its applicability to networks having different technologies can also be envisaged, if complemented with the necessary handover functionalities.

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CERTIFICATE

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This is to certify that Mr./~~Ms.~~ P. BHAGATHSINGH, ME (COMMUNICATION SYSTEMS) KUMARAGURU COLLEGE OF TECHNOLOGY, COIMBATORE. has ~~attended~~ / presented a paper titled THE CAC MODEL FOR WIRELESS MULTISERVICE NETWORK in the 3rd National Conference on **COMMUNICATION, INFORMATION AND TELEMATICS (CITEL 2011)** on 3rd & 4th March 2011, organized by the Department of Electronics and Communication Engineering, Kumaraguru College of Technology, Coimbatore.

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