CHAPTER 5

UTILITY BASED SCHEDULING AND CALL ADMISSION CONTROL FRAMEWORK

This chapter presents the design of call admission control algorithms which schedules the channels in real time and non-real time users based on the utility of the call request. In LTE systems, there are three main groups in scheduling algorithms which includes persistent scheduling algorithm group, dynamic and semi-persistent scheduling. In the persistent scheduling algorithm the persistent scheduler assignments to the UEs are predefined. So during DL period and when the eNB assigned resources to them, the UEs need to listen to a group of predefined resources. There is no necessity to specify UEs for every DL period in this kind of scheduling which seems to be a major advantage.

5.1 INTRODUCTION

To support real time and non real time service there are extensive amount of research are taken place in the area of scheduling in LTE networks. The scheduling in the LTE networks is classified into two broad categories those are static and dynamic resource allocation schemes. The static scheduling algorithms allocate entire subcarriers to the one used in the scheduling moment or entire resource is allocated to a user at the scheduled time. These algorithms are used in the multiple access techniques like TDMA and FDMA networks. In the dynamic algorithm the channel conditions are taken into consideration at the time of decision
making in the scheduling process. To manage the multi user diversity the Channel Quality Indicators (CQI) are used by the LTE to get back the quality of the channel link between the user terminal and eNB.

The delay-prioritized approach is used to ensure the Quality of Service in the delay sensitive applications. Based on the users, CQI value the users are allocated with suitable resource block from the available resource block. Here the UEs can find the assigned RBs and the kind of modulation and coding scheme used is identified. But in the dynamic scheduling scheme, the scheduling decisions are taken for every DL period. The CQI feedbacks from the UEs are considered for scheduling decisions and there are chances for DL to change to another period proposed by Piro et al (2010).

Types of scheduling algorithms in LTE include,

- Max-Prod Scheduling Algorithm
- Max_Sum Scheduling Algorithm
- Round Robin-Max_Sum Scheduling Algorithm
- Multi Groups- Max_Sum Scheduling Algorithm

The users can be scheduled in two dimensions namely time and frequency which are considered as the key feature of packet scheduling in LTE networks. For resource management, the aggregate available bandwidth is divided in to subcarriers of 15 kHz. A sub channel is formed with a bandwidth of 180 kHz by grouping twelve consecutive subcarriers. The main objective of the proposed work is to provide efficient ensured quality of service and improve the spectral efficiency trough the utility
based scheduling for each user. The channel quality plays the major role in the allocation of resources to the real time and non real time services; the network traffic is configured for realistic network.

5.2 LITERATURE SURVEY

Dimitrov et al. (2011) had discussed the mutual interference scheduling and inter-cell interference has on each other. It has been discussed that the particular service policy used by the scheduler is the basis for inter cell interference pattern. The impact of inter-cell interference on user performance at flow level is examined in this approach.

Dimitrova et al. (2010) had presented a performance comparison of two distinct scheduling schemes for LTE uplink (fair fixed assignment and fair work-conserving) taking into account both packet level characteristics and flow level dynamics due to the random user behavior. Giuseppe Piro et al. (2011) had proposed a novel two-level scheduling algorithm. At the upper level, discrete time linear control theory is the basis for this novel approach. A proportional fair scheduler is customized at the lower level. The performance and the complexity of the proposed scheme have been evaluated both theoretically and by using simulations.

Makara and Ventura (2011) had proposed a scheme that is optimized for offering improved quality of service for a diverse mix of traffic including real-time VBR traffic in the downlink of LTE networks. The application quality of service requirements can be satisfied and the overall system throughput can improve in this algorithm using multiuser diversity. David Vengerov (2005) stated that the average delay experienced by the real time packets in the network needs to be minimized and the users
in the sub channels which experience the best link quality are scheduled in order to imply higher data rates.

Elias Yaacoub et al (2011) had proposed a pricing-based power control scheme was in the presence of BS cooperation. The interference mitigation schemes were implemented in conjunction with a low complexity scheduling algorithm.

Capozzi et al (2012) had investigated the design of downlink packet scheduling mechanism for LTE cellular network. In this the radio resource management procedure are analyzed to improve system performance. The distribution of radio resources considers the channel condition and QoS requirements like fairness and spectral efficiency. The classification of user request is performed to allocate the resources based on the predominant survey.

Camilo Orejuela Mesa (2006) had analyzed the packet scheduling mechanism in the enhanced uplink of HSPA. The rate scheduling is introduced in the low rate data transmission and Round Robin scheduling is used for high data rate transmission. The best channel conditions are evaluated by UCQI schedulers to provide resources to prioritized users. The estimation scheduler is implemented to estimate the maximum supported rate based on the path.

Oana Iosif and Banica (2011) had investigated the packet scheduling for downlink LTE systems. It uses the round robin mechanisms in time domain and frequency domain. The analysis is performed in terms of prioritized and non prioritized user request based on the limitation in the PDCCH downlink control channel. The traffic mixtures are designed to evaluate the cell throughput, fairness and system capacity. I-Hong Hou et
al (2009) had introduced admission control and scheduling mechanism to guarantee the QoS for variable bit rate applications. On this QoS requirements are identified based on traffic patterns, channel reliability, delay bound, and throughput bound. A mathematical model is developed to handle the variable bit rate applications. The latencies and throughput are regulated to deploy the efficient scheduling process to support a wide range of wireless applications with variable bit rate.

Ren-Hung Hwang et al (2012) had proposed downlink packet scheduling mechanism and resource allocation for hybrid networks. To improve the system throughput and ensure the QoS requirements of multimedia applications a two stage packet scheduling mechanism with application layer forward error correction is incorporated.

5.3 PROBLEM IDENTIFICATION AND PROPOSED SOLUTION

In previous chapter CSBCAC has proposed a call admission control algorithm based on channel state for LTE networks. The call requests are classified into HC, NC, VoIP call and video type and prioritized. When there is no sufficient resource available for VoIP or video calls, then resource degradation process is triggered for the existing calls. In this case, the resources that are used by the users in bad channel condition can be allocated for the requested VoIP and video calls. But in this UBSCAC, only the VoIP calls and the video calls are allocated to channels while the channels are not allocated for real time and non-real time users. This work considers the real time and non real time users for allocating the resources.
5.3.1 Resource Block Allocation for Non VoIP Users

In this section considers B resource blocks for each TTI for K mobile users which are serviced by an eNB. Among these users, W users run an application with an active connection to a server. This server is connected to the packet data network which is connected to eNB. This system includes the following parameters:

\[ D_i \] The average data rate achieved by the \( i \)th user at a time \( t \) when a scheduling decision is to be made.

\[ d_i \] The minimum required data rate for the \( i \)th user at \( t \).

\[ T_i \] The playback delay threshold for the \( i \)th user; the maximum allowable time before the user’s head of line packet in its queue can be delivered.

\[ Brb(i) \] number of resource blocks allocated to the time variable bit rate user in one TTI.

\[ Bn_{rb}(i) \] number of resource blocks allocated to a non real-time user.

\[ \eta_i \] the effective data rate of the \( i \)th user computed from the utility function of all the subcarriers (as if the user was allocated the whole of the system’s available band).

The number of resource blocks allocated to real time VBR users is then is determined using Equation (5.1).
\[ B_{\text{del},n} = \left( \frac{\mu}{\mu + \lambda} \right) \left( \frac{\Phi}{\sum_{i=0}^{\infty} \Phi \sum_{i=0}^{\infty} \left( \frac{d_i}{D_i} \right)} \right) B \] (5.1)

The network's operator assigns the parameters \( \mu \) and \( \lambda \). These parameters are selected such that the ratio signifies the amount of real-time users flowing through the network and amount of non-real-time users flowing through the network. For non-real traffic, the following rule is applied in the determination of resource blocks allocated.

\[ B_{\text{del}(1)} = \left( \frac{\mu}{\mu + \lambda} \right) \left( \frac{1}{T_1} \sum_{i=0}^{\infty} \frac{d_i}{D_i} \right) \] (5.2)

Few resource blocks may be allocated since both the rules have components that are rounded down. In order to improve the system's overall throughput users with highest utility functions in each block is used for allocating the remaining blocks.

### 5.3.2 Selection of Utility Function

For ensuring channel quality, the utility function with RSS as explained in Section 3.3 is considered. Hence, the utility function used in resource assignment for real-time and non-real-time users are given by Equation (5.3).

\[ Y_a(E_a(P_n, \delta_{x,n})) = \frac{E_a(P_n, \delta_{x,n})}{RSS_a} \] (5.3)

\[ Y_N = \text{Utility function of user N} \]
\[ E_N = \text{Rate} \]

\[ \rho_N = \text{Transmit power on the subcarrier} \]

\[ \delta_{\mu, \nu} = \text{Set of subcarriers} \]

\( RSS_N \) is the received signal strength achieved by user \( N \) over the last \( T \) TTIs. The utility function is calculated based upon the set of subcarriers; transmit power and the received signal strength.

Marginal utility calculation is calculated as follows:

\[ M_{n,c} = Y_n (E_n (\rho_n, \delta_{rb}, n, c^{1}) - Y_n (E_n (\rho_n, \delta_{rb}, n)) \]  \hspace{1cm} (5.4) \]

where \( \delta_{rb} \) is set of resource blocks and the marginal utility \( M_{n,c} \) represents the gain in the utility function \( Y_n \).

5.3.3 Scheduling Algorithm

Algorithm

Consider the \( n \) user requests \{R_1, R_2, ..., R_n\}.

Let us consider the user requests with good channels as \( G = \{G_1, G_2, ..., G_k\} \) and bad channels as \( B = \{B_1, B_2, ..., B_r\} \), where \( k, r < n \).

Among \( G \), handover calls are represented as \( H = \{H_1, H_2, ..., H_m\} \) and new calls as \( N = \{N_1, N_2, ..., N_p\} \), where \( m, p < k \).

Among \( H \), the VoIP calls and the video calls are represented as \( H_{vo} = \{V_1, V_2, ..., V_q\} \) and \( H_t = \{I_1, I_2, ..., I_t\} \) respectively, where \( q, t < m \).
Among $H$, the real time users and the non-real time users are represented as

$$H \bar{=} \{K_1, K_2, \ldots, K_t\} \text{ and } H_N = \{S_1, S_2, \ldots, S_t\} \text{ respectively.}$$

Let necessary RSS condition for satisfying handover is $RSS_v$ and the RSS threshold value is $RSS_L$.

Let $\eta_A$ be the total available bandwidth, $\eta_{VoT}$, $\eta_{in}$, $\eta_{B}$ be the reserved bandwidth for VoIP, video and bad channel classes, respectively. Let $M_{N,c}$ be the marginal utility function. $\delta_{\text{real},N(c)}$ be the set of available users, $\delta_{\text{mb},N(c)}$ be the set of resource blocks. $c$ is the user and $c-1$ is previous user.

1. For each $\{R_1, R_2, \ldots, R_n\}$
   1.1 If $RSS_v > RSS_L$,
      1.1.1 $G = \{G_i, i = 1, 2, \ldots, k\}$
      Else
      1.1.2 $B = \{B_i, i = 1, 2, \ldots, r\}$
      End if
   End for

2. For each $G = \{G_i, i = 1, 2, \ldots, k\}$
   2.1 For each $H = \{H_1, H_2, \ldots, H_m\}$
      2.1.1 For $H_{V_0} = \{V_1, V_2, \ldots, V_q\}$
      2.1.1.1 If $\eta_{VoT} < \eta_A$, then
      Bandwidth is reserved based on traffic density and call is admitted
      Else Go to Step 4.0
      End if
   End for
2.1.2 For $H_1 = \{l_1, l_2, \ldots, l_n\}$,

2.1.2.1 Check TOL of each call
2.1.2.2 Call with lower TOL is admitted first.
2.1.2.3 $\eta_A = (\eta_A - \eta_{th})$
2.1.2.4 If $\eta_{th} < \eta_A$, then
   Bandwidth is reserved and call is admitted
   Else
   Go to Step 4.0
   End if
End for

2.1.3 For $H_2 = \{K_1, K_2, \ldots, K_n\}$

2.1.3.1 Allocate first RB to user with highest marginal utility.
   \[ N^{(c)} = \arg\max_N M_{N,c} \]

2.1.3.3 If $M_{N,c} > 0$
   Allocate RB to user $N^{(c)}$
   \[ \delta_{th,N^{(c)}} = \delta_{th,N^{(c-1)}} + \{c\} \]
2.1.3.4 Delete RB from a set of available RBs
   \[ \delta_{avail,N^{(c)}} = \delta_{avail,N^{(c-1)}} - \{c\} \]
2.1.3.5 If $N^{(c)} = N^{(c-1)} - \arg\max_N M_{N,c}$
   Keep $N^{(c)}$ in $\delta_{avail,N^{(c)}}$
   Else
   Delete the user $N^{(c)}$ from the set of available users
   \[ \delta_{avail,N^{(c)}} = \delta_{avail,N^{(c-1)}} - \{N^{(c-1)}\} \]
   End if
2.1.3.6 If $\delta_{\text{avail},N^{10}}$ is empty
  
  Stop
  
  Else
  Repeat from Step 2.1.3.1
  
  End if
  
2.1.4 For $H_N = \{S_1, S_2 \ldots S_i\}$
  
  Repeat from Step 2.1.3.1
  
  End for
  
3. For each $N = \{N_1, N_2, \ldots, N_p\}$
  
  Repeat from Step 2.1.1
  
  End For
  
  End For
  
4. When resources availability is insufficient,
  
  For each $B = \{B_i, i = 1,2,\ldots r\}$
  
  4.1.1 $\eta_A \cdot \eta_A : \eta_B$
  
  Repeat from Step 2.1.1
  
  End for

In the admission control algorithm, $n$ denotes the number of user requests. Initially, RSS is calculated and when this RSS exceeds a threshold value RSS$_L$, then channel condition is considered as good channels. When RSS$_v$ is below the threshold value, then the channel condition is considered as bad channels.

Initially, the handover calls are considered which includes the VoIP calls and the video calls. Taking the VoIP calls, when the reserved bandwidth for the VoIP calls is lesser than the total available bandwidth, the bandwidth is reserved based on traffic density and the call is admitted. If the reserved bandwidth is larger, then the bandwidth reservation is done
using the resource of bad channels. The available bandwidth is the sum of available bandwidth and the bandwidth reserved for the bad channels.

Next, UBSCAC consider video calls for allocating in the good channels. The tolerance of latency is checked for each video call and the call having lower TOL is admitted first. Now the available bandwidth becomes the difference between the total available bandwidth and the reserved bandwidth for video call. If this reserved bandwidth becomes lesser than the available bandwidth, then the bandwidth is reserved for the calls and the call is admitted. If the reserved bandwidth is larger, then the bandwidth reservation is done using the resources of bad channels. The available bandwidth is the sum of available bandwidth and the bandwidth reserved for the bad channels.

Next UBSCAC considers the allocation for Real time users in good channels. Initially allocate first RB to the user with highest marginal utility. If marginal utility is greater than zero, then allocate resource block to user c. This resource block is deleted from the set of available RBs. If the user c and the previous user c-1 are allocated to the same RB, then user c is stored in the available users otherwise it is deleted from the list of available users. This process is repeated until the set of available users are empty. The same process is repeated for the non-real time users also. After allocating the handover calls, the new calls are considered which reserves the remaining bandwidth for VoIP and the video calls in the same way as described for the good channels.
5.4 RESULTS AND DISCUSSION

In this section, the proposed UBSCAC scheme is simulated using the Network Simulator 2.33 (NS2) which is a general-purpose simulation tool that provides discrete event simulation of user defined networks.

Table 5.1 USBCAC simulation parameters

<table>
<thead>
<tr>
<th>Number of Servers</th>
<th>02</th>
<th>Traffic Types</th>
<th>CBR, VBR, VoIP, Video</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number of aGW</td>
<td>02</td>
<td>Traffic rate</td>
<td>50–100 Kbps</td>
</tr>
<tr>
<td>Number of eNB</td>
<td>20</td>
<td>Number of packets per frame</td>
<td>02</td>
</tr>
<tr>
<td>Number of UE/eNB</td>
<td>100</td>
<td>VoIP Codec</td>
<td>GSM, AMR, H.323</td>
</tr>
<tr>
<td>Mobility speed</td>
<td>3–20 km/hr</td>
<td>Mobility way</td>
<td>Random Way point</td>
</tr>
</tbody>
</table>

In the simulation settings, servers are provided for, HTTP, FTP and signaling services, one aGW to provide HTTP cache and flow control, each end provides flow control information to 1000 UEs. The simulation topology is given in Figure 5.1. In this model, UL AirQueue is used for uplink flows in the link between UE and eNB. For the downlink flow, (i.e.) in the link between eNB and UE, DL AirQueue is used. For both the links, the link bandwidth is set as 500 kb and link delay as 2 ms. For the link between eNB and aGW, ULS1Queue is used and for the downlink between aGW and eNB, DLS1Queue is used.
Figure 5.1  Simulation topology for UBSC

For both the links, the link bandwidth is set as delay as 2 ms. For the link between the server and aGW, a queue is used with link bandwidth as 50 Mb and link delay model, DL_AirQueue provides each flow and each cell’s initial buffer size and average data rate; DL_AirQueue uses this current packet size to decide whether the packet is allowed into DL_AirQueue. If the buffer of the flow or the cell where the packet is going to be overflowing, the packet is blocked until both the cell’s buffer is not overflowing. UBSC AC meas...
Case - 1 (CBR)

A. Based on Rate

In this experiment, UBSCAC varies the data sending rate from 10 to 50 \text{kb} to measure the received bandwidth, fairness, throughput and delay for the CBR non-real time traffic.

It can be inferred from Figure 5.2 that the received bandwidth gradually increases when the rate is increased. The received bandwidth of the UBSCAC is higher than the existing CSBCAC scheme.

![Figure 5.2 Bandwidth of UBSCAC based on data rate](image)

From Figure 5.3, it can be observed that the delay of the proposed UBSCAC is less than the existing CSBCAC scheme.
Figure 5.3 Delay of UBSCAC based on data rate

Figure 5.4 and 5.5 shows the fairness and throughput obtained, respectively for the UBSCAC and CSBCAC schemes.

Figure 5.4 Fairness of UBSCAC based on data rate
Figure 5.5  Throughput of USBCAC based on data rate

From the figures, it can be seen that the throughput and fairness of both schemes are increased, when the rate is increased from 10 kb to 50 kb. It shows that, based on data rate the bandwidth utilization, throughput, fairness, is improved up to 4.4%, 3.2% and 0.80% respectively. Delay, call blocking rate, call dropping rates is reduced up to 6.6%, 4.1% and 2.9% respectively. Due increase in the availability of resources is reduced up to the call blocking probability is reduced up 4.1% and call dropping probability 2.9%. The throughput and fairness are slightly more for UBSCAC, because it minimizes packet losses using the channel estimation and tolerance of loss techniques.

B. Based on Users

In this experiment, UBCAC vary the number of UEs from 1 to 1000 in order to measure the received bandwidth, fairness, throughput and delay for the CBR non-real time traffic.
Figure 5.6 shows that the received bandwidth of the proposed UBSCAC is higher than the existing CSBCAC scheme. Scheduling decisions are made based on CQI, the appropriate resources are allocated. Due to this throughput increases up to 5.2% and the bandwidth is improved up to 6.4%.

![Bandwidth Graph](image)

**Figure 5.6 Bandwidth of UBSCAC based on users**

![Delay Graph](image)

**Figure 5.7 Delay of UBSCAC based on users**
From Figure 5.7, it can be observed that the delay of the proposed UBSCAC is less than the existing CSBCAC scheme. It produces 7.6% reduced delay when compared to existing CSBCAC.

![Graph showing fairness index vs number of users for CSBCAC and UBSCAC]

**Figure 5.8** Fairness of USBCAC based on users

![Graph showing packets vs number of users for CSBCAC and UBSCAC]

**Figure 5.9** Throughput of UBSCAC based on users
It can be observed from Figure 5.8 that the fairness of the proposed UBSCAC is 3.9% higher than the existing CSBCAC scheme. Due increase in the availability of resources is reduced up to the call blocking probability is reduced up 2.1% and call dropping probability 1.9%.

Case - 2 (Video)

A. Based on Rate

In this experiment, UBSCAC vary the data sending rate from 10 to 50 kb to measure the received bandwidth, fairness, throughput and delay for the video exponential traffic.

![Graph showing bandwidth of UBSCAC for video based on data rate](image)

Figure 5.10  Bandwidth of UBSCAC for video based on data rate
Figures 5.12 and 5.13 show the fairness and throughput obtained, respectively, for the UBSCAC and CSBCAC schemes. From the Figures 5.12 and 5.13, it can be observed that the throughput and fairness of both schemes are increased when the rate is increased from 10 kb to 50 kb. But it shows that the throughput is 5.2% more for UBSCAC when compared with CSBCAC, because it minimizes packet losses using the
channel estimation and tolerance of loss techniques. Due to increase in the availability of resources, the call blocking probability is reduced up to 4.1% and call dropping probability is reduced up to 2.9%.

![Graph showing throughput of UBSCAC for video based on rate](image)

**Figure 5.13** Throughput of UBSCAC for video based on rate

B. Based on Users

In this experiment, UBSCAC vary the number of UEs from 1 to 1000 in order to measure the received bandwidth, fairness, throughput and delay for the video exponential traffic.

From Figure 5.14, it can be observed that the received bandwidth of the proposed UBSCAC is 7.6% higher than the existing CSBCAC scheme. From Figure 5.15, it can be observed that the delay of the proposed UBSCAC is 9.8% less than the existing CSBCAC scheme.
Figure 5.14  Bandwidth of UBSCAC for video based on users

Figure 5.15  Delay of UBSCAC for video based on users

From Figure 5.16, it can be observed that the fairness of the proposed UBSCAC is 3.6% higher than the existing CSBCAC scheme. From Figure 5.17, it can be observed that the throughput of the proposed
UBSCAC is 9% higher than the existing CSBCAC scheme. Due to increase in the availability of resources, call blocking probability is reduced upto 2.6% and call dropping probability is reduced upto 1.6%.

![Graph showing fairness of UBSCAC for video based on users](image1)

**Figure 5.16  Fairness of UBSCAC for video based on users**

![Graph showing throughput of UBSCAC for video based on users](image2)

**Figure 5.17  Throughput of UBSCAC for video based on users**
Case - 3 (VoIP)

In this experiment, the data sending rate is varied from 10 to 50 kb to measure the received bandwidth, fairness, throughput and delay for the VoIP traffic. From Figure 5.18, it can be observed that the received bandwidth of the proposed UBSCAC is 7.2% higher than the existing CSBCAC scheme. From Figure 5.19, it can be observed that the delay of the proposed UBSCAC is 13% less than the existing CSBCAC scheme.

![Figure 5.18 Bandwidth of UBSCAC for VoIP based on users](image)

![Figure 5.19 Delay of UBSCAC for VoIP based on users](image)
Figures 5.20 and 5.21 show the fairness and throughput for the UBSCAC are improved up to 3.8% and 13.2% respectively when compared with CSBCAC schemes.

![Graph of Fairness of UBSCAC for VoIP based on users](image)

**Figure 5.20 Fairness of UBSCAC for VoIP based on users**

![Graph of Throughput of UBSCAC for VoIP based on users](image)

**Figure 5.21 Throughput of UBSCAC for VoIP based on users**

From the Figures 5.20 and 5.21, it can be seen that the throughput and fairness of both schemes are increased, when the users vary from 200 to 1000. But it shows that the throughput and fairness are slightly more for UBSCAC, due to the allocation of the separate resource blocks.
for VoIP schemes and increase in the availability of resources the call blocking probability is reduced up 4.3% and call dropping probability 2.9%. At the outset the UBSCAC produces the bandwidth utilization, throughput, fairness, is improved up to 5.4%, 4.2% and 1.80% respectively. Delay, call blocking rate, call dropping rates is reduced up to 5.6%, 3.1% and 1.9% respectively. Based on video traffic, bandwidth utilization, throughput, fairness, is improved up to 7.4%, 10% and 3.8% respectively. Delay, call blocking rate, call dropping rates is reduced up to 11%, 2.6% and 1.6% respectively. The prioritization and classification of call request based on utility factor, CAC mechanisms are freed to allocate the maximum resources. But delay and response time can be reduced further to improve the efficiency.

5.5 SUMMARY

In this chapter, UBSCAC is proposed to design a call admission control algorithm which schedules the channels in real time and non-real time users. The call requests are classified into NC request and HC request and the type of services are classified as VoIP and video. Then based upon the RSS value, the channel is estimated as good channel or bad channel. Resource allocation is made for VoIP calls based on traffic density and for video calls: it is done based on the tolerance of limit. For allocating resources to other users, the utility function is calculated based on the channel condition. Then the real time users and the non-real time users allocate resource blocks based upon the highest marginal utility function. When there are no sufficient resources to allocate, it allocates the resources of bad channel users there by degrading their service. The simulation results show that this admission control algorithm provides channel quality and prioritizes the handover calls over new calls which allocates resources to all kinds of users.