CHAPTER 4

CHANNEL STATE BASED CALL ADMISSION CONTROL FRAMEWORK

This chapter discusses about Channel State Based Call Admission Control (CSBCAC) framework for real time and non-real time data transfer in 3GPP LTE networks. There are several works done on real data transfer in LTE networks but these works rarely considers the channel quality. The existing techniques didn’t provide differentiation between the new call and handoff call, video traffic is not considered and channel quality for each user is not considered. In order to overcome the above mentioned problems, the proposed to design of call admission control algorithms based on channel state. This call admission control algorithm includes three phases: call classification, channel state estimation and call admission. Initially, call requests are classified into NC request and HC request and the type of services are classified as VoIP and video.

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. When there are no sufficient resources to allocate, it allocates the resources of bad channel users there by degrading their service. Thus the simulation result shows that this admission control algorithm provides channel quality and prioritizes the handover calls over new calls. This proposed method is extended from non real time traffic and provides the allocation mechanism for ensuring the QoS in LTE networks.
4.1 INTRODUCTION

LTE provides various kinds QoS services based on Radio Resource Management (RRM) in wireless networks. In order to provide best effort QoS, the incoming call requests and ongoing call requests are guaranteed with a certain level of QoS and no congestion during transmission. Oyebisi and Ojesanmi (2008) had proposed congestion control scheme for wireless network. The call admission control mechanism has several conditions to provide the efficient service to the user request based on channel status. CAC is classified into two major categories, static and dynamic. In static CAC method channel reservation is not appropriate due to inefficient spectrum utilization. Due to the variation in availability of resources in the RRM, the required resource for the call request is not predicted statically.

In dynamic CAC algorithms, the required resources are assumed for all the call requests based on QoS request. Call admission control algorithm considers the availability of the resources needed to guarantee the required QoS of the new call, and the QoS maintenance of already accepted calls in order to decide upon the admission of a call request. Physical Resource Blocks (PRBs) are estimated for each service request from the user. Sueng Jae Bae et al (2009) proposed resource estimated call admission control in which the minimum data rate for each call request are estimated by the CAC and resources are utilized at maximum level.

4.1.1 Channels in LTE System

CAC algorithm monitors all incoming call requests periodically and estimates the required resources continuously. The incoming call requests are received beyond the levels of available resources, some of the
incoming calls are blocked or dropped based on the arrival rate. To maintain the adequate arrival rate of the handoff call and new call, the system uses pre request scheme and guard channel scheme. Feng Ming et al (2012) had investigated that Forced Termination Probability (FTP) mechanism is used to terminate the failure call requests in the network in order to maintain the QoS. Figure 4.1 shows the process of call admission process based on channel estimation. In this the current load of the new call request is estimated and comparison is made with the acceptable load of the system.

Figure 4.1 Call admission control based on channel estimation

The following steps are involved in the call admission control based on channel estimation:

i) If the estimated load (LE) is greater than the acceptable load (LA) then the calls will be terminated.
ii) If the LE is less than LA the call request is admitted to the network and resources will be allocated based on the availability of the channel.

4.1.2 Channel Types

In LTE system the data are transferred to all nodes of the radio interface through the dedicated facility called “Channels”. The channels are used to classify the data into different categories and process them for the transportation in throughout the network. The protocol architecture of the LTE system provides the interface for the different types of the channels to perform the effective communication. The channels are classified into three major categories. Those are as follows:

- Physical Channels

- Transport Channels

- Logical Channels

4.1.2.1 Physical channels

Physical channels are responsible for the transfer of user data and control information for various nodes of the system. Physical channel’s operations are differed for uplink and downlink transmission. Figure 4.2 shows the presence of various channels of the 3GPP LTE system. Physical channels provide the interface between the transceiver and the physical layer of protocol architecture. For downlink operations, LTE uses the following channels:
Physical Broadcast Channel (PBCCH)

PBCCH is used to transfer required UE’s information for accessing the network and it uses Quadrature Phase Shift Keying (QPSK) modulation scheme. It broadcasts the required access information with the parameters like Master Information Block (MIB), downlink system bandwidth, channel structure, Hybrid ARQ and system frame number. PBCCH uses sub carriers to map the system bandwidth with available resources of the system.

(Source: http://wired-n-wireless.blogspot.in/2009_09_01_archive.html)

Figure 4.2 Layered channel architecture

Physical Control Format Indicator Channel (PCFICH)

If the signal is received from the system it indicates to the UE about the format of the signal received with reception parameters. The control frame indicator consists of number of OFDM symbols used for
control channel transmission for each frame of the sub frame and it uses a QPSK modulation scheme.

- **Physical Downlink Control Channel (PDCCH)**

  This channel defines scheduling parameters for resource allocation for downlink, paging and uplink power control parameters. PDCCH coordinates the controls for transferring the of Downlink Control Information (DCI) to one or more UEs in the network.

- **Physical Hybrid ARQ Indicator Channel (PHICH)**

  PHICH is used to define the ARQ status for the channel and it carries the reception information with acknowledgement (HARQ ACK). The transfer message contains HARQ indicator to provide the received status for HARQ. If the HARQ indicator bit is set 1, the ACK is received else NACK is received from the transport block.

- **Physical Random Access Channel (PRACH)**

  PRACH is a corresponding channel for transport layer’s Random Access Channel (RACH). It is one to one mapped with RACH, PRACH is used to transfer the information about access slots, power control information, signatures, data spreading factor and sub channel access information. PRACH is used to identify the propagation delay during the uplink.

- **PUSCH (Physical Uplink Shared Channel)**

  PUSCH transmits uplink shared channel data and control information.
4.1.2.2 Transport channels

In LTE system, transport channels are mapped with physical channels in a direct manner. Transport channels are designed as a gateway to access the MAC PDU from the physical channel. The transport channels are classified as follows:

- **Broadcast Channel (BCH)**

  Broadcast channels are used to provide the signaling information to all the UEs in the network. UEs use this signaling information to discover a network, synchronize the network and to make the connection.

- **Downlink Shared Channel (DL-SCH)**

  DL-SCH is focused on downlink data transfer to the UE’s of the network. Based on the availability of the resource blocks, UEs shares the resources dynamically and that are controlled dynamic multiplexing of UEs on the PDSCH.

- **Paging Channel (PCH)**

  In paging channel, the paging procedures are transmitted among the UE’s of the cell. PCH is associated with the paging indicators which are generated by the physical layer.

- **Multicast Channel (MCH)**

  Multicast channels are used to provide point to multi point downlink communication between the network and the UE. It also used to transfer MBMS data through multicast transmission.
Uplink Shared Channel (UL-SCH)

UL-SCH carries traffic data and transfers control signals for the corresponding higher layers of the network. Figure 4.3 shows the mapping of UL-SCH with the higher layer channel.

![Diagram showing mapping of channels]

**Figure 4.3  Mapping of channels**

4.1.2.3 Logical channels

Logical channels are used to transfer the information that is offered by MAC. Logical channels are classified into two broad categories.

- Control Channels (Transfer control plane information)
- Traffic Channels (Transfer user plane information)
• **Control Channels**

Control channels transfers of control plane information. The control channels offered by MAC are:

- **Broadcast Control Channel (BCCH)**

  BCCH is a downlink channel for broadcasting system that transfers the control information.

- **Paging Control Channel (PCCH)**

  It transfers paging information and system information change notifications.

- **Common Control Channel (CCCH)**

  This channel for transfers the control information between UEs and network.

- **Dedicated Control Channel (DCCH)**

  A point-to-point, bi-directional channel transfers dedicated control information between a UE and network.

• **Traffic Channels**

Traffic channels transfers the user plane information. The traffic channels offered by MAC are:
Dedicated Traffic Channel (DTCH)

A Dedicated Traffic Channel (DTCH) is a point-to-point channel, dedicated to one UE, for the transfer of user information.

Multicast Traffic Channel (MTCH)

It is a point-to-multipoint downlink channel for transmitting traffic data from the network to the UE. This channel is only used by UEs that receive MBMS.

4.2 LITERATURE SURVEY

Sueng Jae Bae et al (2009) had proposed resource estimated CAC algorithm. The numbers of PRBs which are allocated to the requested call are estimated in this algorithm. The type of the requested service and current Modulating and Coding Scheme (MCS) level of UE are required for determining the number of required PRB. In addition, the resource-estimated CAC algorithm calculates available PRBs based on PRB usage of ongoing call measured by eNB.

Spaey et al (2010) had defined the reference admission control algorithm. This algorithm considers the time-varying cell capacity in order to distinguish between the real-time (RT) and non-real time (NRT) calls so that the handover calls can be given priority over the fresh calls. In addition to the reference admission control algorithm, they proposed the self-optimizing algorithm. The reference admission control algorithm auto-tunes the HO which is considered as a main parameter of this algorithm.

Haipeng Lei et al (2008) had proposed a resource-scheduling algorithm and a connection access control scheme for LTE systems with
heterogeneous services. In this scheduling algorithm, the transmission
Guard interval gives higher priority to the RT service packets which
Approaches the delay deadline. This proposed CAC based on RB allocation
can balance the ongoing connections of different class of traffic and easy to
Reserve the resource and support handover users potentially.

Antonopoulos and Verikoukis (2010) had introduced a CAC
Mechanism for IEEE 802.16 broadband wireless access standard.
The problem of “busy hour” in communications traffic variation is
Considered in this algorithm which provides the basis for the bandwidth
Reservation concept. WiMAX infrastructure is mainly considered in this
Algorithm so that it can be adapted for LTE.

By far, there are limited works for call admission control in LTE
And not all those works consider the channel quality while performing call
Admission control. In the proposed adaptive connection admission control
Algorithm for LTE networks, considers real time and non real time
Services. However, this work does not provide differentiation to new call
And hand off call, which is the major issue of LTE.

VoIP calls get the priority whereas the video traffic is not
Considered. To overcome the above-mentioned problem, channel-state
Estimation based call admission control algorithm for LTE networks is
Proposed. It is quite wearisome when ongoing call is lost when compared
to preventing a new call. QoS controlling is essential so that blocking of
New connections and dropping of handoff connection can be reduced
effectively. Thus efficient call admission control algorithm is needed to
Implement in the cellular networks.
4.3 SYSTEM MODEL

4.3.1 Channel State based Call Admission Control

The channel estimation technique is based on the RSS value. Initially, an optimal RSS value is set as a minimum threshold value. RSS of the channel is calculated periodically and compared with a threshold value. If the calculated RSS value is greater than threshold value then the channel is considered as a good channel otherwise bad. When a call request arrives to the network, it is checked for HC or NC. If it is an HC, then it is handled first by the scheduler. After classifying the call as HC or NC, the scheduler checks for its class. If it is a VoIP call, then its bandwidth requirement is checked.

If it is less than the total available bandwidth, the bandwidth can be reserved based on the traffic density of the base station. For video calls, if the requested bandwidth meets the remaining available bandwidth, it can be admitted. If there are multiple video call requests, then the Tolerance of Latency (TOL) of each call is checked. The call with low TOL can be admitted first. 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 with bad channel condition can be allocated for the requested VoIP and video calls.

4.3.2 Classification of Call Requests

Both types are further divided into an RT class of services and best effort services (BE). RT class can be prioritized based on the type of service as VoIP and video. Oversubscription of VoIP networks can be prevented using an admission control algorithm and it is used in the call
setup phase. The real-time media traffic uses the call admission control as its main application. The harmful effects of other voice traffic can be avoided due to the distinctive characters of QoS tools and unwanted voice traffic can be excluded from the network. This happens to be a preventive congestion control procedure since it prevents the voice traffic congestion and ensures sufficient bandwidth for authorized flows.

4.3.3 Received Signal Strength

The handover procedure is used by the UE in the network controlled LTE in order to provide mobility in connected mode. The serving node receives a measurement report after calculating the power of signal strength RSS by UE. The distance between UE and its associated Node K is calculated as the RSS of UE and its RSS value at the time t is denoted as:

\[
\text{RSS}_n = \text{Tr} - 10\log(1) + X_{\text{dB}}
\]  \hspace{1cm} (4.1)

where

\( l \) : is the distance between the UE and the associated AP

\( \text{Tr} \) : is the transmitted signal power

\( X_{\text{dB}} \) : is a Gaussian random variable with zero mean.

The \( \text{RSS}_n \) value of the node K candidates relate to significant RSS higher than a threshold value \( \text{RSS}_U \):

\[
\text{RSS}_n > \text{RSS}_U
\]  \hspace{1cm} (4.2)
The received measurement reports help in decision making for the handover process at serving node K. Based upon the RSS value related to the serving node K, handover decision is made. Once the RSS value goes below the limiting value threshold $\text{RSS}_{K}$, the channel is considered as bad channel. Thus the necessary condition of handover decision is checked by the following conditions:

$$\text{RSS}_{1} < \text{RSS}_{K} \quad (4.3)$$

When this condition is satisfied, then the channel is considered as a good channel.

### 4.3.4 Bandwidth Reservation

The bandwidth reservation concept is the basis for the admission control mechanism and execution of this concept is under the "busy hour" conditions. The class of the connection request that has arrived recently is verified for the UGS connections which has high arrival rate. When the total available bandwidth ($\text{BW}_T$) of the UGS connections is adequate in order to serve the incoming connection then the request can be accepted. The VoIP calls need to be prioritized over other types of connections, so the service types of real time polling services/ non-real time polling services (rtPS / nrtPS) should be provided with a restricted bandwidth (Tb-Rb). The requests need to be admitted to dealing with BE connections but there is no need of considering bandwidth allocation since QoS guarantees are not needed by the BE flows. According to the traffic intensity of the VoIP calls, there is a need to change the reserved bandwidth for UGS connections:

$$n_B = [1 \times B] \times n_h \quad (4.4)$$
where

\[ I = Ar \times Sr \]

\( I \) - traffic intensity which is a measure of the average occupancy of the base station during a specified period of time.

\( Ar \) - the arrival rate for UGS connections

\( Sr \) - mean service rate

\( n_l \) - bandwidth needed for each UGS connection

\( B \in [0, 1] \) - Bandwidth reservation factor.

UGS connection and its bandwidth reservation is displayed as Equation (4.4). The rtPS and nrtPS service types of the available bandwidth can be decreased by bandwidth reservation scheme and it also has an effect on increasing the blocking probabilities for the specific service types. When the portion of the bandwidth is entirely dedicated to this service type, the blocking probability of UGS connections can be decreased. The usage of ineffective system resources needs to be avoided in this bandwidth reservation scheme. The increase in VoIP calls due to frequent traffic variation can be predicted and solved using this technique.

4.3.5 Tolerance of Latency

The partition between the incoming traffic for each class is considered in the CAC algorithm so that handoff calls can be prioritized over new calls effectively. Based upon the QoS profile such as latency tolerance the arrival calls can be categorized into three classes namely,
a. Non Real time service (NRT)

b. Real Time tolerant service (RT-TLR)

c. Real Time intolerant service (RT-INTLR)

The numbers of resource blocks are insufficient when similar types of call arrive at the network. This causes overloading of cell and the connection requests cannot be satisfied. Then the delayed requests are stored in specific queues and due to latency depended type of traffic, these calls are considered in a different manner. Thus, three different queues are used (for each class of service) for each type of call. The latency $\delta q$ of a user requiring a request depends only on emission $\delta e$ and the reception time $\delta r$ of the request:

$$
\delta q = \delta e - \delta r
$$

Based upon the condition of latency, the requests in the wait state are treated. Initially, the requests having minimum tolerated latency take into consideration provided that this value doesn’t exceed the maximum latency. The temporal constraints need to be verified when a call has two requests for HC or NC are asking for two different applications of class of service.

A request for HC (or NC) with the class of service $i$

$$
\delta q, i < L_{max,i}
$$

A request for HC (or NC) with class of service $j$

$$
\delta q, j < L_{max,j}
$$
The HC (or NC) which will be treated at first that is solved by the Equation (4.8).

\[ P = \min (L_{\max, i} - \delta_{qi}, L_{\max, j} - \delta_{qj}) \tag{4.8} \]

To satisfy the priorities for the handover call over the new call taking the QoS requirement, the CAC proposes a RBs reservation algorithm.

4.4 IMPLEMENTATION

Admission Control Algorithm

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

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

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

Among \( H \), the VoIP calls and the video calls are represented as \( H_{v0} = \{V_1, V_2, \ldots, V_q\} \) and \( H_l = \{l_1, l_2, \ldots, l_t\} \) respectively, where \( q, t < m \).

Let RSSv and RSSl is used to validate the necessary condition.

Let \( \eta_A \) be the total available bandwidth, \( \eta_{\text{hot}}, \eta_{\text{lt}}, \eta_{\text{lb}} \) be the reserved bandwidth for VoIP, video and bad channel classes respectively.

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 \( G = \{G_i, i = 1,2,\ldots,k\} \)

\[ 2.1 \quad \text{For each } H = \{H_1, H_2, \ldots, H_m\} \]

\[ 2.1.1 \quad \text{For } H_{k_0} = \{V_1, V_2, \ldots, V_q\} \]

\[ 2.1.1.1 \quad \text{If } \eta_{k_0} < \eta_{\text{A}}, \text{ then} \]

\text{Bandwidth is reserved based on traffic density and call is admitted}

Else Go to Step 4.0
End if
End for

\[ 2.1.2 \quad \text{For } H_1 =\{I_1, I_2, \ldots, I_t\}, \]

\[ 2.1.2.1 \quad \text{Check TOL of each call} \]

\[ 2.1.2.2 \quad \text{Call with lower TOL is admitted first.} \]

\[ 2.1.2.3 \quad \eta_{\text{A}} = (\eta_{\text{A}} - \eta_{\text{I}}) \]

\[ 2.1.2.4 \quad \text{If } \eta_{\text{I}} < \eta_{\text{A}}, \text{ then} \]

\text{Bandwidth is reserved and call is admitted}

Else Go to Step 4.0
End if
End for
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 $B = \{Bi, i = 1, 2, \ldots, r\}$

4.1.1 $\eta_{A} = \eta_{A_1 \cdot \eta_{B}}$

   Repeat from Step 2.1.1

   End for

In the admission control algorithm, initially the RSS is calculated and when this RSS exceeds a threshold value $RSS_{th}$, then channel condition is considered as good channels. When $RSS_{th}$ is below the threshold value, then the channel condition is considered as bad channels.

Now, the algorithm considers the requests with good channels in which the handover call requests and the new call requests are allocated. 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, video calls are considered in 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. 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.

4.5 RESULTS AND DISCUSSION

In this section, the simulation of the proposed Channel State Based CAC (CSBCAC) scheme is performed using Network Simulator 2.33 (NS2) proposed by Qiu et al (2009) which is a general-purpose simulation tool that provides discrete event simulation of user-defined networks. The simulation parameters are given in Table 4.1.

<table>
<thead>
<tr>
<th>Table 4.1</th>
<th>Simulation parameters for CSBCAC</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number of Servers</td>
<td>02</td>
</tr>
<tr>
<td>Number of aGW</td>
<td>02</td>
</tr>
<tr>
<td>Number of eNB</td>
<td>20</td>
</tr>
<tr>
<td>Number of UE/eNB</td>
<td>100</td>
</tr>
<tr>
<td>Mobility speed</td>
<td>3~20 km/hr</td>
</tr>
</tbody>
</table>

In the simulation settings, one server to provide HTTP, FTP and signaling services, two aGW to provide HTTP cache and flow control, one
eNB is to provide flow control information and 1000 UE topology is given in the Figure 4.4.

Figure 4.4  CSBCAC simulation topology

In this model, UL AirQueue is used for uplink between UE and eNB. For the downlink flow, (i.e.) in eNB and UE, DL AirQueue is used. For both the links, the link delay is set as 500 kb and link delay as 2 ms. For the link between ULS1Queue is used and for the downlink between DLS1Queue is used. For both the links, the link bandwidth and link delay as 2 ms. For the link between the server an
where the flow locates is going to be overflowing, the packet is blocked until both the flow's buffer and the cell's buffer is not overflowing. The received bandwidth is measured in MB/s and throughput in terms of number of packets and end-to-end delay for both VBR and CBR traffic flows. Comparison is made for the proposed scheme with the DBACAC scheme.

- Case 1 (UDP)

A. Based on UE

In this experiment, CSBCAC vary the number of UEs from 1 to 1000 in order to measure the received bandwidth, throughput and delay for the UDP non-real time traffic.

Figure 4.5 shows the comparison of CSBCAC with DBACAC based on bandwidth utilization. CSBCAC produces the efficient bandwidth utilization, which is improved up to 10% when compared with DBACAC.
Figure 4.6 shows comparison of CSBCAC with DBACAC based on delay. CSBCAC reduces the delay in the uplink and the downlink queue also it reduces the transmission delay. The delay reduction is improved up to 7.1% when compared with DBACAC.

Figure 4.6  CSBCAC delay reduction based on UE

Figure 4.7 shows the throughput obtained with the CSBCAC and DBACAC schemes.

Figure 4.7  CSBCAC’s throughput based on UE
From the figure, it can be inferred that the throughput of both schemes is increased, when the UEs are increased. But it shows that the throughput is more for CBECAC, because it minimizes packet losses using the channel estimation and tolerance of loss techniques.

**B. Based on Rate**

In this experiment, CSBCAC vary the data sending rate from 150 to 300 kb to measure the received bandwidth, throughput and delay for the UDP non-real time traffic.

It can be inferred from Figure 4.8, the received bandwidth gradually increases when the number of users is increased. It shows that CSBCAC better than DBACAC.

![Figure 4.8 Bandwidth based on data rate](image)

Figure 4.9 shows the comparison of CSBCAC and DBACAC based on delay. Due the higher throughput and the bandwidth utilization the delay is reduced up to 6.1%.
Figure 4.9  Delay based on data rate

Figure 4.10 shows the throughput obtained with the CSBCAC and DBACAC schemes. The throughput for each node in the eNB increases up to 28%.

Figure 4.10  Throughput based on data rate
From the figure, it can be inferred that the throughput of both schemes is increased, when the UEs are increased. But it shows that the throughput is more for CSBCAC, because it minimizes packet losses using the channel estimation and tolerance of loss techniques.

Case - 2 (Video)

A. Based on Number of UEs

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

Figure 4.11 shows the throughput obtained with the CSBCAC and DBACAC schemes. The throughput of both schemes is increased, when the UEs are increased. But it shows that the throughput is more for CSBCAC, because it minimizes packet losses using the channel estimation and tolerance of loss techniques.

![Graph showing Bandwidth utilization for video based of UE](image-url)
It can be inferred from Figure 4.12 that the received bandwidth gradually increases when the number of users is increased. It shows that CSBCAC provides better results when it is compared with DBACAC.

Figure 4.12  Delay for video based of UE

Figure 4.13  Throughput for video based of UE
B. Based on Rate

In this experiment, CSBCAC vary the data sending rate from 150 to 300 kb to measure the received bandwidth, throughput and delay for the VBR exponential traffic.

![Bandwidth utilization for VBR based on data rate](image)

**Figure 4.14** Bandwidth utilization for VBR based of data rate

Along with simulation settings mentioned in the Chapter 1, CBSCAC uses ULS1Queue for uplink and for the downlink between aGW and eNB, DLS1Queue is used. For both the links, the link bandwidth is set as 5 Mb and link delay as 2 ms. Figures 4.13 - 4.16 shows the comparison of CSBCAC with DBACAC. The channel estimation of each call request produces the improvement in bandwidth utilization, throughput and fairness, by 10%, 27.5% and 1.3% respectively when compared with DBACAC.
The availability of adequate resources and good channels leads to the reduction in delay, call blocking rate and call dropping rate up to 6.3%, 3% and 2% respectively. The bandwidth utilization, throughput, fairness for video signals, are improved up to 12%, 29% and 1.6%
respectively. Delay, call blocking rate, call dropping rates is reduced up to 6.46%, 1.4% and 2% respectively. CSBCAC endow with maximized resource utilization with the incorporation of DBACAC, conversely CBSCAC fall short inadequacy in terms of QoS based channel utilization and resource allocation.

4.6 SUMMARY

In this chapter, CSBCAC has been proposed to design a call admission control algorithm based on channel state. This call admission control algorithm includes three phases, call classification, channel state estimation and call admission. Initially, the call requests are classified into NC request and HC request and the type of services are classified as VoIP and video. CSBCAC prioritize HC over NC and VoIP over video type. Then based upon the RSS value, the channel is estimated as good channel or bad channel. When the RSS of a channel is greater than threshold it can be allocated to the good channel else they are allocated to bad channels. Tolerance of latency is taken into consideration for multiple video calls. The latency of a user requiring a request depends on the emission and the reception time of the request. After allocating all the good channels to the call requests, resource degradation is processed for allocating the bad channels to the remaining calls. Thus from the simulation results CSBCAC have proved that this admission control algorithm provides channel quality and prioritizes the handover calls after new calls. But it fails to allocate the optimum resources to the user request. This can be further improved to make the availability of good quality resources.