Efficient Channel State Based Call Admission Control for Non Real Time traffic in LTE (3GPP) Networks

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Abstract

In Long Term Evolution (LTE) 3GPP Networks, several works were done on call admission control but these works rarely considers the channel quality in the controlling process. The existing techniques don't provide differentiation between the new call and the handoff call. We propose 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, call requests are classified into new call request and handoff call request and the type of services are classified as VoIP and Video. We prioritize HC over NC and VoIP over Video type. Then based upon the received signal strength value, the channel is estimated as good channel or bad channel. Thus from our simulation results we show that this admission control algorithm provides channel quality and prioritizes the handover calls over new calls.

Keywords: CAC, Handoff, LTE, RSS.

1. Introduction

I. Call Admission Control (CAC)

Call admission control algorithm considers the availability of the resources needed to guarantee the required Quality of- Service (QoS) of the new call, and the QoS maintenance of already accepted calls in order to decide upon the admission of a call request. [5]

II. Call Admission Control in LTE networks

The eNodeB in LTE provides basis for the admission control algorithm and is capable of operating separately on a per cell basis. [5] Congestion avoidance is the main aim of CAC scheme which limits the number of ongoing connections in the system or denies new connection request so that QoS can be maintained and delivered to different connections at the target level. [7]

The below two conditions need to be satisfied in the CAC algorithm in order to admit the user to the network: [9]

- Good signal strength

  Since eNB provides maximum signal, the mobile selects this node and shortage in coverage can be caused when signal goes below a certain threshold. The mobile may get blocked in this situation.
- Resource availability in the selected eNB

  Huge amount of physical resources between a minimum and maximum threshold are provided by the mobile. Available resources are checked by the eNB once the initial condition is checked. Call gets blocked once the eNB goes below a minimum resource threshold.

III. Issues of Call Admission Control algorithm

There are some issues, while designing an efficient call admission control algorithm (CAC),

- Diverse QoS requirements for delay-sensitive and delay-tolerant applications and the heterogeneous traffic patterns of calls originating from different applications makes the design of efficient admission control algorithms more challenging. [5]
- It is quite frustrating for a user when an ongoing call is dropped while handed over than when a fresh call is rejected initially. Thus it is important for the LTE system to accept the handover calls prior to the fresh calls. [5]

IV. Problem Identification and Proposed Solution

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 **[7]** they have proposed adaptive connection admission control algorithm for LTE networks, which concerns 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.

In **[8]**, VoIP calls get the priority whereas the video traffic is not considered.

To overcome the above-mentioned problems, we propose channel-state estimation based call admission control algorithm for LTE networks. 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 needs to implement in the cellular networks. This call admission control algorithm includes three phases Call Classification, Channel State Estimation and Call Admission.

2. Related Work

Sueng Jae Bae et al. **[2]** have 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 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 on-going call measured by eNB.

K. Spaey et al. **[5]** have 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-optimising algorithm. The reference admission control algorithm auto-tunes ThHO which is considered as a main parameter of this algorithm.

Haipeng Lei et al. **[7]** have proposed a resource-scheduling algorithm and a connection access control scheme (CAC) 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 traffics and easy to reserve the resource and support handover users potentially

A. Antonopoulos et al. **[8]** have introduced a connection admission control (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 bandwidth reservation concept. WiMax infrastructure is mainly considered in this algorithm so that it can be adapted for long term evolution (LTE).

3. Proposed Work

I. Overview

In this paper, the call requests are classified into two types as new call (NC) request and handoff call (HC) request. Both types are further divided into real time (RT) class of services and best effort services (BE). Real time (RT) class can be prioritized based on the type of service as, VoIP and Video. We prioritize HC over NC and VoIP over Video type.

After the classification of the call requests, the channel estimation technique is based on the received signal strength (RSS) value. Initially, we set an optimal RSS value as a minimum threshold value. RSS of the channel is calculated periodically and compared with threshold value. If the calculated RSS value is greater than threshold value then the channel is considered as good channel otherwise bad.

When a call request arrives to the network, it is checked for HC or NC. If it is a 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 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.

II. Classification of Call Requests

The call requests in the network are classified as new call (NC) request and handoff call (HC) request. When a initial call connection establishment is failed it is referred as the new call and the in-service calls moving from one cell to another is blocked then it refers to handoff call. Both types are further divided into real time (RT) class of services and best effort services (BE). Real time (RT) class can be prioritized based on the type of service as, VoIP and Video. We prioritize HC over NC and VoIP over Video type.

Oversubscription of VoIP networks can be prevented using Admission Control Algorithm and it is used in the call set-up 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 Quality of Service tools and also 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. [10]

III. Received Signal Strength

The handover procedure is used by the User Equipment (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. Distance between an UE and its associated Node K is calculated as the RSS of UE and its RSS value at the time t is denoted as:

$$\mathbf{RSSn = Tr - 10log\,(l) + Xdb} \quad \ldots\ldots\ldots\ldots\ldots\ldots \mathbf{(1)}$$

Where **l:** is the distance between the UE and the associated. AP

Tr: is the transmitted signal power, and

Xdb: is a Gaussian random variable with zero mean.

The Node K candidates are those which relate to significant RSS higher than a threshold value RSS_U:

$$\mathbf{RSS_L > RSS_U} \quad \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots \mathbf{(2)}$$

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 RSS_L, the channel is considered as bad channel.

Thus the necessary condition of handover decision is checked by the following condition:

$$\mathbf{RSS_V < RSS_L} \quad \ldots\ldots\ldots\ldots\ldots\ldots\ldots \mathbf{(3)}$$

When this condition is satisfied, then the channel is considered as a good channel. [10]

IV. 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 (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 for 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 need to change the reserved bandwidth for UGS connections and it is represented in equation 4.

$$\eta_T = [I \times B] \times \eta_1 \quad \ldots\ldots\ldots.. (4)$$

Where

I =*Ar/ 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

η_1 - bandwidth needed for each UGS connection

B€[0, 1] - Bandwidth reservation factor.

The rtPS and nrtPS service types of the available bandwidth can be decreased by this bandwidth reservation scheme and it also has effect on increasing the blocking probabilities for the specific service types. Conversely when the portion of the bandwidth is entirely dedicated to this service type, the blocking probability for UGS connections can be decreased.

The usage of ineffective system resources needs to avoided in this bandwidth reservation schemes. The increase in VoIP calls due to daily traffic variation can be predicted and solved using this technique. [8]

V. Tolerance of Latency

The partition between the incoming traffic for each class is considered in 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) and

c) real time intolerant service (RT-INTLR).

The number of resource blocks is 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 δq of a user requiring a request depends only on emission δe and the reception time δr of the request:

$$\delta q = \delta e - \delta r \quad \ldots\ldots\ldots\ldots (5)$$

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

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

$\delta q,i<Lmax,i$ ………… (6)

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

$\delta q,j<Lmax,j$ ………….. (7)

The HC (or NC) which will be treated the first is that which solves the following equation:

P = min (Lmax,i-δq,i , Lmax,j-δq,j) … (8)

To satisfy the prioritization for the handover call over the new call taking the QoS requirement, the CAC proposes a RBs reservation algorithm. [10]

VI. Admission Control Algorithm

Algorithm 1:

Consider the n user requests $\{R_1,R_2,\ldots.Rn\}$.

Let us consider the user requests with good channels as G = $\{G_1, G_2, \ldots G_k\}$ and bad channels as B = $\{B_1, B_2, \ldots Br\}$, 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 $H_{v0} = \{V_1,V_2,\ldots Vq\}$ and $H_I = \{I_1,I_2,\ldots I_t\}$ respectively, where q,t<m.

Let necessary RSS condition for satisfying handover be **RSSv** and the RSS threshold value be **RSS_L.**

Let η_A be the total available bandwidth, η_{vot}, η_{it}, η_B be the reserved bandwidth for VoIP, video and bad channel classes , respectively.

1. For each $\{R_1, R_2, \ldots. Rn\}$
   1.1 If $RSS_V > RSS_L$,
      1.1.1 $G = \{Gi, i = 1, 2,\ldots k\}$
   Else
      1.1.2 $B = \{Bi, i = 1,2,\ldots r\}$
   End if
   End for
2. For each $G = \{Gi, i = 1, 2, \ldots k\}$
   2.1 For each $H = \{H_1, H_2, \ldots.H_m\}$
   2.1.1 For $H_{v0} = \{V_1,V_2,\ldots Vq\}$
   2.1.1.1 If $\eta_{vot} < \eta_A$, then
   Bandwidth is reserved based on traffic density and call is admitted
   Else
   Goto step 4.0
   End if
   End for

   2.1.2 For $H_I = \{I_1,I_2,\ldots I_t\}$,
   2.1.2.1 Check TOL of each call
   2.1.2.2 Call with lower TOL is admitted first.
   2.1.2.3 $\eta_A = (\eta_A - \eta_{it})$
   2.1.2.4 If $\eta_{it} < \eta_A$, then
   Bandwidth is reserved and call is admitted
   Else
   Goto step 4.0
   End if
   End for
3. For each $N = \{N_1, N_2, \ldots.N_p\}$,
   repeat from step 2.1.1
   End For
4. When resources availability is insufficient,
   For each $B = \{Bi, i = 1,2,\ldots r\}$
   4.1.1 $\eta_A = \eta_A + \eta_B$, Repeat from step2.1.1

The algorithm is explained as below.

In the Admission Control Algorithm, we consider *n* number of user requests. Initially, the received signal strength (RSS) is calculated and when this RSSv 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.

Now, we consider 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, we 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.
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. Simulation Results

In this section, we simulate the proposed channel based efficient CAC (CBECAC) scheme using Network simulator (NS2) [12] which is a general-purpose simulation tool that provides discrete event simulation of user defined networks. We have used the LTE/SAE implementation model for NS2 [11].
The simulation parameters are given in table 1.

No. of Servers	1
No of aGw	1
No of eNB	1
No. of UEs	5
Traffic Types	CBR and VBR
Traffic Rate	150 to 300 kb

Table 1: Simulation Parameters

In the simulation settings, we have one server to provide HTTP, FTP and signaling services, one aGW to provide HTTP cache and flow control, one eNB to provide flow control information and five UEs. The simulation topology is given in the following figure 1.

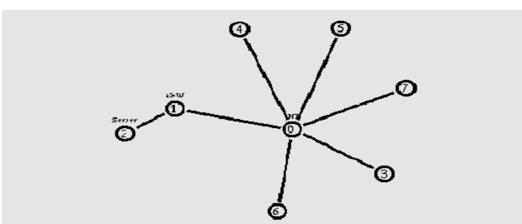

Figure 1: Simulation Topology

In this model, ULAirQueue is used for uplink flows in the link between UE and eNB. For the downlink flow, (ie) in the link between eNB and UE, DLAirQueue is used. For both the links, the link bandwidth is set as 500kb and link delay as 2ms.
For the link between eNB and aGw, ULS1Queue is used and for the downlink between aGW and eNB, DLS1Queue is used. For both the links, the link bandwidth is set as 5Mb and link delay as 2ms.For the link between the server and aGW, a simple DropTail queue is used with link bandwidth as 50Mb and link delay as 2ms.In the model, DLAirQueue provides each flow and each cell's information, such as buffer size and average data rate; DLS1Queue uses these information and the current packet size to decide whether the packet is allowed to be sent to DLAirQueue. If the buffer of the flow or the cell where the flow locates is going to be overflow, the packet is blocked until both the flow's buffer and cell's buffer is not overflow.
We measure the received bandwidth in Mb/s and throughput in terms of number of packets and end-to-end delay for both VBR and CBR traffic flows
We compare the proposed channel based efficient CAC (CBECAC) scheme with the Resource Block allocation based CAC (RBCAC) scheme [7].

Case 1 (UDP) :

A. **Based on UE**

In this experiment, we vary the number of UEs from 1 to 5 in order to measure the received bandwidth, throughput and delay for the UDP non-real time traffic.

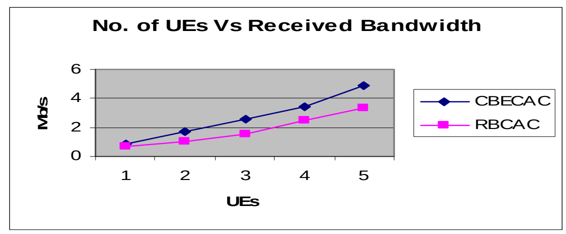

Fig 2: Number Vs Bandwidth

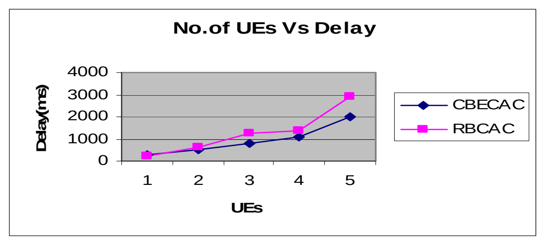

Fig 3: Number Vs Delay

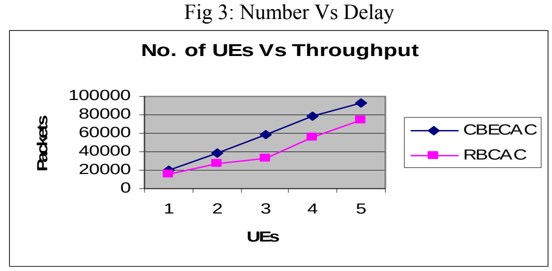

Fig 4: Number Vs Throughput

Figure 4 shows the throughput obtained with the CBECAC and RBCAC schemes. From the figure, it can be seen that, the throughput of both schemes are 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.
It can be seen from Figure 2, the received bandwidth gradually increases when the number of users is increased. It shows that CBECAC better than CBECAC.

B. **Based on Rate**

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

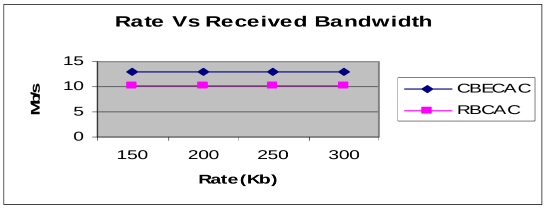

Fig 5: Rate Vs Bandwidth

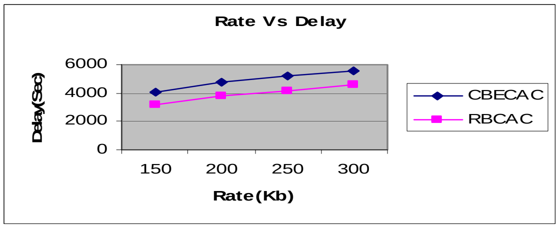

Fig 6: Rate Vs Delay

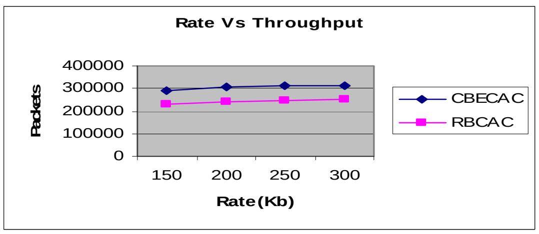

Fig 7: Rate Vs Throughput

5. Conclusion

In this paper, we have 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 new call (NC) request and handoff call (HC) request and the type of services are classified as VoIP and Video. We prioritize HC over NC and VoIP over Video type. Then based upon the received signal strength (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. When a call request arrives to the network, it is checked for HC or NC. Then their class is checked whether it is a VoIP call or a video call. Then the bandwidth is reserved for each call based upon the traffic density. 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 our simulation results we have proved that this admission control algorithm provides channel quality and prioritizes the handover calls over new calls.

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