

# Call Admission Control in 3GPP LTE Systems at High Vehicular Communications

Rajeshkumar RAMRAJ, Daryoush HABIBI, Iftekhar AHMAD

**Abstract** — In high-speed wireless communication, call dropping and blocking probability plays an important factor in providing connection-level Quality of Service (QoS). The main goal of this research is to develop an admission control algorithm to reduce call block and call drop. These parameters are mostly affected due to insufficient resources in the target cell. Radio resources management (RRM) plays a major role in providing better QoS in high speed vehicular communication. In this paper, we presents a novel CAC algorithm for high vehicular speed communication systems where the UE (mobiles) contain real time constant bit rate (rt-CBR) and non-real time variable bit rate (nrt-VBR) and the performance for these two traffic conditions are compared at different vehicular speeds.

**Index Terms**— Call Admission Control (CAC), 3GPP LTE, Call Dropping, Call Blocking, Channel Utilization, Bit Error Rate, Vehicular Communication.

## 1. INTRODUCTION

Long Term Evolution (LTE) of Universal Mobile Telecommunication System (UMTS) has become one of the advanced series in mobile telecommunication industries. In 1947, Bell labs, USA started the development of cell concept in mobile communications to increase the capacity of mobile networks by dividing the coverage area into smaller areas with its own base station, operating in different frequency. LTE or the Evolved Universal Terrestrial Access Network (E-UTRAN) which is a part of Evolved Packet Access (EPS), specified by Third Generation Partnership Project (3GPP) has continued dominating the cellular communication industries and progressed its development towards Fourth Generation (4G) networks. Considering the use of multimedia applications like, online mobile games, video calls, video chatting, online video streaming, mobile TV etc. in mobile communications, the telecommunication industry needs to satisfy the requirements (data rates) of mobile users. These requirements succeeded by the next generation wireless technology such as LTE, which is marketed as 4G LTE, as per 3GPP standard and LTE can operate at high-speed mobile communications.

LTE system is based on Global System for Mobile Communications / Enhanced Data Rate for GSM Evolution (GSM/EDGE), Universal Mobile Telecommunication System / High Speed Packet Access (UMTS/HSPA) technologies and some of the important requirements of this technology are high spectral efficiency, high peak data rates, and short round trip time and frequency flexibility [1]. LTE designed to be an IP-centric network that can provide QoS. LTE consists of enhanced nodeBs (eNBs) in Radio Access Network (RAN), Mobility Management Entities (MMEs) and Serving Gateways (S-GW) in the core. X2 interface is used to interconnect eNBs in RAN and S1 interface to connect MMEs and S-GWs at the core (figure 1).

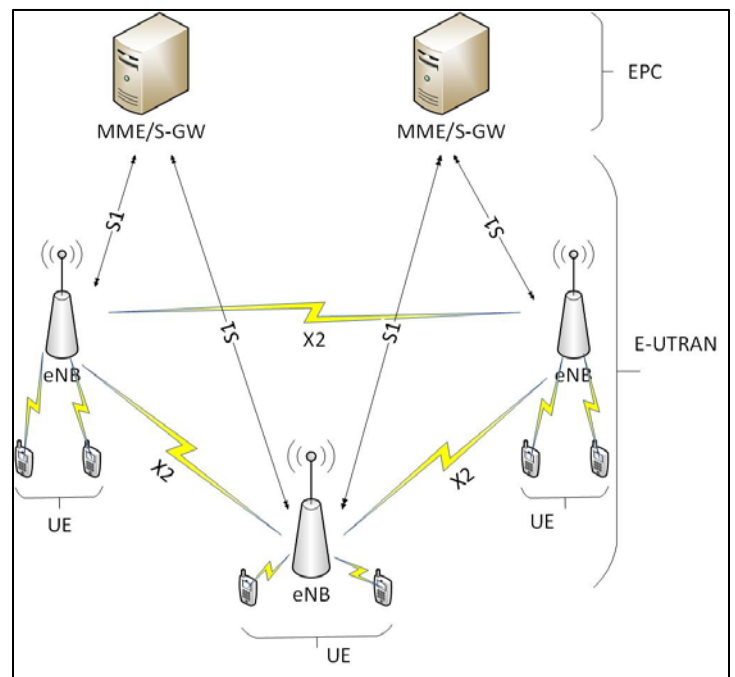


Fig 1. A Simple LTE Architecture

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LTE supports peak download rates up to 300 Mbps and up-link rates up to 75 Mbps depending on the user equipment (UE) category by adopting Orthogonal Frequency Division Multiple Access (OFDMA) and Multiple-Input and Multiple-Output (MIMO) systems. LTE supports both Time Division Duplex (TDD) and Frequency Division Duplex (FDD) with same RAN and supports different spectrum flexibility with at least 200 active UE in every 5 MHz cell. Furthermore, LTE targets for smooth evolution and co-existence with legacy standards like GSM/EDGE, UMTS and CDMA2000, table 1 gives you the key specifications of 3GPP LTE. In the recent development of mobile communication industry, LTE technology plays a vital role and leads the industry towards the next generation networks. Even though the previous standards had some good development, researchers throughout the world are working on the current technology where the upcoming mobile wireless infrastructures are considered to support high-speed services with different QoS. Hence, some of the key aspects like multimedia services with real-time QoS guarantees are expected to improve further. To achieve this goal in multimedia services, this paper focuses on improving wireless transmission [2]. This can be achieved in many ways and here we aim to develop a Call Admission Control (CAC) algorithm at vehicular speeds in LTE communication systems. In our simulation, we are considering the scenario (figure 3) where we are comparing real time-constant bit rate (rt-CBR) traffic against non-real time variable bit rate (nrt-VBR) traffic. BER and high data rate will have significant impact on UEs QoS in non-real time traffic (ftp data, web browsing) condition. Radio Resource Management (RRM) handles the BER guarantee, by adaptive power allocation. In real time traffic condition (VoIP, live online video streaming), in addition to the QoS for non-real time traffic, delay is another important factor that need to be taken into account. When it comes to multimedia service, delay should be given high importance in terms of QoS [3, 4] and bandwidth management in admission control framework [5].

The rest of the paper is organised as follows, section 2 will talk about the pre-study on call admission control, section 3 will

talk about call admission control in high vehicular speeds, and section 3 will talk about our proposed call admission control algorithm and strategy, followed by results and discussion and conclusion in section 5 and 6 respectively.

## 2. CALL ADMISSION CONTROL – A PRE-STUDY

The wireless network for satisfying multimedia services must include a resourceful CAC, which is a process of regulating traffic in Voice over Internet Protocol (VoIP). This CAC will ensure the quality and performance level of audio and voice. CAC uses the strategy to limit the number of call connections to the network, so that the network congestion and call dropping is reduced. CAC is the process of regulating traffic volume in voice communications, mainly in wireless mobile communication networks and in VoIP protocol. The objective of this CAC is to maintain a certain level of audio and voice quality in the communication network and to provide a desired QoS [6-9]. Ahmad discusses some of the main reason for CAC to have a guarantee QoS like improving signal quality, reducing call dropping and increasing the packet level performance. The author also discusses the CAC classification with their pros and cons [10]. There are number of factors that the signal quality of a UE relies on, such as base station distance, path loss, fading and noise. Motorola has proved in [11] that, AMC uses link adaption method to raise the overall system capacity [12]. CAC algorithm uses a key technology called Adaptive Modulation Coding (AMC) to optimize spectrum utilization [13]. In order to satisfy the QoS and utilize the available resources efficiently in vehicular communication, the call-level and packet-level service requirements should guarantee an effective communication. A simple adaptive modulation technique used in LTE network in shown in figure 2. Link adaption technology is widely used in most wireless technology for an OFDMA downlink systems. Zuh et al, proposes a new CAC algorithm for OFDMA systems, where the link adaption technique is implemented to provide better QoS in a cell considering the probability distribution of the incoming traffic to determine the resource utilization [14]

CAC algorithm is of two types, static and dynamic CAC. Static CAC deals with resource allocation for hand-off calls [15, 16] and dynamic CAC deals with the radio channel status and available resources to perform the admission control [17, 18]. There are several research methods identified for new and hand-off request, Zarai et al, addressed on a scheme where the load balance is taken into account to reduce the hand-off dropping. The author also proposes a CAC algorithm where the RB allocation is considered and it prioritises the hand-off calls over the new calls [19]. Bae et al, propose a resource-estimated CAC algorithm for a guarantee QoS and packet delay [9]. In order to provide better QoS, RRM is very important in CAC, Based on the resources available, the RRM will decide whether to accept or to drop the call. In [20] a delay aware CAC is proposed which guarantee a certain level of packet delay and prevents congestion in the base station. The traditional guard channel scheme has been the foundation to reserve a fixed number of RBs and this was proved by Ramjee et al. in [21], where he also defines

TABLE 1  
 3GPP Key Features

Frequency Range	TDD and FDD Bands (UMTS)					
Channel BW	1.4 MHz	3 MHz	5 MHz	10 MHz	15 MHz	20 MHz
1RB = 180 KHz	6 RBs	15 RBs	25 RBs	50 RBs	75 RBs	100 RBs
Modulation Scheme	Downlink: QPSK, 16QAM, 64QAM Uplink: QPSK, 16QAM, 64QAM					
MIMO Technology	Downlink: Single User-MIMO, transmit diversity, spatial diversity and cyclic delay diversity Uplink: Multi User-MIMO					
Multiple-Access	Downlink: OFDMA Uplink: SC-FDMA					
Data Rates	Downlink (Peak): 150 Mbps (UE category 4, 2x2 MIMO, 20 MHz) 300 Mbps (UE category 5, 4x4 MIMO, 20 MHz) Uplink (Peak): 75 Mbps (20 MHz)					

the linear objective functions for dropping and blocking probabilities. Angelos et al. propose a dynamic CAC mechanism considering the issues of busy hour and the bandwidth allocation and request mechanism like Grant per Connection (GP) and Grant per Subscriber Station (PST) were discussed [22].

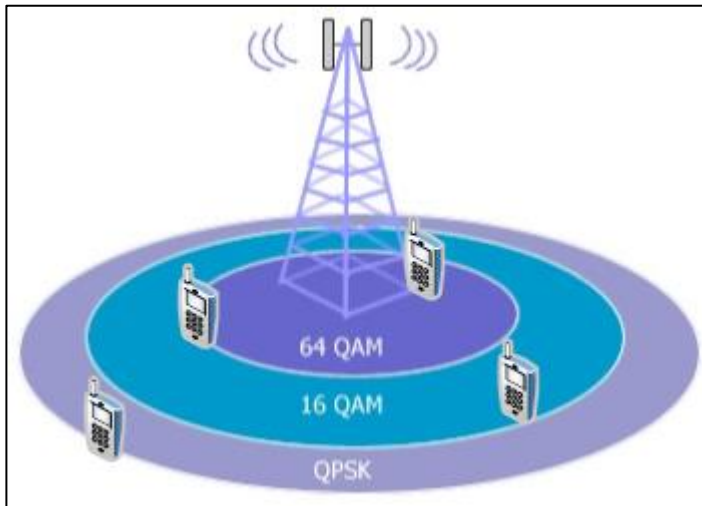


Fig 2. Adaptive Modulation in LTE

### 3. CALL ADMISSION CONTROL IN HIGH VEHICULAR SPEED

LTE offers a high data rate of 300 and 75 Mbps downlink and uplink capacity up to release 10. Release 11 has been considered as LTE-A with 1 Gbps and 300 Mbps for downlink and uplink [13, 23]. In reality due to line of sight (LOS) issues, the recommended target is difficult to achieve. However researchers attempt to prove their work with the help of simulators. LTE communication standard incorporates OFDMA for downlink and SC-FDMA for uplink to achieve better QoS and spectral efficiency. OFDMA extends system support by adopting PHY architecture and implementing the scalability factors from 1.5 MHz to 20 MHz. LTE anticipates the bandwidth spectrum range from 1.4 MHz to 20 MHz and this choice of SC-FDMA in uplink places the complex, power hungry operation of frequency domain [24]. During high vehicular speeds due to the rapid variation of time and frequency, the relative motion between transmitters and receivers reacts with very high Doppler shift. In such conditions, it causes signals to overlap and signal interference. This is widely addressed by various researchers as a fading problem and many solutions are provided at various simulation conditions. Here in this paper we are going to address such kind of problem with a different simulating scenario. A simulation scenario used in this research with real time LTE communication system shown in figure 3.

The classical Doppler spectrum derived mathematically for a sin wave is given below.

$$S(f) = \begin{cases} \frac{A}{\sqrt{1 - (f/f_d)^2}}, & |f| < f_d \\ 0, & |f| > f_d \end{cases} \quad (1)$$

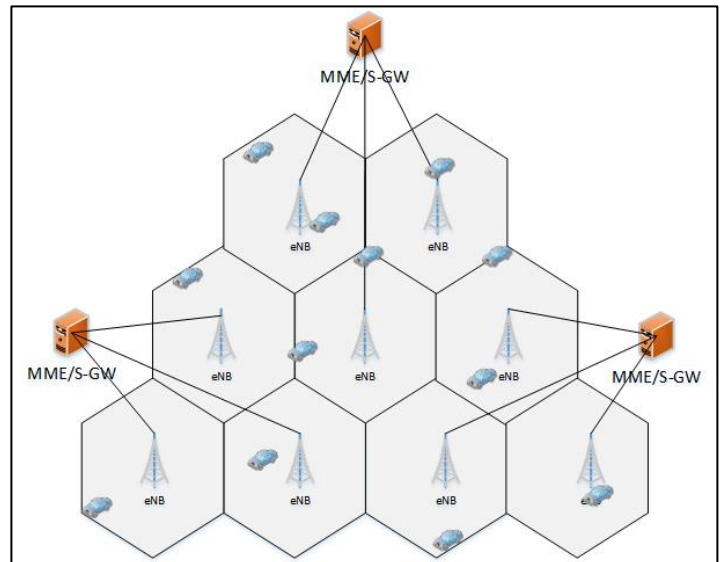


Fig 3. A Simple Simulation Scenario

Fading problem causes bit error rate (BER) issues at the receiver side, therefore BER probability increases at high vehicular speed resulting unused bandwidth. OFDMA reduces inter-symbol interference (ISI) by utilizing guard channel. Due to the complexity at the receiver during transmission, SC-FDMA, chosen as a medium access for uplink in LTE [25]. The ISI is widely measured by "rms delay",  $\tau_{rms}$  seconds, a magnitude of components and the time difference. The maximum delay,  $\tau_{max}$  seconds, is the difference between shortest and the longest path. If the greatest path difference is  $D_{max}$  km, then  $\tau_{max} = D_{max}/0.3\mu s$ .  $D_{max}$  depends on PHY characteristics and cell coverage area [24]. When it comes to BER estimation, Rayleigh fading is proved as a first-rate model which can rival error rate in the radio signal. Abdelhadi et al. proposed a unified approach to derive BER for mobile WiMAX in uplink direction. The analytical expression was derived and the bit rate was obtained for BPSK, QPSK and 16QAM in adaptive modulation [26]. Gangane et al. in his paper, studied about the influence of LOS on SC-FDMA uplink BER behaviour for Rayleigh fading [27].

CAC in high vehicular communication systems is an important factor in providing better QoS. In our scenario, consider we have a eNBs controlled by MME/S-GW, and UE which is at "time  $t$ " (current time) moving at a initial speed of  $v_i$  and a maximum speed of  $v_{max}$ . Consider we have a total usable (available) bandwidth of  $B_{total}$  in Mbps, when the eNB receives a request from the UE with a demand of  $B_{req}$  in Mbps, the CAC scheme should satisfy the below condition.

$$U_{cap} \geq B_{req} + B_{total} \quad (2)$$

$U_{cap}$  is the usable capacity of the UE and it varies according to the speed. When the UEs increases its speed the  $U_{cap}$  drops and the dropping rate increases significantly resulting in QoS degradation.

#### 4. PROPOSED SCHEME

##### 4.1. Bit Error Rate Estimation at High Vehicular Speed

In the proposed CAC scheme, our initial step is to estimate BER using Rayleigh fading model. Based on this the CAC decision is made. For high speed vehicular communication, error control mechanism like Forward Error Control (FEC) is used as they use extra parity bits to recover the corrupted bits and with this we can avoid re transmission of packets in wireless channels. At high vehicular speeds the rate of transmission goes down rapidly and this creates very high impact on effective throughput. To overcome this issue we use a FEC mechanism in OFDMA DL to avoid relatively high throughput. In LTE communication channels, we have parameters illustrated as follows:

We have  $N$  as the number of OFDMA sub-carriers,  $f_D$  as maximum Doppler frequency,  $V_{UE}$  as the speed of the UE,  $C$  as the speed of the wavelength (light), and  $f_c$  as carrier frequency [28].

Now we have the Doppler frequency as,

$$f_D = f_c \left( \frac{V_{UE}}{C} \right) \quad (3)$$

We have error probability of various forms of PSK signals, BPSK transmits 1 bits /symbol, so energy  $E_n = E_b$ , where  $E_b$  is the bit energy. Therefore, error probability is given by,

$$P_b(\gamma_b) = Q(\sqrt{2\gamma_b}) \quad (4)$$

Where  $\gamma_b$  is defined as received bit energy-to-noise ratio and  $\bar{\gamma}_b$  is defined as average received bit energy-to-noise ratio. The below mathematical calculation will derive equation (4) to calculate the error probability.

In a cell area with  $x$  and  $y$  axis, the QPSK signal can be rotated at  $45^\circ$  without changing the error probability due to rotation invariance property. In such cases the decision boundaries corresponds to real and imaginary axis of complex signals. In OFDMA DL air interface, inter-carrier interference causes main issue for BER; in order to overcome this, Additive white Gaussian noise (AWGN) is used. Considering AWGN, the noise vectors in a cell area is  $\tilde{n} = \tilde{n}_I + j\tilde{n}_Q$  where  $\tilde{n}_I$  and  $\tilde{n}_Q$  are independent Gaussian variables with variance  $N_0$ . Considering this the probability symbol error  $P_M$  is given as,

$$P_M = 1 - \left( 1 - Q(\sqrt{\gamma_s}) \right)^2 \quad (5)$$

Where  $\gamma_s$  is defined as the received symbol energy-to-noise ratio and  $\bar{\gamma}_s$  is defined as the average symbol energy-to-noise ratio. Let  $P_b$  be the probability of bit error, then we have,

$$P_M = 1 - (1 - P_b)^2 \quad (6)$$

From equation (5) and (6) we can derive the bit error for 1 bits/symbol  $P_b$

$$P_b = Q(\sqrt{\gamma_s}) \quad (7)$$

When QPSK transmits 2 bits/symbol, symbol energy is  $E_h = 2E_b$   $E_b$  is bit energy and  $\gamma_s = 2\gamma_b$  which will derive equation (4).

When we consider bit verses symbol error probability, the probability decision error  $P[e]$  otherwise known as  $P_M$  is derived in equation (5). Though we are interested in error probability  $P_b$ , and the data symbols corresponds to  $\log_2 M$  data bits, we have,

$$\frac{P_M}{\log_2 M} \leq P_b \leq P_M \quad (8)$$

Hence,  $P_b$  and  $P_M$  relate each other, we have:

$$P_b \approx \frac{P_M}{\log_2 M} \quad (9)$$

When we calculate the error probability for Rayleigh fading, the received bit and symbol energy-to-noise ratios  $\gamma_b$  and  $\gamma_s$  have exponential PDFs, therefore we have,

$$P_{\gamma_b}(x) = \frac{1}{\bar{\gamma}_b} e^{-\frac{x}{\bar{\gamma}_b}}, x \geq 0 \quad (10)$$

$$P_{\gamma_s}(x) = \frac{1}{\bar{\gamma}_s} e^{-\frac{x}{\bar{\gamma}_s}}, x \geq 0 \quad (11)$$

Since there are  $\log_2 M$  bits/modulated symbol, we have:

$$\gamma_s = \gamma_b \log_2 M \quad (12)$$

$$\bar{\gamma}_s = \bar{\gamma}_b \log_2 M \quad (13)$$

For such channels the error probability is given as,

$$P_b = \int_0^\infty Q(\sqrt{2x}) P_{\gamma_b}(x) dx \quad (14)$$

$$P_b = \int_0^\infty P_b(x) P_{\gamma_b}(x) dx \quad (15)$$

$$= \frac{1}{2} \left( 1 - \sqrt{\frac{\bar{\gamma}_b}{1 + \bar{\gamma}_b}} \right) \quad (16)$$

The above BPSK and QPSK bit error probability when plotted for an AWGN channel and a Rayleigh fading channel, it will result in a huge loss of performance [28]. Therefore for M-PSK the average symbol error probability is given as:

$$P_s = \int_0^\infty P_M(x) P_{\gamma_s}(x) dx \quad (17)$$

Now with the above equation, the average received symbol energy-to-noise ratio is given as:

$$\bar{\gamma}_s = \frac{1}{1 - \frac{1}{N^2} [N + 2 \sum_{i=1}^{N-1} (N-i) J_0(2\pi f_D T_s^g i)] + \frac{NT_s}{\left(\frac{E_s}{N_0}\right)}} \quad (18)$$



The average received bit energy-to-noise ratio  $\tilde{\gamma}_b$  can be derived from equation (13), therefore we have the following:

$$\tilde{\gamma}_b = \frac{\tilde{\gamma}_s}{\log_2 M} \quad (19)$$

Now  $\tilde{\gamma}_b$  is expressed as,

$$\tilde{\gamma}_b = \frac{1}{\log_2 M} \left( 1 - \frac{1}{N^2} [N + 2 \sum_{i=1}^{N-1} (N-i) J_0(2\pi f_D T_s^g i)] + \frac{NT_s}{\log_2 M} \left( \frac{1}{N_0} \right) \right) \quad (20)$$

The above average received bit energy-to-noise ratio  $\tilde{\gamma}_b$  and average received symbol energy-to-noise ratio  $\tilde{\gamma}_s$  is verified in order to examine ICI and the Bessel function  $J_0(.)$  order zero is used to estimate the symbol error for  $M$ -ary QAM. And finally for the value  $M=2$  and  $M=4$ , the error probability  $P_b$  can be derived from the equation (15). With higher order of  $M$  and with a speed factor  $v$ , the error probability can be calculated using the expression below,

$$P_b(v) = \frac{P_M(v)}{\log_2 M} \quad (21)$$

In OFDMA with a low  $\tilde{\gamma}_b$  AWGN dominates the performance and with the extra noise due to ICI has little effect. At the same time high  $\tilde{\gamma}_b$ , the ICI dominates the performance and cause an error floor [28].

#### 4.2 Call admission Control Strategy and Throughput Estimation

The call admission control strategy and its decision is based on the throughput estimation, consider the simulation scenario from figure 3, say we have a UE moving at a speed of 120 km/hr, speed can be represented as  $v_{max}$ , with  $M$ -ary the symbol and bit error probability  $P_s, P_b$  is represented as,

$$P_s(v_{max})^M = 1 - (P_b(v_{max}))^M \quad (22)$$

As we are going to compare the real time and non-real time traffic conditions in our scenario, let us assume that the traffic generation rate from the traffic source is  $g$  bits per second and the traffic arrival rate in the cell area is  $i$ , therefore the mean traffic arrival  $\varphi_i$  is expressed as:

$$\varphi_i = \tilde{n}_i g \quad (23)$$

Where  $\tilde{n}_i$  is the number of UEs in the cell area, now let us consider the UE in the cell is sending a request for  $N$  number of RBs which contains  $Y$  symbols, the corrupted or ruined symbols  $Z$  can be calculated as follows:

$$Z_{max} = Y \times P_s(v_{max}) \quad (24)$$

When the UE receives the requested RBs from the header  $h$ , it is very important to check for an overhead rate ( $H_{over}$ ) per bit received and this can be expressed as:

$$H_{over} = h + \left( \frac{K(v_{max})}{Y} \right) \quad (25)$$

Parameters	Values
Number of cells	3, 7, 9
Number of UEs	6, 12, 24
Radius	250m (hexagonal shape)
Speed	120km/hr
Queue length	10mb
Traffic type	rt-CRB (VoIP, live video streaming), nrt-VBR (web browsing, ftp data)
Traffic rate	128kbps - 512kbps
Bandwidth	20 MHz

Table 2: Simulation Parameters

Where,  $K$  is the parity bit recovered through forward error correction (FEC).

The CAC algorithm used in this paper is given below:

- When the UE is active in a cell and its speed and direction is calculated by the eNB. The eNB provides the bandwidth information to all the active UE in the cell area. The average bit error rate probability  $P_b(v_{max})$  and average symbol error probability  $P_s(v_{max})$  at the maximum speed is estimated at this stage.
- UE will now send a request to the eNB asking  $N$  number of RBs to answer the call.
- Now the eNB will calculate the requested RBs with the available RBs, based on this the decision will be made and the required RBs will be allocated to the UE.
- However, if there are insufficient RBs, then the remaining available RBs will be reserved.
- eNB will now receive a new request from a UE and both the reserved RBs and allocated RBs will be subtracted from the total cell capacity to allocated the new required RBs
- If the requested RBs is less than or equal to the total available RBs, then the bandwidth is allocated. This also should satisfy the condition specified in equation (2)
- Otherwise the call is rejected.

#### Algorithm 1: Proposed CAC Algorithm

```

initialization;
gather UE information;
compute total available RBs;
verify call type;
if  $U_{cap} \geq B_{req} + B_{total}$  then
| accept call;
else
| drop call and go begin the process again;
end
    
```

### 5. SIMULATION RESULTS AND DISCUSSION

The simulator used for this research is NS2, and the simple simulation scenario is given in figure 3. The simulation parameters were given in Table 2. The maximum bandwidth for the LTE channel was 20 MHz with data rate capacity of 20 Mbps. The queue length is 10mb and traffic rate was distributed in the range 128 kbps to 512kbps. In the simulation, we are considering two different types of traffic, first as real time CBR and second as non-real time VBR. The simulation was conducted at a speed of 120km/hr, finally at the end of the simulation call blocking rate, dropping rate and channel utilization during the call duration was calculated and analyzed for above mentioned traffic conditions.

The simulation results are evaluated as per the following performance evaluation metrics:

- Estimating the bit error rate probability  $P_b$  from equation (4).
- Estimating the amount of available RBs before bandwidth allocation.
- Evaluate the type of call based on the traffic condition.
- Estimate the amount of available RBs after resource allocation.
- Physical RBs utilization, (i.e.), number of allocated RBs for the UE during the whole simulation time.

Considering the scenario, when the UE starts to move at a speed of 120km/hr, the corresponding eNB will get all the relevant information such as speed, UE direction, position etc. and the eNB will compute the available BW ( $U_{cap}$ ) every minute. Now the UE will send a request ( $B_{req}$ ) to BS and the eNB will check for the condition given in the equation (2). If the above conditions satisfy, the eNB will allocate the requested RBs. At the same time, the eNB will be checking the call type and traffic, because the eNB will consider only request which follows the CAC algorithm. Here in this simulation, the speed factor plays an important role in estimating the error probability ( $P_b$ ) and throughput estimation given in section 4.2. When the UE moves at a maximum speed  $v_{max}$  the channel usable capacity is calculated based on the estimated throughput at the maximum speed  $v_{max}$  and the CAC decision is made as follows:

$$U_{cap}(v_{max}) \geq B_{req}(v_{max}) + B_{total}(v_{max}) \quad (26)$$

The simulation will be scheduled to perform in the cell area where the UE is active and the call blocking rate, call dropping rate channel utilization is calculated. When the UE moves from one cell to another cell (adjacent cell), the packet generation is continued and the correspondence is started in the hand-off in time and when the UE connects to the next eNB and that is the hand-off out time, and here the delay is calculated. In RRM of CAC algorithm,  $B_{req}$  is decided based on the service type and traffic.

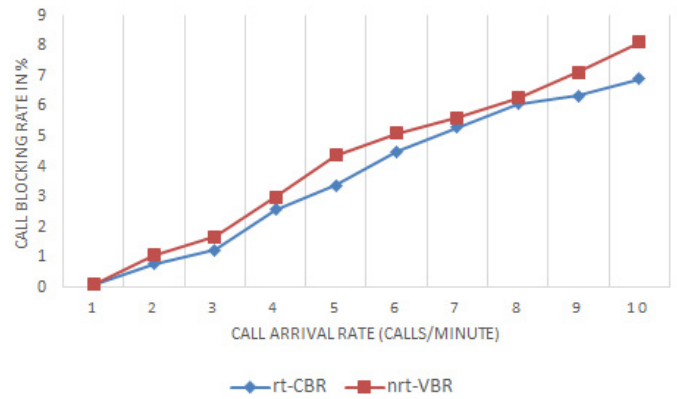


Fig 4. Call Blocking Rate when the UEs at 120km/hr Speed

Figure 4 represents the call blocking rate, x-axis shows the number of call arrival per minute and y-axis shows the call blocking rate in %. The simulation is analyzed for the above mentioned two traffic types and the results shows the call blocking rate for rt-CBR is less compared to nrt-VBR. The simulation results shows the blocking percentage for rt-CBR is around 6.8% for 10 calls per minute and 8% for nrt-VBR for 10 calls per minute. As explained in [3, 4] CBR traffic has a regular and constant traffic flow compared to VBR, where the traffic conditions have high impact on user condition.

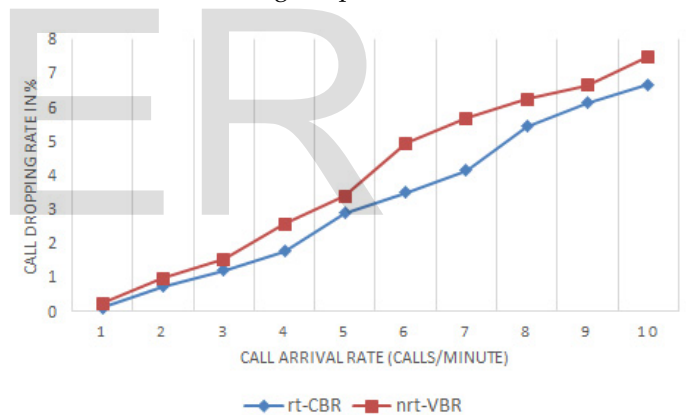


Fig 5. Call Dropping Rate when the UEs at Speed 120km/hr

Figure 5 represents the call drop ration for the proposed CAC algorithm, where the percentage of call drop for real time traffic is less than the non-real time traffic. Call drop percentage for rt-CBR is around 6.8% and 7.5% for nrt-VBR, this was calculated at 10 calls per minutes with maximum of 24 UEs. In rt-CBR traffic, after the requested the RBs allocated, the RRC connection is established with MME and UE. The connection session is smooth because of the constant flow in the traffic, where the connection between the MME and UE drops in nrt-VBR, where the traffic flow is not stable.

Fig 6. Channel Utilization

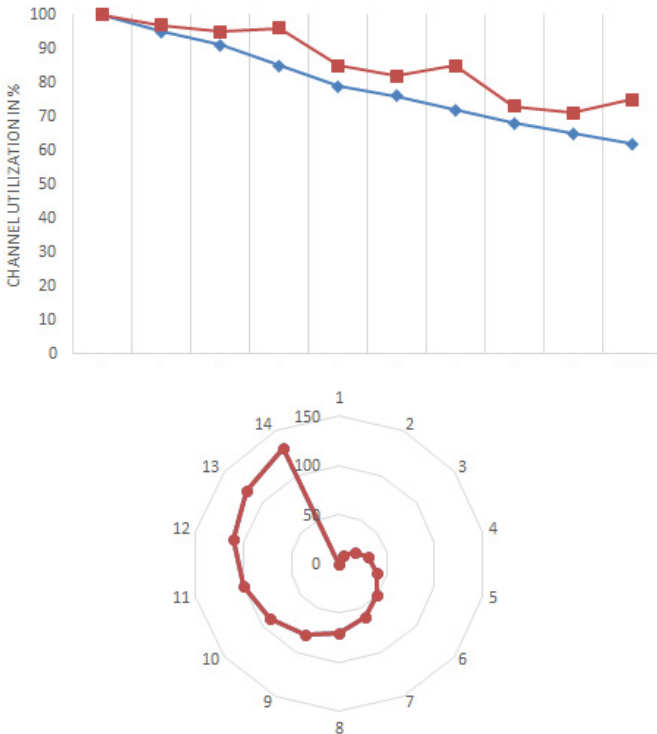


Figure 6 shows the channel utilization in both traffic conditions, which confirms the better bandwidth utilization of rt-CBR, where the flow of data rate is at a constant speed when compared to nrt-VBR, where the data rate differs on users usability. Channel utilization for rt-CBR is calculated as 62% and the utilization for nrt-VBR varies due to variable bit rate. We should notice the utilization ratio reduces gradually for real time traffic and for non-real time traffic, the utilization curve varies according to the variable bit rate. RRM is the primary reason for the CAC RBs reservation by adjusting the attribution of resources intelligently.

Fig 7. UEs Speed Vs Simulation Time

Figure 7 shows the UEs position and speed at various simulation time. When the UE increases its speed, and moves away from the eNB, the usable capacity of the UE drops significantly which will affect the drop and block probability and other QoS factors.

## 6. CONCLUSION

This research developed a new CAC algorithm which reduces the call blocking, call dropping rates and improved channel utilization. The proposed algorithm based on RBs reservation for current and ongoing calls, and new calls. The eNodeB will calculate the available and total RBs from the available information and the eNB manages the RB allocation. We evaluate the call drop probability and call block probability via the proposed CAC scheme and simulation. The future

work will target on cell management and resource allocation for high-speed wireless vehicular communications.

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