A implementation of call switching over volte networks using dynamic initiation of voice request

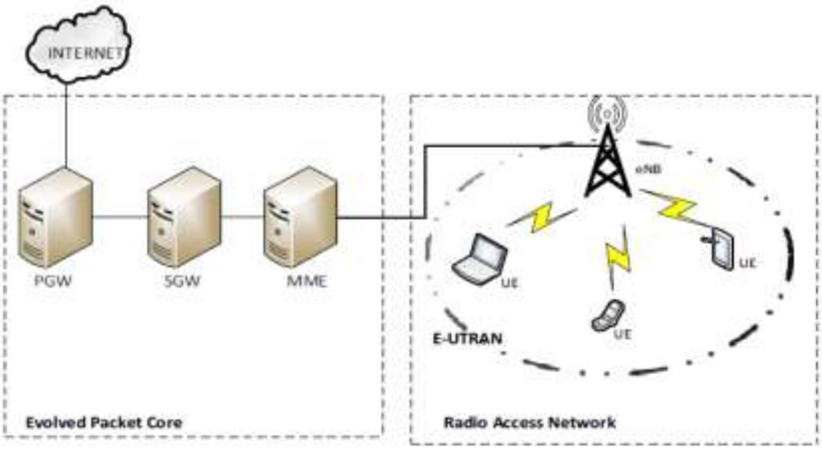
[1] Bankapalli Goutham, [2] Prof. Kuda Nageswara Rao
[1]M.Tech Scholar, Department of Computer Science and System Engineering,
Andhra University College of Engineering (A), Visakhapatnam, AP, India.
[2]Department of Computer Science and System Engineering,
Andhra University College of Engineering (A), Visakhapatnam, AP, India

***Abstract**:* Provisioning of Quality of Service (QoS) is a key issue in any multi-media system. However, in wireless systems, supporting QoS requirements of different traffic types is more challenging due to the need to minimize two performance metrics - the probability of dropping a handover call and the probability of blocking a new call. Since QoS requirements are not as stringent for non-real-time traffic types, as opposed to real-time traffic, more calls can be accommodated by releasing some bandwidth from the already admitted non-real-time traffic calls. Long Term Evolution (LTE) network standard defines requirements to guarantee Quality of Service (QoS) for diverse applications such as VoIP, video and web browsing according to the Third Generation Partnership Project (3GPP) specifications. The Radio Resource Management (RRM) techniques such as Call Admission Control Schemes play an important role in providing such guarantees. Consequently, several schemes have been proposed to manage resources while ensuring QoS to wireless applications. This paper presents a survey of Call Admission Control (CAC) Schemes. These algorithms are classified into CAC with Pre-emption, Resource Reservation (RR), Resource Degradation (RD), Delay Awareness (DA) or Channel Awareness (CA). The operational procedure, strengths and weaknesses of each scheme are discussed. If we require that such a released bandwidth to accept a handover call ought to be larger than the bandwidth to accept a new call, then the resulting probability of dropping a handover call will be smaller than the probability of blocking a new call. In this paper we propose an efficient Call Admission Control (CAC) that relies on adaptive multi-level bandwidth-allocation scheme for non-real-time calls. The scheme allows reduction of the call dropping probability along with increase of the bandwidth utilization. The numerical results show that the proposed scheme is capable of attaining negligible handover call dropping probability without sacrificing bandwidth utilization. In addition, we proposed reservation and degradation approach to admit many users when there is a limited number of bandwidth, which also achieved effective utilization of network resources. Simulation results show that the proposed scheme significantly outperforms the reservation-based scheme and bandwidth degradation schemes in terms of admitting many calls and guaranteeing QoS to all the traffic types in the network. Numerical results imitate to experimental results with in significant differences.

Index Terms – **Adaptive bandwidth allocation, Quality of Service, multi-class services, multi-class traffic, call dropping probability, call blocking probability, call admission control, CAC, LTE Networks, Call Admission Control, Radio Resource Management,**

I. INTRODUCTION

The growing demand in network applications such as VoIP, Video, Web browsing e. t. c with different Quality of Service (QoS) requirements poses a great challenge to the wireless networks. Report according to Cisco indicates that the demand of network applications has grown exponentially and will continue to increase by 1000 times in the next five years .The 3GPP introduced the LTE networks as one of the solution to the challenge. [1]The network provides higher data rate, low latency, scalable bandwidth, mobility and extended coverage. The LTE network adopts Orthogonal Frequency Division Multiple Access (OFDMA) for downlink transmissions. It adopts a scalable radio resource bandwidth of 1.4 MHz to 20 MHz. This radio resource bandwidth is divided into equal sub-channels of 180 KHz each in frequency domain and a Transmission Time Interval (TTI) of 1ms each in time domain. [2] A TTI comprises of two time slots of 0.5 ms each. Thus, a radio resource in time/frequency domain across one time slot in time domain and one sub-channel in frequency domain is termed a Resource Block (RB). A RB is the smallest unit of radio resource that can be allocated to a User Equipment (UE) for data transmission .The QoS adaptability of some multimedia traffic types has been used by several schemes to reduce the call blocking probability. The adaptive QoS schemes proved more flexible and efficient in guaranteeing QoS than the guard channel schemes. D. D. Vergados et al. proposed an adaptive resource allocation scheme to prioritize particular traffic classes over others. Their scheme is based on the QoS degradation of low priority traffic to accept higher priority traffic call requests. W. Zhuang et al. proposed an adaptive QoS (AQoS) scheme which reduces the QoS levels of calls that carry adaptive traffic to accept the handover call requests. F. A. Cruz-Pérez et al.[3] proposed flexible resource-allocation (FRA) strategies that prioritizes the QoS of particular service types over the others. Their scheme releases bandwidth from the low priority calls based on the prioritized call degradation policy to accept the higher priority call requests.. A cell can borrow channels from any neighboring cell to reduce the call blocking probability. To ensure QoS for diverse network applications in LTE networks, Radio Resource Management (RRM) such as CAC schemes is of great importance[5]. CAC schemes admit or block call requests(new or handoff) and maintain required QoS while circumventing possible congestions hence the scheme is highly needed. Therefore, several schemes have been proposed to manage call requests while ensuring QoS to wireless application. The CAC schemes is presented. The schemes are classified into CAC with Pre-emption, Resource Reservation (RR), Resource Degradation (RD), Delay Awareness (DA) or Channel Awareness (CA). The operational procedures, strengths and weaknesses of each scheme are highlighted. The comparative analysis of these schemes is also discussed in order to provide open research issues for future direction. we propose a novel CAC scheme named, An Adaptive Call Admission Control with Bandwidth Reservation for Downlink LTE Networks to amend the inefficiency of Reservation-Based. Firstly, the mechanism deter- mines new CAC criteria based on traffic types. In the new CAC criteria, to create opportunities for the new calls, bandwidth degradation approach is introduced when the networks have scarce resources under heavy load scenario. Subsequently, an adaptive threshold value is applied to reserves available bandwidth for handoff calls by considering its traffic strength intensity[6,7,8]. The major contributions of this paper are threefold. First, is maximizing the throughput of the BE traf c which is blocked because of lack of efficient utilization of network resources. The second contribution is the reducing of call blocking probability (CBP) and call dropping probability (CDP) by using an adaptive threshold value which adjusts the network status. Lastly, an analytical model using two levels Markov chain was developed to measure the performance of the proposed scheme.



The Service Architecture Evolution of LTE Network

II. OVERVIEW OF THE LTE NETWORKS

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]

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

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]

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]

LTE Networks

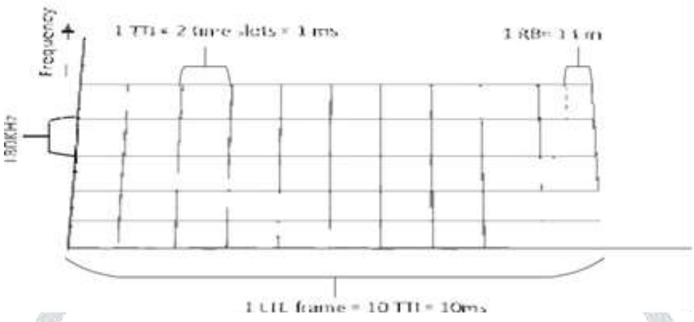
The LTE network was designed to surpass the attributes of 3G networks [2].It targets doubling the spectral efficiency; improving on the bit rate of cell edge users compared to the earlier networks [4].

Table1. shows a summary of the main LTE performance targets.

Table 1:Main LTE Performance Targets [3] .

Performance Metric	Target
Peak Data Rate	• Downlink: 100 Mbps • Uplink: 50 Mbps
Spectral Efficiency	• 2 - 4 times better than 3G systems
Cell-Edge Bit-Rate	• Increased whilst maintaining same site locations as deployed today
Mobility	• Optimized for low mobility up to 15 km/h • High performance for speed up to 120 km/h • Maintaining connection up to 350 km/h
Scalable Bandwidth	• From 1.4 to 20 MHz
RRM	• Enhanced support for end-to-end QoS • Efficient transmission and operation of higher layer protocols
Service Support	• Efficient support of several services (e.g., web-browsing, FTP, video-streaming, VoIP) • VoIP should be supported with at least a good quality as voice traffic over the UMTS network

The LTE network is built on a flat architecture called the Service Architecture Evolution shown in Figure 1.The figure consists of the radio access network and the Evolved Packet Core (EPC). The EPC provides the overall control of the UE and establishment of the bearer [5] which consists of Mobility Management Entity (MME), Serving Gateway (SGW), and Packet Data Network Gateway (PGW). The MME controls handover within LTE, user mobility, and UEs paging as well as tracking procedures on connection establishment. The SGW performs routing and forwarding of user data packets between LTE nodes as well as handover management between the LTE and other 3GPP technologies. The PGW connects the LTE network with other IP networks around the globe and provides the UEs access to the internet [2].The radio access network known as the Evolved-Universal Terrestrial Radio Access Network (E-UTRAN) performs all radio related functions [6] ,which comprises of the eNB and the UE. The UE represents the different types of devices used by the users while the eNB performs radio resource management (RRM) functions along with control procedures for the radio interface such as packet scheduling, CAC etc.



III. 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]

IV. 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:

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

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:

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

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:

$$\mathbf{RSS_V < RSS_L} \qquad \text{…………………} \ \mathbf{(3)}$$

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.

T [I B] 1 ……….. (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

η1 - 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:

$\delta q = \delta e - \delta r$ ………… (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:
$P = \min (Lmax,i-\delta q,i , Lmax,j-\delta q,j)$ … (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 {R1,R2,….Rn}.

Let us consider the user requests with good channels as G = {G1, G2, . . . Gk} and bad channels as B = {B1, B2, . . . Br}, where k , r<n.

Among G, handover calls are represented as H = {H1, H2, ….Hm} and new calls as N = {N1, N2, ….Np}, where m,p<k.

Among H, the VoIP calls and the video calls are

represented as Hv0 = {V1,V2,…Vq} and HI = {I1,I2,. . . It} respectively, where q,t<m.

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

Let ηA be the total available bandwidth, ηvot , ηit , ηB,be the reserved bandwidth for VoIP, video and bad channel classes , respectively.

For each {R1, R2, ... Rn} 1.1 If RSSV > RSSL,
G = {Gi, i = 1, 2,…k}

Else

B = {Bi, i = 1,2,…r}

End if

End for

For each G = {Gi, i = 1, 2, . . . k}

For each H = {H1, H2, ….Hm}

For Hv0 = {V1,V2,…Vq}

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

For $H_I = \{I_1, I_2, \ldots I_t\}$,

Check TOL of each call

Call with lower TOL is admitted first.

$$\eta_A = (\eta_A - \eta_{it})$$

If $\eta_{it} < \eta_A$, then

Bandwidth is reserved and call is admitted Else

Goto step 4.0

End if

End for

For each $N = \{N_1, N_2, \ldots N_p\}$, repeat from step 2.1.1

End For

When resources availability is insufficient,

For each B = {Bi, i = 1,2,…r}

$\eta_{A,} = \eta_{A,} + \eta_{B,}$ 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

VI. 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].

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.

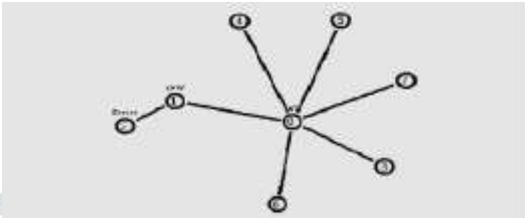


Figure 1: Simulation Topology

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.

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.

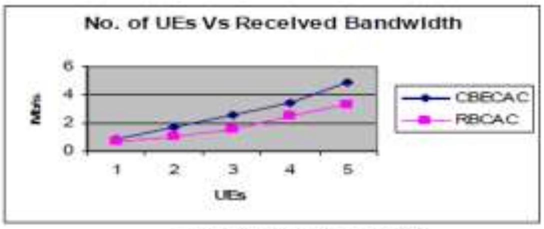


Fig 2: Number Vs Bandwidth

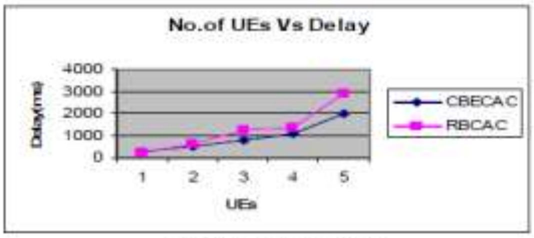


Fig 3: Number Vs Delay

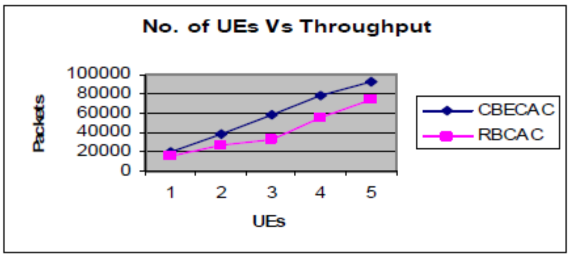


Fig 4: Number Vs Throughput

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.

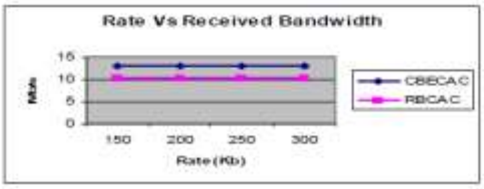


Fig 5: Rate Vs Bandwidth

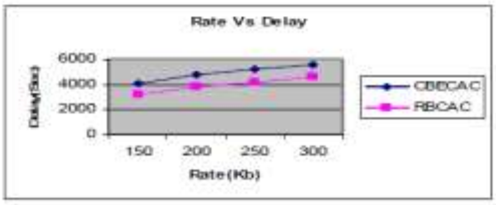


Fig 6: Rate Vs Delay

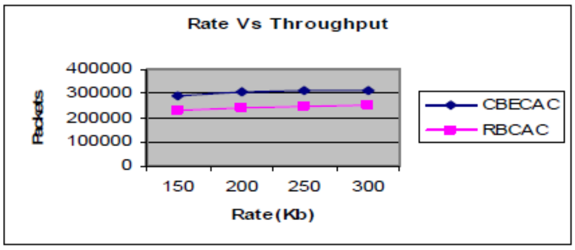


Fig 7: Rate Vs Throughput

CONCLUSION

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. CAC schemes proposed in recent literature, aiming at admitting call requests into the LTE network based on available resources. The schemes are classified into CAC with Resource Reservation (RR) , Resource Degradation (RD) ,Pre-emption ,Delay Awareness (DA) and Channel Awareness (CA).The way each scheme operates as well as the advantages and the disadvantages are also discussed. 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. .

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