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Efficient Bandwidth Usage in Cellular Networks for Better Quality of Service

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Abstract
In this paper, an efficient bandwidth usage scheme was formulated that guaranteed different Quality of Service (QoS) levels for multimedia traffic in constrained bandwidth cellular networks. The scheme takes an efficient approach to guarantee QoS without needing bandwidth over-provisioning to the end users. The scheme also incorporates admission control for QoS support. Four performance metrics, namely throughput, packet loss, end to end delay and jitter were used to evaluate performance. Network simulator 2 (NS2) released 2.35 simulator was used to test and validate this scheme. Through the simulation results, it was observed that adaptive bandwidth allocation yields better performances than conventional Internet Protocol (IP). It has also been proven that conventional IP does not guarantee service when the network experience congestion due to increased traffic in a cell during handover. That is the reason the proposed scheme will be of great interest in multimedia services as they continue to grow at an alarming rate.
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1. Introduction

As users for multimedia services increase, it results in a drastic demand for network bandwidth. Bandwidth provisioning becomes a major issue as it is a limited resource. This has fueled the need for more efficient bandwidth allocation strategies. To achieve efficiency in a cellular network, it will be required that bandwidth be shared fairly among individual users according to necessity. Specifically, users requiring more bandwidth for transmission should not be limited by bandwidth resources, conversely, those requiring less bandwidth should not be over allocated (Chandra and Kumar, 2018). Establishment and management of connections is important if quality of service is to be maintained in a cellular network as mobile equipment are in constant motion during communication sessions experiencing handovers from one cell to another (Meyyappan, 2019).

The cells to which the mobile traffic would be handed-over to must have sufficient bandwidth available if QoS is to be maintained. However, if available bandwidth is insufficient to accommodate the handover, forced termination occurs. Bandwidth reservation has been a method of choice in the recent past to mitigate this problem (Clarke, 2017). Reserving bandwidth for handover calls has its short comings too. First, it is highly inefficient as it would require large amounts of bandwidth to be reserved, (Shri and Ravi, 2015). Secondly, reservation of bandwidth in the network helps in seamless interactive multimedia services provisioning, but if the bandwidth set aside is too wide, the number of new calls blocked will be high due to a large amount of bandwidth reserved for handover calls though the traffic in the network might be low (Sri et al., 2019). In this case, the bandwidth is underutilized by not giving service to either handover call or new call. If the amount of bandwidth reserved is too small, the handover call success cannot be assured during high traffic situations in the network.

Due to random access of cellular networks by mobile users, traffic fluctuations are inevitable. This will lead to instances of burstiness in the network that causes congestion. Under this undesirable condition bandwidth cannot be over-provisioned as this would be inefficient. Therefore, when traffic surge occurs, traffic with stricter QoS requirements must be given preferential treatment (Chowdhury and Jang, 2020). Different multimedia services have different QoS requirements. For example, voice traffic may require low latency, while video download may require high assurance.

Other multimedia services like web traffic may tolerate up to a reasonable amount of delay. To provide these different levels of multimedia traffic services to satisfy all QoS requirements for each level is a hard task (Puschita et al., 2018). Call Admission Control (CAC) and bandwidth adaptation (BA) for handover calls are some of the provisioning strategies to limit the number of call connections into the networks and to degrade low priority traffic respectively (Pahlavan, 2018). A worthwhile CAC strategy has to balance between the admitted calls and the rejected calls for it to provide the desired QoS requirements. So, coordination amongst all parties such as admission control and bandwidth adaptation should be observed (Chowdhury et al., 2019).

2. Materials and methods

2.1 Model description

Understanding the nature of traffic in a system and choosing an appropriate traffic model is important for the simulation study to succeed. A general model with classes of multimedia traffic in mobile cellular network was considered. In this model, three cells were simulated in NS2 to evaluate the QoS performance of a mobile cellular network. Fig. 1 depicts the three cells arrangement in a Universal Mobile Telecommunication System (UMTS) network, where:

SGSN is the Serving Genera Packet Radio Service (GPRS) Support Node

RNC1 is the Radio network controller of the BS3

RNC2 is the Radio network controller of theBS1 and BS2

BS1 is the Originating base station of traffic

BS2 is the Base station that is handed over the traffic

BS3 is the destination base station of the traffic

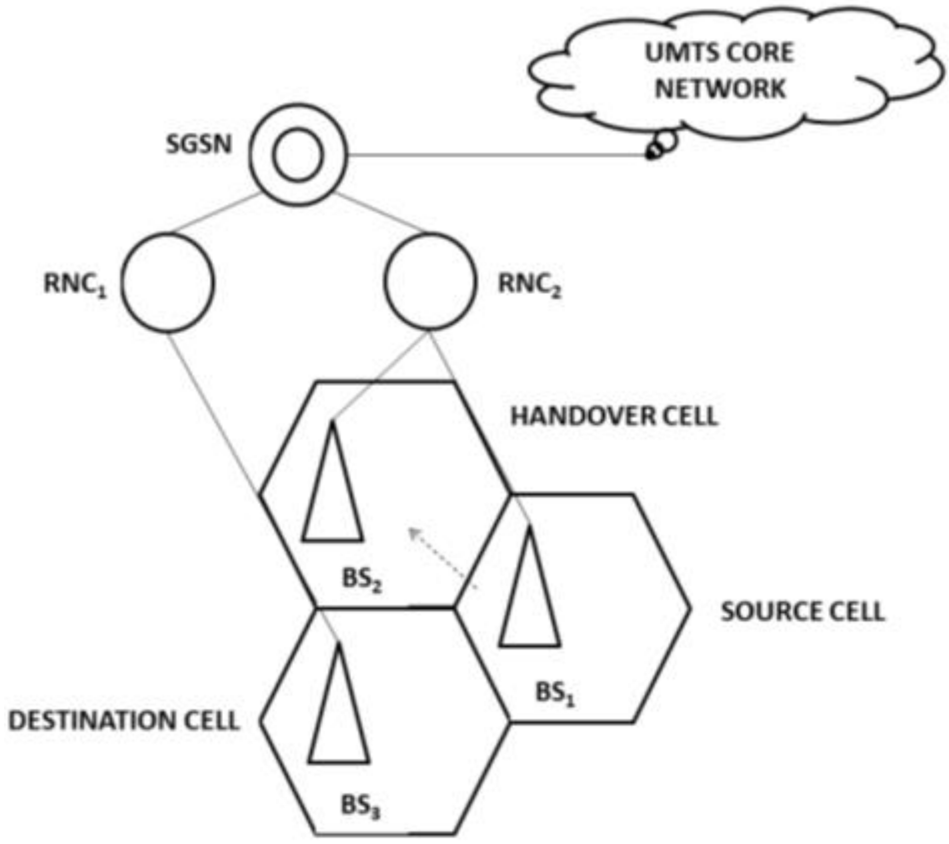

Fig. 1: Three cells arrangement in a Universal Mobile Telecommunication System (UMTS) network

2.2 Traffic model

Before analyzing the performance of a mobile cellular network, it was crucial to come up with a traffic model. The study consisted of three cells, namely the traffic originating cell, the traffic hand-over cell and the traffic destination cell. The following assumptions were made on handover behaviours in the traffic modelling. The calls made carried constant bit rate for voice and variable bit rate video and web content (files downloading). The model made use of common assumptions that handover calls follow a Poisson process (Alagu and

Meyyappan, 2017). Thus, packet inter-arrival times were assumed to follow an exponential distribution with a mean of 1 = l. The first assumption was that the number of arrivals (n) in a period from 0 to t obeys the Poisson distribution, $P_n(t)$.

$$P_n(t) = \frac{(\lambda t)^n}{n!} e^{(-\lambda t)} \quad (1)$$

where: t is used to define the window of time between 0 and t, n is the sum of the number of arrivals between 0 and t and l is the sum of average arrival rate. The second assumption was that the inter-arrival times are independent and obey the Exponential distribution, (P_0(t)):

Let us consider a special case of Poisson distribution where it is assumed that no arrivals occur in a given time period. Without any doubt, it is a straight forward answer that by substituting n with 0 in Equation (1), Equation (2) was deduced.

$$P_n(t) = e^{(-\lambda t)} \quad (2)$$

Let T_n be the call holding time with an indiscriminate variable exponentially distributedwith parameter m (in 1/seconds), $\frac{1}{\mu}$is the average duration of a call in seconds. Let T_h be the cell residence time with an indiscriminate variable exponentially distributed with parameter h (in 1/seconds), $\frac{1}{\eta}$is the average time a terminal stays in a cell in seconds. Let T_c be the channel holding time and is defined as T_c = min (T_n;T_h).

A call handover probability is determined by these features: (a) the average cell residence or sojourn time ($\frac{1}{\eta}$) and (b) the average call duration ($\frac{1}{\mu}$). As an exponential distribution assumption is made to both the call duration and the cell residence time, then the probability of a call handover (Ph) at a given time interval is:

$$P_n = P(T_n > T_h) \quad (3)$$

$$=\int_0^{\infty} f_h(T_h) \left[\int_{T_h}^{\infty} f_h\left(\frac{t}{T_h}\right) dt\right] dT_h \quad (4)$$

$$=\int_{T_h}^{\infty} \eta e^{-\eta T_h} \left[\int_{T_h}^{\infty} \mu e^{-\mu t} dt\right] dT_h \quad (5)$$

$$=\frac{\eta}{\eta+\mu} \quad (6)$$

The traffic flow in and out of cell was computed thus:

Let $\lambda_n$be the average intensity of new traffic (in calls per sec)

Let P_b be the call block probability

The new traffic intensity into a cell is

$$\lambda_n(1 - P_b) \quad (7)$$

2.3 Simulation parameters

The simulation settings and parameters are summarized in Table 1.

Table 1: Simulation parameters

Parameter	Specification
Users per cell	3
Number of cells	3
Simulation time (seconds)	10
Packet size	512 bytes
Voice traffic source	CBR
Video traffic source	Exponential on/off
Web traffic source	FTP
Transport protocol for voice and video	UDP
Transport protocol for web	TCP
Rate in cell before handover	1.8 Mbps
Rate in handed-over cell	2 Mbps
Queue length voice	20 (min) - 40 (max)
Queue length video	11 (min) 20 (max)
Queue length web	1 (min) 5 (max)

2.4 Simulation of a scalable adaptive bandwidth allocation policy

The simulation model consisted of 6 routers: Edge1 (BS1); Edge2 (BS2); Edge3 (SGSN), Edge4 (BS3); Core1 (RNC1); Core2 (RNC2) and six source nodes (S0-S5) and three destination nodes (D0-D2). The simulation was conducted using NS2 version 2.35. Matab 8.5 version was also used for graphical interface with the NS2. Three multimedia traffic were simulated. They included:

1. S0-D0 source destination pair which represented voice traffic with EFPHB,

2. S1 -D1 source destination pair which represented video traffic with AF PHB, and
3. S2 -D2 source destination pair which represented web (files download) traffic with BE PHB.

where PHB is the Per Hop Behaviours, EF is Expedited Forwarding, AF is the Assured Forwarding and BE is the Best Effort. After handover the traffic sources S0, S1 and S2 were not moved but were shifted with a certain probability. The voice traffic was marked with Differentiated Service code point (DSCP) 46, video traffic was marked with DSCP 20 and web traffic was marked with DSCP 0. If a traffic flow refused to conform to its profile as defined, it was assigned a reduced bandwidth. For example, video with DSCP 20 was downgraded to DSCP 21 and in the same way web with DSCP 0 was downgraded to DSCP 1. If a traffic flow did not still conform to the downgraded traffic profile, then it was discarded.

2.5 QoS measuring instruments

Throughput is the number of successfully received packets in a unit time and it is represented in bps.

$$\text{Throughput} = \frac{\text{received data} * 8}{\text{Data Transmission Period}} \qquad (8)$$

where the 8 represent the number of bits in a byte. Packet loss is the difference between the packets transmitted and packets received. Packet loss is caused by traffic congestion in a network. Packet Loss = Packets Transmitted-Packets Received End-to-end delay is the time taken by a data to arrive at its destination. A lower value of end-to-end delay implies better performance. Packet end-to-end delay is calculated as:

Delay = Packet Transmission Time – Packet Arrival Time

Jitter is an absolute value which is defined as the variation between the arrival time of two packets that are next to each other in a traffic flow and their departure time. A lower value of jitter implies better performance. It is calculated as:

$$\text{Jitter} = (a_k - a_j) - (d_k - d_j) \qquad (9)$$

where j and k are consecutive traffic packets arriving at a node, a_k and $a_j$are the arrivaltimes of the j and k packets at the node respectively and d_k and d_j are the departuretimes of the j and k packets at the node respectively.

3. Results and discussion

3.1 Adaptive bandwidth allocation and conventional IP

Primarily two scenarios were simulated. One was adaptive bandwidth allocation and another was Conventional IP (just plain IP network). Different sets of results from the two scenarios gave the opportunity to investigate the performance of these schemes. To analyse quantitatively the simulation results, the traffic was traced during the transmission process. For every packet that passes a trace object, information about the packet was written to a specified trace file. Final output results from trace files were visualized in plotted graphs. Packets were categorized depending upon whether they were very urgent, real-time (voice and video) and non-real-time (files download from web). Once the categorization was done the packets were sent through the separate queues according to their priority.

3.2 Voice handover and its effects on other multimedia traffic

It should be noted that voice handover occurred 3 seconds after the start of simulation.

3.2.1 Packet loss

Fig. 2 showed the packet loss in conventional IP for voice traffic handover. From Fig. 2, it was observed that voice traffic had the highest dropped packets followed by video traffic while files download traffic had the lowest packet drop. The packet drops for voice traffic and video traffic grew exponentially while for files download traffic, it increased slightly with time until it accumulated to 2200, 1900 and 20 respectively after 10 seconds.

Fig. 3 presented packet loss in adaptive bandwidth allocation for voice traffic handover. From Fig. 3, it was apparent that the line for voice traffic overlapped with the line for video traffic. It was seen that only files download traffic incurred packet losses while voice traffic and video traffic had zero packet drop. Files download traffic packet drop, increased exponentially with time until it reached 2640 at the 10 second’s mark.

It was noted that before handover, there were no packets dropped in both adaptive bandwidth allocation and conventional IP as the bandwidth available was enough to accommodate the traffic. Both voice traffic and video traffic are real-time traffic while files download traffic is non-real-time traffic. According to the adaptive bandwidth allocation algorithm real-time traffic gets priority over non-real time traffic hence there was no loss for both voice traffic and video traffic. In conventional IP the mechanism for bandwidth provisioning is based on best effort which does not guarantee bandwidth hence the loss experienced for the three types of traffic. Packet loss in conventional IP is highest for voice traffic because User Datagram Protocol (UDP) transport protocol does not attempt error recovery for erroneous packets; instead, it discards them while files download traffic has the lowest packet loss due to Transmission Control Protocol (TCP) transport protocol's error checking and recovery mechanism through retransmission of erroneous packets. Video traffic had medium packet loss in conventional IP as its packet transmission was not constant. The packet loss ratio in adaptive bandwidth allocation had the reverse order to the priority.

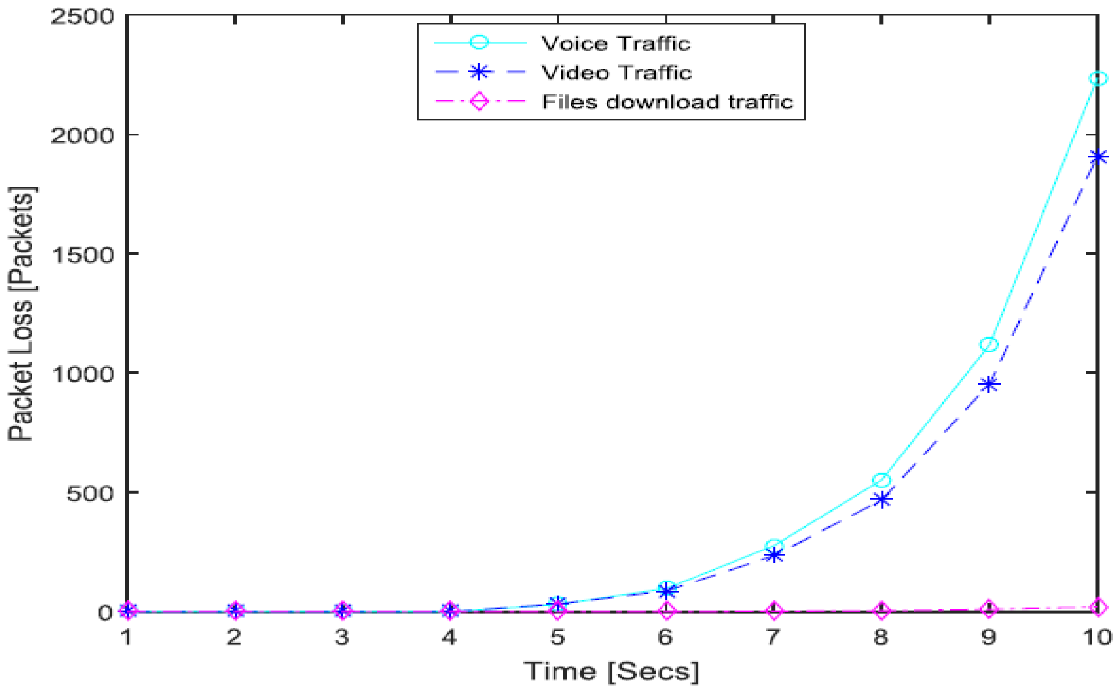

Fig. 2: Packet loss in conventional IP for voice traffic handover

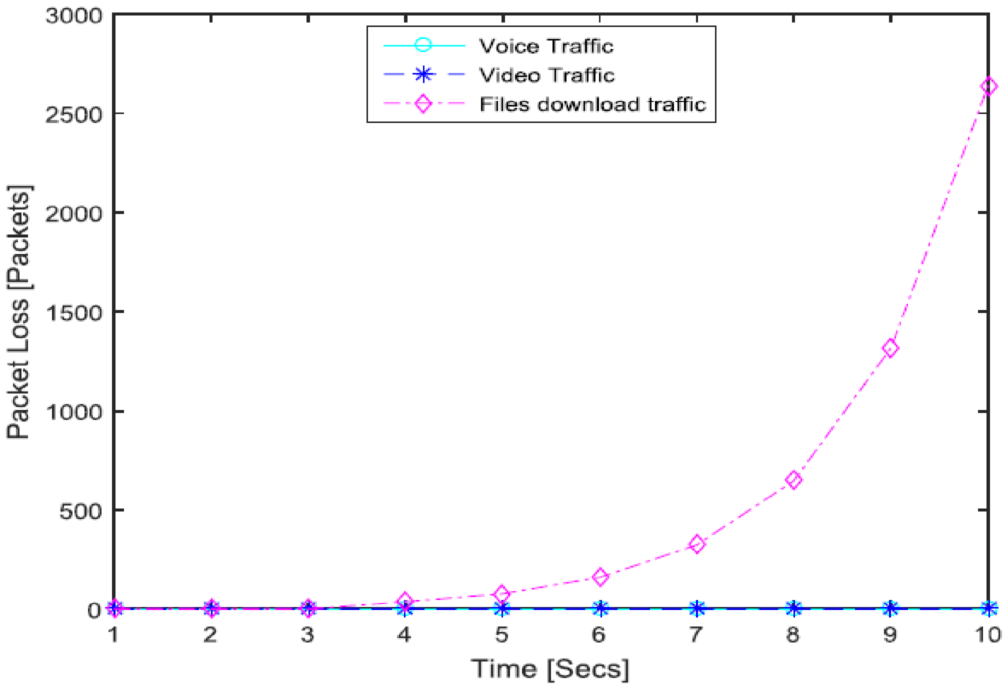

Fig. 3: Packet loss in adaptive bandwidth allocation for voice traffic handover

3.2.2 Throughput

Fig. 4 shows the Throughput in conventional IP for voice traffic handover. From Fig. 4, in conventional IP, the throughput for voice traffic fluctuated between1.75Mb/s and 2Mb/s. The throughput for video traffic zigzagged between 0.7Mb/sand 1.75Mb/s while files download traffic throughput ranged between 0.5Mb/s and1.2Mb/s in the course of 10 seconds simulation though there was a slight decrease at around 3 seconds when handover took place as a result of TCP control mechanisms. Throughput measures how fast we can transmit data. Voice traffic which used UDP protocol that is fast, had the highest throughput while files download traffic had the lowest throughput due to its use of TCP protocol that has error correction and acknowledgement mechanism making it slow. Video traffic uses UDP but its bit rate is variable during the on and off periods. During off period there is no transmission hence low throughput at that time.

Fig. 5 shows the Throughput in adaptive bandwidth allocation for voice traffic handover. As depicted by Fig. 5 in adaptive bandwidth allocation, throughput for voice traffic remained steady at 2Mb/s after its handover. For video traffic, throughput reached a peak of 2Mb/s and sank to a trough as low as 0.25Mb/s during the simulation. Files download traffic throughput rose to 1.2Mb/s for the first 3 seconds then fell to range between 0.1Mb/s and 0.5Mb/s after handover for the rest of the simulation. Generally, the throughput for voice traffic remained constant at 2Mb/s while files download traffic fell a lot after handover as the control mechanisms implemented regulated it by dropping some of its packets. The adaptive bandwidth allocation algorithm gave priority to voice, video and files download traffics in that order while in conventional IP, traffic was treated equally on first come first serve basis. That explained the high throughput for voice traffic and video traffic while files download traffic throughput sank a lot as offered load in the bottleneck link surged beyond the available bandwidth in adaptive bandwidth allocation after handover. This was due to high priority traffic starving low priority files download traffic by allowing them to have small amount of link capacity. It was observed that in adaptive bandwidth allocation high priority voice traffic had the highest throughput (stable at 2Mb/s) while low priority files download traffic had the lowest throughput compared to conventional IP. It was also true that before handover throughput was the same for both adaptive bandwidth allocation and conventional IP as bandwidth was sufficient. Of great importance is to take note that quality of service in a limited environment is a trade-off.

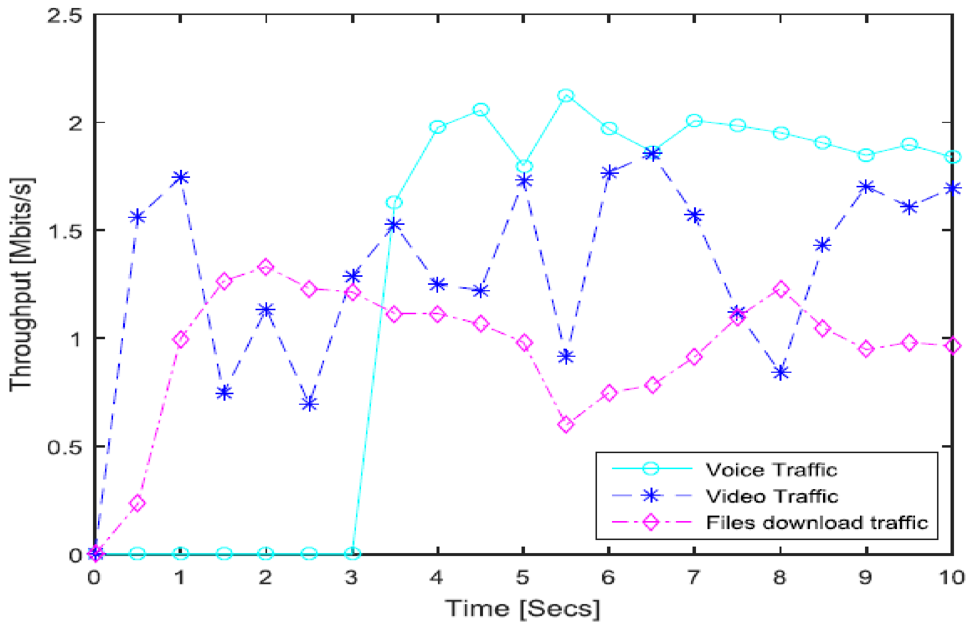

Fig. 4: Throughput in conventional IP for voice traffic handover

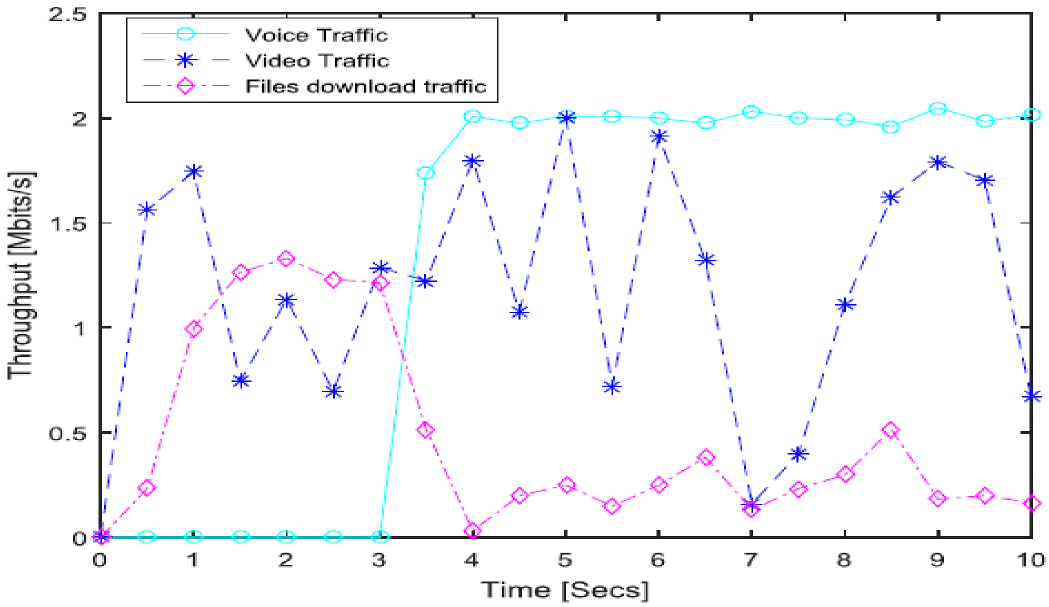

Fig. 5: Throughput in adaptive bandwidth allocation for voice traffic handover

3.2.3 End-to-end delay

Fig. 6 compared the end-to-end delay of voice traffic, video traffic and files download traffic in conventional IP for voice traffic handover. The end-to-end delay of voice traffic ranged between 65 milliseconds and 113 milliseconds after handover. The end-to-end delay of video traffic ranged between 65 milliseconds and 70 milliseconds for the first three seconds then shot up to 115 while for web traffic, the values zigzagged between 60 milliseconds to around 72 milliseconds for the first 3 seconds and 60 milliseconds and 118 milliseconds thereafter.

Fig. 7 highlighted the end-to-end delay among voice traffic, video traffic and files download traffic in adaptive bandwidth allocation. The end-to-end delay of voice traffic ranged between 65 milliseconds and 69 milliseconds after handover. The end-to-end delay of video traffic ranged between 65 milliseconds and 70 milliseconds while for files download traffic the values zigzagged between 60 milliseconds up to 72milliseconds before handover. After handover end to end delay of video traffic ranged between 65 milliseconds to 82 milliseconds while for files download traffic the values zigzagged between 60 milliseconds up to 84 milliseconds. It was observed that end to end delay before handover in both adaptive bandwidth allocation and conventional IP were the same as available bandwidth was sufficient. After handover end to end delay in adaptive bandwidth allocation was less than in conventional IP. Also, files download traffic had more delay between consecutive traffic as evidenced by clustering together of traffic in conventional IP than adaptive bandwidth allocation. This was due to an admission control in adaptive bandwidth allocation that admitted traffic that could be comfortably be accommodated in the network.

From Fig. 7, files download traffic also experienced traffic spikes in between because of burst data causing high end to end delay during that interval. The clustering together of lines was due to severe end to end delay between consecutive packets. The reason why the adaptive bandwidth allocation network showed a substantial impact on reducing premium packets delay was due to its routers using the priority queue mechanisms. It treated the entire premium class packet with the first priority queue according to the Service Level Agreement (SLA). The first priority queue transferred its packets first; therefore, the packets reached their destination quickly and spent less time waiting in the queue. The voice packets still suffered some queuing delays when there were other voice packets ahead of them in the same queue.

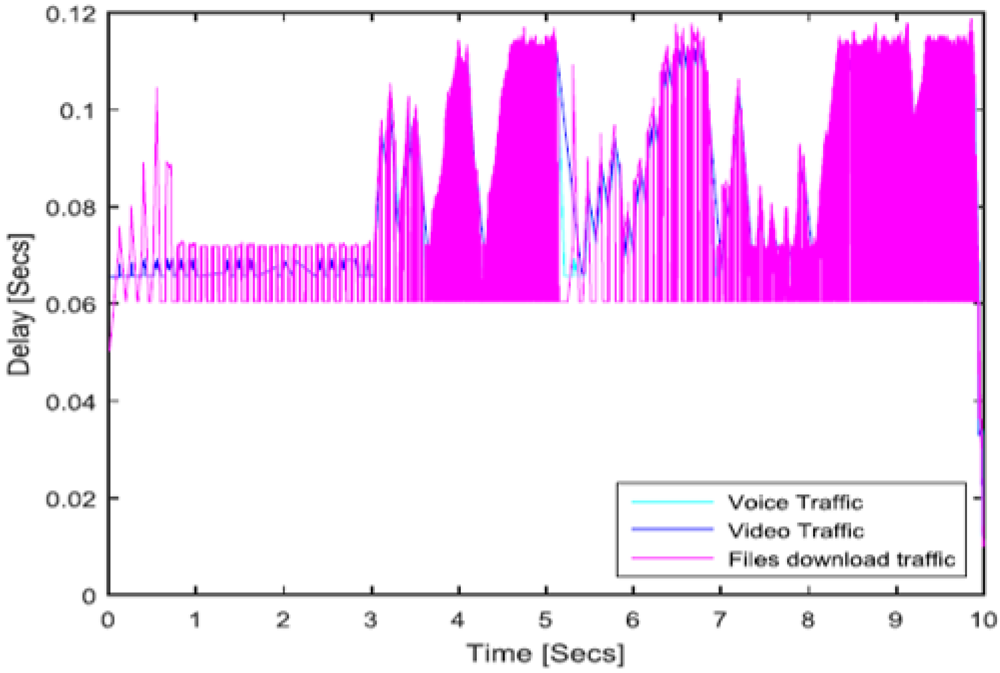

Fig. 6: Delay in conventional IP for voice traffic handover

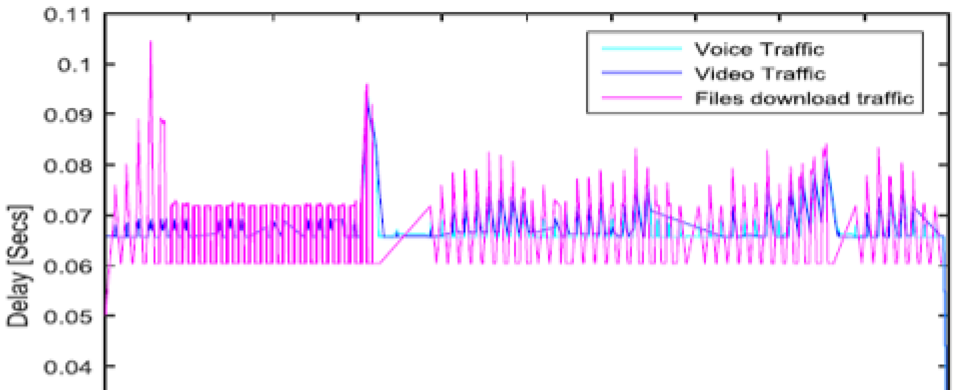

Fig. 7: Delay in adaptive bandwidth allocation after voice traffic handover

3.2.4 Jitter

Fig. 8 illustrated the jitter for voice traffic, video traffic and files download traffic in conventional IP. The video traffic jitter ranged between 4 milliseconds to -3 milliseconds for the first three seconds while files download traffic jitter ranged between 2.4 milliseconds to -2.4 milliseconds. After handover voice traffic jitter ranged between 4 milliseconds to -1 milliseconds for the rest of the simulation period while video traffic jitter ranged between 4 milliseconds to -45 milliseconds. Files download traffic jitter ranged between 40 milliseconds to -5 milliseconds. The files download traffic had the highest jitter because of reduced speed of transmission caused by acknowledgement and correction of errors while voice traffic had the lowest jitter as its transport protocol is unidirectional hence fast.

Fig. 9 showed the jitter for voice traffic, video traffic and files download traffic in adaptive bandwidth allocation. Jitter for video traffic zigzagged from 4 milli-seconds to -3 milliseconds while the jitter for files download traffic zigzagged from 2.4 milliseconds to -2.4 milliseconds. After handover voice traffic jitter ranged between 4milliseconds to -1 milliseconds for the rest of the simulation period while video traffic jitter ranged between 4 milliseconds to -18 milliseconds. Files download traffic jitter ranged between 18 milliseconds to -12 milliseconds. It was deduced that the jitter range for the three traffics in adaptive bandwidth allocation is less than in conventional IP due to admission control that limit the traffic into adaptive bandwidth allocation hence traffic in the network was served faster. Jitter was less before handover as bandwidth was sufficient to cater for the traffics which gave similar results as seen in (Sri et al., 2019). In adaptive bandwidth allocation, files download traffic was given less preference hence it waited longer in queues hence the high jitter range observed. Adaptive bandwidth allocation allowed the premium services to take enough bottleneck bandwidth. It starved the files download traffic by allowing them to have only a small amount of link capacity as its QoS needs were not as crucial as those of real-time traffic. As files download was given less preference, it waited longer in queues.

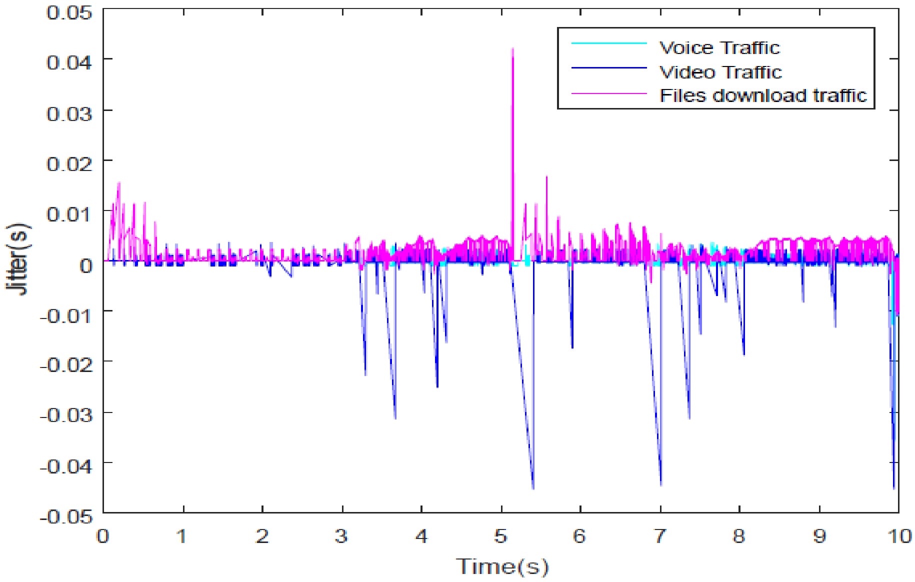

Fig. 8: Jitter in conventional IP for voice traffic handover

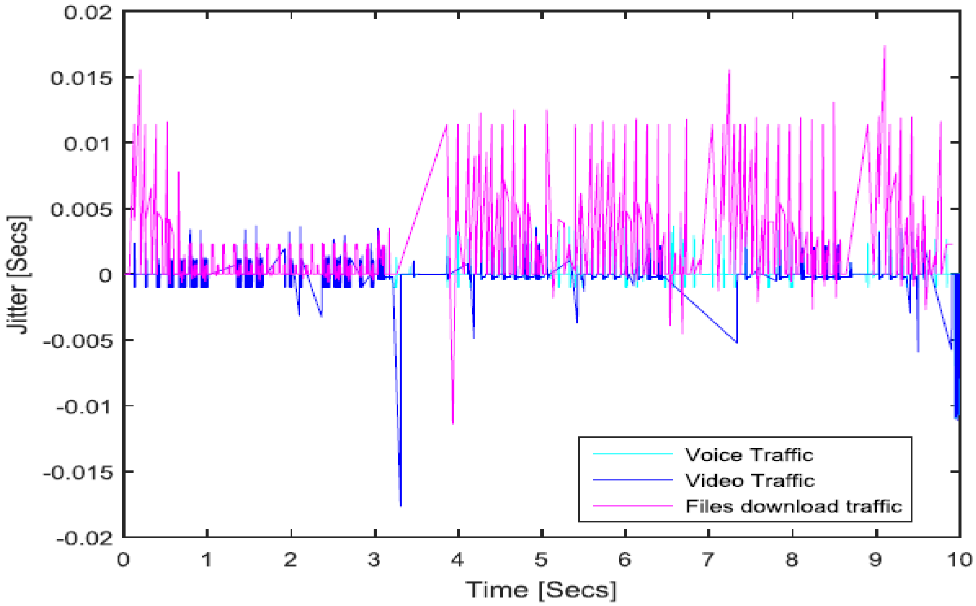

Fig. 9: Jitter in adaptive bandwidth allocation after voice traffic handover

4. Conclusion

In this paper, an efficient bandwidth usage scheme is formulated that guarantee different QoS levels for multimedia traffic in constrained bandwidth cellular networks. The scheme takes an efficient approach to guarantee QoS without needing bandwidth over-provisioning to the end users. The scheme also incorporates admission control for QoS support. The functioning of this scheme is that, if there is enough bandwidth in a cell, the handed-over traffic is admitted without any problem. But if the available bandwidth in that cell is scarce then some bandwidth from non-real time traffic in that cell is borrowed to serve real time traffic. Hence, this strategy gives high priority to handed-over voice traffic then video and finally files download with the lowest priority. Four performance metrics, namely throughput, packet loss, end-to-end delay and jitter are used to evaluate performance. NS2 release 2.35 simulator is used to test and validate this scheme. Through the simulation results, it is observed that adaptive bandwidth allocation yields better performances than conventional IP. It has also been proven that conventional IP does not guarantee service when the network experience congestion due to increased traffic in a cell during handover. That is the reason the proposed adaptive bandwidth allocation scheme will be of great interest in multimedia services as they continue to grow at an alarming rate.

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