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# **Journal of Electronic Networks, Devices and Fields**

## **Aims and Scope**

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# QoS Assured Downlink Proportional Fair Scheduler For Mitigation of Starvation For NRTPS And BE In Wi-Max Networks

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## **ABSTRACT**

*The performance of any network essentially depends on quality of service required and also on the scheduling scheme. In our paper we focus on analyzing essential QoS parameters like delay, jitters, throughput associated to four proportional fair scheduling as proposed by Lim & Kim [2] and Gupta et al [3]. We extend the idea of parameters such that the number of downlink connections of rtPS for starvation is drastically reduced for lower priority of service flows of nrtPS and BE to achieve the optimal QoS requirement without the excessive resource consumption.*

**Key words:** QoS, IEEE 802.16, WiMax, rtPS, nrtPS, BE

## **1. INTRODUCTION**

### **1.1 Wimax Technology and QoS Analysis**

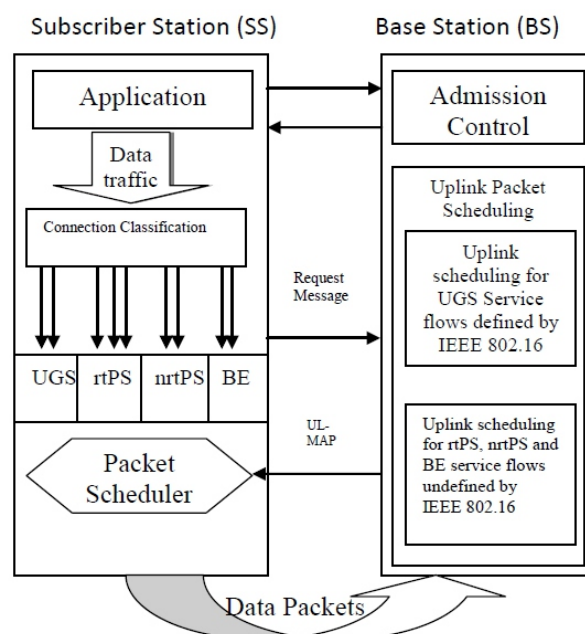
Wimax acronym for Worldwide Interoperability for Microwave Access supports both fixed and wireless mobile broadband. It is one of most promising technology for Broadband Wireless Access (BWA) aiming to provide services on a scale of Metropolitan Area Network (MAN)[3]. IEEE 802.16 based wireless scheme i.e. Wimax not only concentrates on lowering the cost of wired connections by enhancing features but also focusing highly on QoS(Quality of Service) requirements.

On the other hand QoS (Quality of Service) refers to a broad collection of networking technologies and techniques. The goal of QoS is to provide guarantees on the ability of a network to deliver predictable results. Elements of network performance within the scope of QoS often include availability (uptime), bandwidth (throughput), latency (delay), and error rate.

In Broadband Wireless communications, QoS is still an important criterion. So the basic feature of WiMAX network is the guarantee of QoS for different service flows with diverse QoS requirements. While extensive bandwidth allocation and QoS mechanisms are provided, the details of scheduling and reservation management are left not standardized. In fact, the standard supports scheduling only for fixed-size real-time service flows. The scheduling of both variable-size real-time and non-real-time connections is not considered in the standard. Thus, WiMAX QoS is still an open field of research and development for both constructors and academic researchers. The standard should also maintain connections for users and guarantee a certain level of QoS. Scheduling is the key model in computer multiprocessing operating system. It is the way in which processes are designed priorities in a queue and provide mechanism for bandwidth allocation and multiplexing at the packet level.

## 1.2. Wimax Architecture

IEEE 802.16[1] architecture includes one Base Station (BS) and Multiple Subscriber Station (SS). Communication occurs in two directions: from BS to SS is called Downlink and from SS to BS is called Uplink. During downlink, BS broadcasts data to all subscribers and subscribers selects packets destined for it. In IEEE 802.16 the BS (Base Station) centrally allocates the channels in different slots to different SSs (Subscriber Stations) for uplink and downlink which in turn allocates these resources to the various connections they are supporting at that time. Since BS is aware of the channel state of sub channels for all SSs and therefore can exploit channel user diversity by allocating different sub channels to different SSs as shown in the Fig1 below.



**Figure 1.** Wimax Architecture

In our case the packets are transferred from source node (rtPS) to destination node (nrtPS or BE) after following various scheduling, modulation and routing technique Kim and Lim [2] and Gupta et al[3]. Here rtPS connection acts as base whereas nrtPS and BE serve as SSs. The overall system throughput can be maximized by allocating a sub channel to the SS with the best channel state. [2, 12]

### 1.3 QoS Service Classes

To support the different types of traffic with their various requirements IEEE 802.16-2005 defines four QoS service classes: Unsolicited Grant Scheme (UGS), Real Time Polling Service (rtPS), Non Real Time Polling Service (nrtPS), BE(Best Effort)..

**UGS** is designed to support real time data stream consisting of fixed size data packets issued at periodic intervals such as E1/T1 and voice over IP without silence suppression. The main QoS parameters are maximum sustained rate (MST), maximum latency and tolerated jitter (the maximum delay variation).

**rtPS:** This service class is for variable bit rate (VBR) real-time traffic such as MPEG compressed video.

**nrtPS:** This service class is for non-real-time VBR traffic with no delay guarantee. Only minimum rate is guaranteed. In the nrtPS scheduling service, the BS provide unicast uplink request polls on a 'regular' basis, one second or less, which guarantees that the service flow receives request opportunities even during network congestion.

**BE:** This class is designed to support data streams for which no minimum service guarantees are required, like the case in HTTP traffic. The BS does not have any unicast uplink request polling obligation for BE SSs. Therefore, a long period can run without transmitting any BE packets. [16]

These are summarized in Table 1.

**Table 1.** QoS Service Classes [16]

QoS Category	Applications	QoS Specifications
UGS Unsolicited Grant Service	VoIP	-Maximum Sustained Rate -Maximum Latency Tolerance -Jitter Tolerance
rtPS Real-Time Polling Service	Streaming Audio or Video	-Minimum Reserved Rate -Maximum Sustained Rate -Maximum Latency Tolerance -Traffic Priority
ErtPS Extended Real-Time Polling Service	Voice with Activity Detection (VoIP)	-Minimum Reserved Rate -Maximum Sustained Rate -Maximum Latency Tolerance -Jitter Tolerance -Traffic Priority
nrtPS Non-Real-Time Polling Service	File Transfer Protocol (FTP)	-Minimum Reserved Rate -Maximum Sustained Rate -Traffic Priority
BE Best-Effort Service	Data Transfer, Browsing, Web etc.	-Maximum Sustained Rate -Traffic Priority

## 2. SYSTEM DESIGN

PMP mode and mesh mode are the two types of operating modes defined for IEEE 802.16. In the PMP mode SSs are geographically scattered around the BS. The performance of IEEE 802.16 in the PMP mode is verified in [8][9]. Our system model is based on a time-division- duplex (TDD) mode. The IEEE 802.16 frame structure is illustrated in Fig.2 [2] given below. The downlink subframe starts with preamble followed by frame control header (FCH), downlink map (DL-MAP), uplink map (UL-MAP) messages and downlink burst data. The DLMAP message defines the start time, location, size and encoding type of the downlink burst data which will be transmitted to the SSs. Since the BS broadcasts the DLMAP message, every SS located within the service area decodes the DL-MAP message and searches the DL-MAP information elements (IEs) indicating the data bursts directed to that SS in the downlink subframe. After the transmit/receive transition gap (TTG), the uplink subframe follows the downlink subframe. IEEE 802.16 provides many advanced features like adaptive modulation coding (AMC), frame fragmentation and frame packing. In the current work, the focus is on the downlink scheduling scheme. A multiuser scheduler is designed at the medium access control (MAC) layer. Delay requirement is taken into account in the scheduler design. The AMC, packet fragmentation and packet packing have not been considered. [2, 12]

### 2.1 Multi- User scheduler of the MAC Layer

In this section a multi user scheduler is designed at the medium access control (MAC) layer. Here we take the delay requirement into account in the scheduler design. AMC, packet fragmentation and packet



packing have not been considered. In case of the UGS traffic the required bandwidth is reserved in advance. Thus we only take rtPS, nrtPS and BE connections which are focused in the design. [2][12]

## 2.2 Novel design of proportional fair scheduling

The proportional fair scheduling [2] has shown an impressive guideline in the scheduler design because it maximizes the total sum of each SS's utility. The concept of the proportional fair scheduling is widely accepted in scheduling design. Recently, Kim and Lim[2] proposed QoS requirement by adding the delay requirement term in the proportional fair scheduling scheme to support the scheduling scheme that one of the rtPS and nrtPS connections is scheduled on every scheduling instance. They define the scheduling ratio  $x$  as the average number of scheduling times for rtPS connections per one nrtPS connection. If rtPS and nrtPS connections are scheduled equally, the ratio  $x$  becomes unity otherwise if rtPS connection is scheduled more frequently than nrtPS connections, the scheduling ratio  $x$  is taken greater than unity. Recently, Gupta et al [3] have proposed an alternate scheduling scheme based on proportional fairness. The scheduling parameters have been selected based on the number of connections of rtPS connections to specified number of nrtPS connections in the network. The scheduling algorithm must provide fairness to all the requests with different QoS classes. In case of Kim and Lim [2] and Gupta et al [3] there is no starvation for nrtPS whereas starvation for BE in both the cases is 100%. But for fairness of scheduling, in this paper, we extend this idea of scheduling parameters being selected such that the number of connections of rtPS be connected to nrtPS and BE with the least starvation to both the traffics. In this case according to Lim and Kim they have taken the ratio  $x:k$  as to 1:1 whereas Gupta takes ratio as  $x:k$  as to 1:2, 1:3, 1:4 for the above four schedules corresponding to the values of  $k$  and in each case they have given 100% starvation to BE.

Now for fairness of the scheduling, since the starvation of BE is also to be reduced as such we take the connections of rtps to nrtps and BE in the ratio such that  $x:k:k' :: 1:1:1$  which is not possible in their case. They have taken the values  $x_i$  such that  $x_i$  is equal to 1 to 10. We have therefore designed novel set of connections from rtps to nrtps ranging from  $x_i$  equal to 1 to 12 and computed the delay corresponding of four connections each of rtps, nrtps and BE. This results into almost hundred percent mitigation of starvation in both cases of nrtPS and BE though at the cost of increase in delay.

## 2.3 The following notations are used throughout this paper:

$\Phi_i(t)$  : the metric for fair scheduling

$DRC_i(t)$  : rate requested by  $i$ th SS (Subscriber System)  $R_i(t)$ : Average Rate received by  $i$ th SS

$T_c$ : size of window

$R$ : Transmission rate.

$C$  and  $C'$ : intensities associated to the corresponding delays  $d$  and  $D$  Traffic connections...rtps, nrtps and BE

### 3. PFS AS SCHEDULER DESIGN

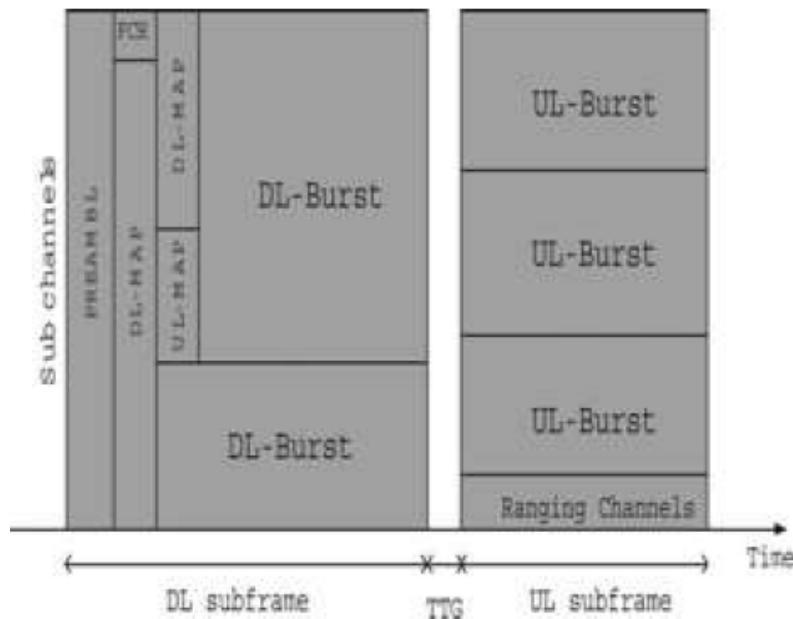
This novel proportional fair scheduling (PFS), [2] has shown an impressive guideline in scheduler design because it maximizes the total sum of each  $SS_i$  utility. The metric as defined in [2] for each connection is given as follows:

$$\Phi_i(t) = DRC_i(t) / R_i(t). \quad (1)$$

Where  $DRC_i$  [12] is the rate requested by the  $SS_i$  and  $R_i$  is the average rate received by the  $SS_i$  over a window of the appropriate size  $T_c$  [2, 3, 4, 12].

The average rate  $R_i$  is updated as

$$R_i(t+1) = (1 - 1/T_c) * R_i(t) + 1/T_c * \text{current transmission rate}. \quad (2)$$



**Figure 2.** IEEE 802.16[2] frame structure

#### 3.1 Proposed Novel Proportional Fair Scheduling (PNPFS)

In the proportional fair scheduling, the strict fairness is guaranteed, however the QoS requirement is not reflected. To the knowledge of authors rtps connections for QoS have been discussed in the literature

with regard to one specified nrtps connection, Kim et. al.[2] and Gupta et al. [3,4] have generalized this concept by associating various parameters such as scheduling ratio  $\alpha_i$  of rtps class parameter associated to  $k$  number of nrtps class. Thus, the general scheduling scheme is being introduced that satisfies the delay requirement. In this paper we have generated a number of fair scheduling schemes corresponding to the parameter  $k$  so that the delay requirements are minimized with regard to corresponding nrtps schemes as mentioned below. The metric value of the rtPS connections with the delay requirement should be increased as the queuing delay increases because the scheduler selects the connection with the highest metric value with nrtps connections, because nrtps connections are in the lowest priority. For the above mentioned conditions the equations for rtps and, nrtps are proposed by the authors in papers [2], [3]. Here we are generalizing the above equation by proposing a new scheduling scheme based on the following metrics for rtPS, nrtPS and BE connections are given as:

$$\Phi_{rt,i}(t) = 1/R_{rt,i}(t) + C(1 + 2/\pi * \arctan(d)). \quad \text{if } q_i > 0 \text{ and } d \geq d_{min} > 0 \quad (3)$$

$$= 1/R_{rt,i}(t) + C. \quad \text{if } q_i > 0 \text{ and } 0 < d < d_{min}.$$

$$= 0 \quad \text{if } q_i = 0$$

$$\Phi_{nrt,i}(t) = 1/R_{nrt,i}(t) + C \quad \text{if } q_i > 0 \quad (4)$$

$$= 0 \quad \text{if } q_i = 0$$

$$\Phi_{BE,i}(t) = 1/R_{BE,i}(t) + C' \quad \text{if } q_i > 0 \quad (5)$$

$$= 0 \quad \text{if } q_i = 0$$

The parameter  $d$  is the queuing delay and  $C$  means the intensity of the delay requirement in the rtPS connections to nrtPS connections. Here we define the parameter  $D$  as the queuing delay and  $C'$  means the intensity of delay requirement in rtPS to BE connections. The parameter  $d_{min}$  is the minimum delay that triggers the service differentiation between the rtPS connection and nrtPS connection, and  $q_i$  means the queue length of the connection  $i$ . We note here that  $R_{rt}$ ,  $R_{nrt}$  and  $R_{BE}$  are updated in the same manner as in the proportional fair scheduling, that is

$$R_{rt,i}(t+1) = (1 - 1/T_c)R_{rt,i}(t) + r/T_c, \text{ if connection } i \text{ is scheduled.} \quad (6)$$

$$= (1 - 1/T_c) R_{rt,i}(t), \text{ otherwise}$$

Where  $T_c$  is the window size to be used in the moving average and  $r$  is the current transmission rate requested by the SS.

The long-term rate is the average sum of the previously scheduled transmission rates during the time window  $T_c$ , where the high  $T_c$  value means that the long-term rate changes slowly because the average is taken over many previous transmission rates. The long-term rate of a connection decreases exponentially before the connection is scheduled, and it increases when the connection is scheduled. We do not consider the AMC, so  $r$  is a constant. On every frame, the scheduler selects the connection that has

the highest metric value. Owing to the delay requirement term in the rtPS metric, rtPS connections are served more frequently than other connections when the queuing delay increases [2, 3, 12].

#### 4. NOVEL PARAMETERS WITH ANALYSIS

In this paper we define the scheduling ratio  $x$  as the average number of rtPS connection per  $k_1$  number of nrtPS and  $k_2$  number of BE connections where  $k_2 \leq k_1$ . In order to avoid BE starvation, we extend this idea to BE connections given by the following two cases:

##### 4.1 Case I:

If rtPS and nrtPS connections are scheduled equally, the scheduling ratio  $x$  equals  $k_1$  corresponding to no connections to BE for  $k_2=0$ . Following Kim and Lim [2] and Gupa et al [3], if rtPS connection is scheduled more frequently than nrtPS connection, the scheduling ratio  $x$  becomes greater than  $k_1$ . Now the average scheduling interval in the rtPS connection is  $((x+k_1)/x)$  frames because, on an average,  $k_1$  nrtPS schedule correspond to  $x$  rtPS connections. As a result of this, the average scheduling interval in nrtPS connection is  $(k_1+x)$  frames. At the steady state, the average long-term rates of rtPS and nrtPS connections at the scheduling instance are as follows:

$R_{rt} = R_{rt} (1 - (1/T_c))((k_1+x)/x) + (r/T_c)$ , at the steady state, we obtain

$$R_{rt} = (r/T_c) / (1 - (1 - (1/T_c))((k_1+x)/x)) \quad (7)$$

Analogously, Since  $R_{nrt} = R_{nrt} (1 - (1/T_c))((k_1+x)) + (r/T_c)$  at the steady state, we obtain

$$R_{nrt} = (r/T_c) / (1 - (1 - (1/T_c))((k_1+x))) \quad (8)$$

We consider the same assumption as in [14] that the average metric value for each of rtPS and nrtPS connection at the scheduling instance becomes similar to each other with delay  $d$ . Hence,

$$\begin{aligned} & 1/R_{rt} (1 - (1/T_c))((k_1+x)/x) + C(1 + (2/\pi)\arctan(d)) \\ & \approx 1/R_{nrt} (1 - (1/T_c))((k_1+x)) + C \end{aligned} \quad (9)$$

From (7) and (8), (9) can be written as

$$\begin{aligned} & ((1 - (1 - (1/T_c))((k_1+x)/x)) / ((r/T_c) / (1 - (1 - (1/T_c))((k_1+x)/x)) + C(1 + (2/\pi)\arctan(d))) \\ & \approx ((1 - (1 - (1/T_c))((k_1+x))) / (r/T_c) / (1 - (1 - (1/T_c))((k_1+x))) + C. \end{aligned} \quad (10)$$

## 5.2 Case II:

Now if rtPS connection is scheduled after  $k_1$  nrtPS connections with  $k_2$  BE connections (with less frequently), the scheduling ratio  $x$  becomes greater than  $k_2$ , where  $k_2 \leq k_1$ . Now the average scheduling interval in the rtPS connection is  $((x+k_2)/x)$  frames because, on the average, the number of  $k_2$  BE schedule correspond to  $x$  rtPS connections subject to  $k_2 \leq k_1$ . As a result of this, the average scheduling interval in BE connection is  $(k_2+x)$  frames. At the steady state, the average long-term rates of rtPS and BE connections at the scheduling instance are as follow:

$R_{rt} = R_{rt} (1 - (1/T_c))(k_2+x)/x + (r/T_c)$ , at the steady state, we obtain

$$R_{rt} = (r/T_c) / (1 - (1 - (1/T_c))(k_2+x)/x) \quad (11)$$

Analogously, Since  $R_{BE} = R_{nrt} (1 - (1/T_c))(k_2+x) + (r/T_c)$  at the steady state, we obtain

$$R_{BE} = (r/T_c) / (1 - (1 - (1/T_c))(k_2+x)) \quad (12)$$

As in [14], the average metric value for each rtPS and BE connection at the scheduling instance with delay  $D$  becomes similar to each other.

Hence,

$$\begin{aligned} & 1/R_{rt} (1 - (1/T_c))(k_2+x)/x + C'(1 + (2/\pi)\arctan(D)). \\ & \approx 1/R_{BE} (1 - (1/T_c))(k_2+x) + C'. \end{aligned} \quad (13)$$

From (11) and (12), (13) can be written as

$$\begin{aligned} & ((1 - (1 - (1/T_c))(k_2+x)/x) / ((r/T_c) / (1 - (1 - (1/T_c))(k_2+x)/x) + C'(1 + (2/\pi)\arctan(D))). \\ & \approx ((1 - (1 - (1/T_c))(k_2+x)) / (r/T_c / (1 - (1 - (1/T_c))(k_2+x)) + C' \end{aligned} \quad (14)$$

We note here

$$0 \leq k_2 \leq k_1, \text{ such that } x:k_1:k_2=1:1:1 \text{ where } x, k_1 \text{ and } k_2 \text{ are positive integers.} \quad \dots (14)'$$

Now generalizing each of the above equations (10) and (14) for  $i$  iterations corresponding to the above parameters such as  $x, k_1, k_2, C, C', d$  and  $D$  and on simplifying these equations we have as follow:

$$d_i = \tan(((\pi * T_c) / (2 * r * C)) * ((1 - 1/T_c)(k_1+x)/x - (1 - 1/T_c)(k_1+x))) / (1 - 1/T_c)((x * x + k_1 * x + k_1 + x)/x) \quad (15)$$

and

$$D_i = \tan(((\pi * T_c) / (2 * r * C')) * ((1 - 1/T_c)(k_2+x)/x - (1 - 1/T_c)(k_2+x))) / (1 - 1/T_c)((x * x + k_1 * x + k_2 + x)/x) \quad (16)$$

## 5. ANALYSIS OF NRTPS AND B.E TRAFFICS WITH REGARD TO RTPS TRAFFIC AS A BASE STATION

In their paper, Gupta et al [3] ,using eqs (7),(8)have obtained the parameters of delay  $d$  from eq.(10) as against scheduling ratio  $x$  corresponding to  $k_1=1,2,3$  and 4. In particular, their findings are such that for  $k_1=1$ , almost all types of parameters including various forms of delays turn out approximately as derived by Kim and Lim [2]. Using simulation based on statistical analysis they have obtained various parameters including different kinds of delays corresponding to the values of  $k_1$  in  $\{1, 2, 3, 4\}$  and obtained that the delays corresponding to first five rtPS connections are smaller than subsequent nrtPS connections but in start these increase more than nrtPS for  $x > 4$  and it is true for all values of  $k_1$ . Recently, a number of papers have discussed that BS scheduler can guarantee minimum bandwidth for each service flow and ensure fairness and QoS in distributing excess bandwidth among all connections. At the same time, for the downlink scheduler in SS (rtPS) can provide differentiated and flexible QoS support for all of the four scheduling service types. It can both reduce the delay of real time applications and guarantee the throughput of non real applications also enhancing bandwidth utilization of the system and fairness of resources even at lower traffic intensity.

In view of the downlink service we propose rtPS as an efficient scheduling scheme which eliminates the starvation problem of lower priority class services nrtPS and BE .Recently Raina et al [12] have discussed and presented a scheduling scheme reflecting the delay requirements by introducing the same delay intensities corresponding to different ratios which does not give the maximum mitigation of starvation to lower priority classes such as BE traffic. Now in this paper, we generalize the idea of [2,3,12] and to study delay  $D$  as associating to rtPS so that it associates  $k_1$  connections to nrtPS traffic and  $k_2$  connections to BE traffic in the fair proportional ratio subject to  $x:k_1:k_2=1:1;1$  to allow maximum mitigation of starvation to BE traffic. Analogously using eqns. (11) and (12) we obtain the parameters of delay  $D$  from eq. (14) subject to (14)'. We then study the relative behavior of  $d$  and  $D$ .

Now we determine the solution set  $(d_i, D_i)$  corresponding to the various parameters  $C, C', x_i$  ( $i=1..12$ ) and  $k_1$  and  $k_2$  take the values  $\{1, 2, 3, 4\}$ . As each of parameters  $C$  and  $C'$  increase, each of the delays  $d_i$  and  $D_i$  decrease because each of queuing delays  $D_i$  and  $d_i$  are inversely proportional to  $C'$  and  $C$  respectively. We notice here that our results turn out on parallel lines with the results given in papers [2, 3] for the values  $k_1=1$  such that  $x: k_1=1:1$  with no connections of BE. We observe that with increase in  $x$ , the delays corresponding to BE for all values of  $k_2 \leq k_1$  are smaller than respective delays of nrtPS and rtPS as can be seen from the below given diagrams.



## 6. SIMULATION RESULTS OF FOUR FAIR SCHEDULING SCHEMES

Using Matlab, the values of  $D$  (delays) corresponding to different prescribed values of  $x_i, k_1, k_2$  for  $1 \leq x_i \leq 12, 1 < k_1, k_2 \leq 4$ ,  $C_i, C_i'$  taking each of the three values of  $(C, C') = \{(0.1, 0.2), (0.05, 0.06), (0.01, 0.02)\}$  as given in the following tables. Since rtPS connections are whole numbers therefore, their connections with nrtPS and BE have to be in ratio of  $x: k_1:k_2$ . Thus in this paper we design the three parameters having the ratio  $x: k_1:k_2:: 1:1:1$  which is only possible if we set the nodes  $x$  of rtPS as multiples of 3 and this case we take varying from 1 to 12. Then in this case we get the maximum starvation mitigated for the lower priority traffic BE.

We further observe here that with increase in delays corresponding to rtPS, nrtPS and BE for all values of  $k_1$  and  $k_2$  corresponding to  $C'$  are smaller than the respective corresponding to rtPS and nrtPS for values of  $C$  as can be seen by the following diagrams

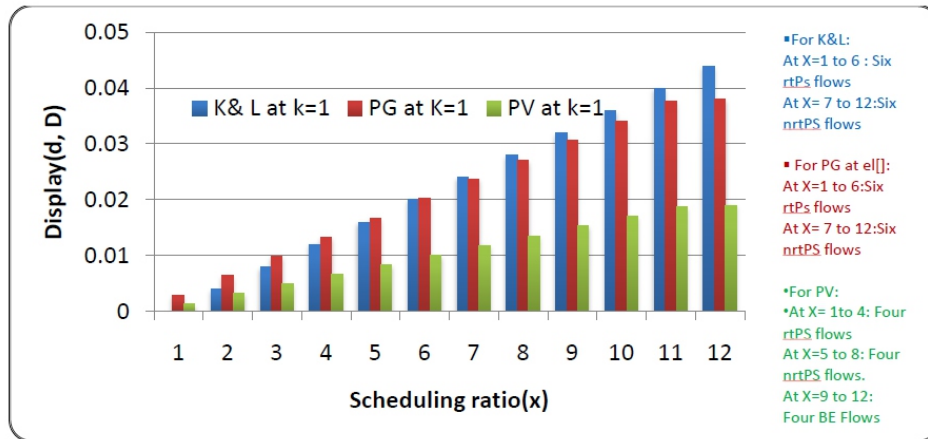
### 6.1 Table-Analysis

A. For,  $C=0.1$  and  $k_1=1 \dots 4$   $d$ , delays of 6 flows of rtPS traffic and 6 flows of nrtPS traffic  
And

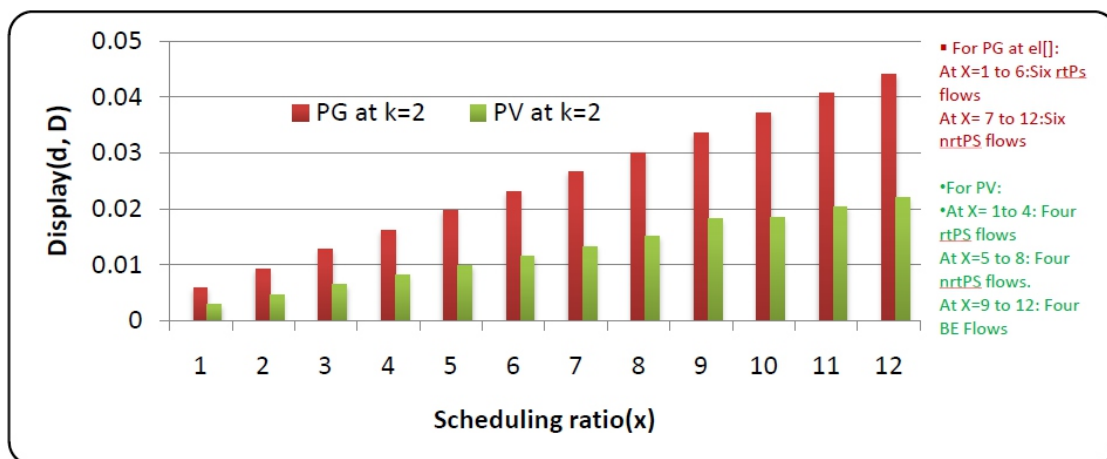
For,  $C'=0.2$  and  $k_2=1 \dots 4$ .  $D$  delays of 4 flows of rtPS traffic, 4 flows of nrtPS traffic and 4 flows of BE traffic.

We note here that for K&L,  $k=k_1=1$ , for P.G,  $k_1=k=1$  and P.V,  $K_2=k=1$ , etc.

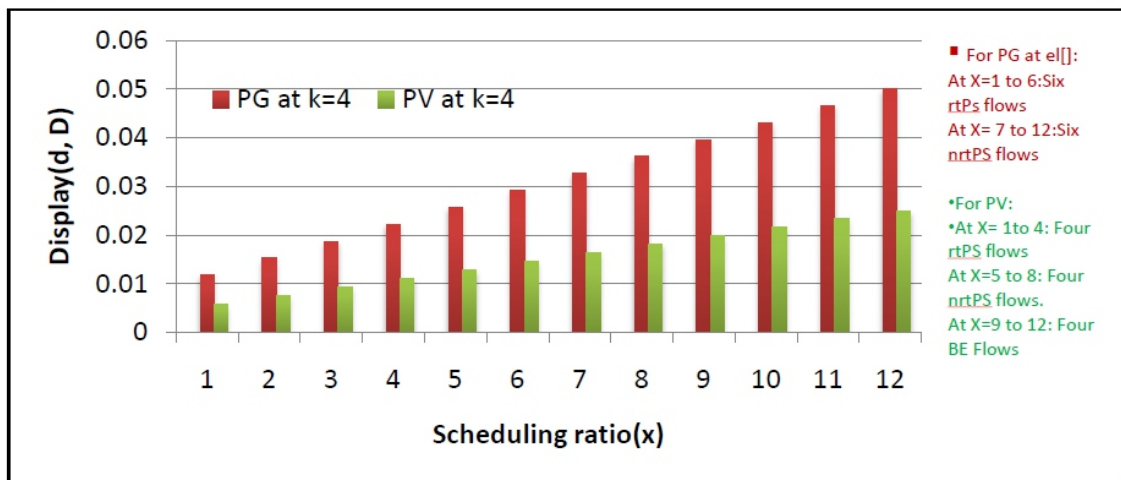
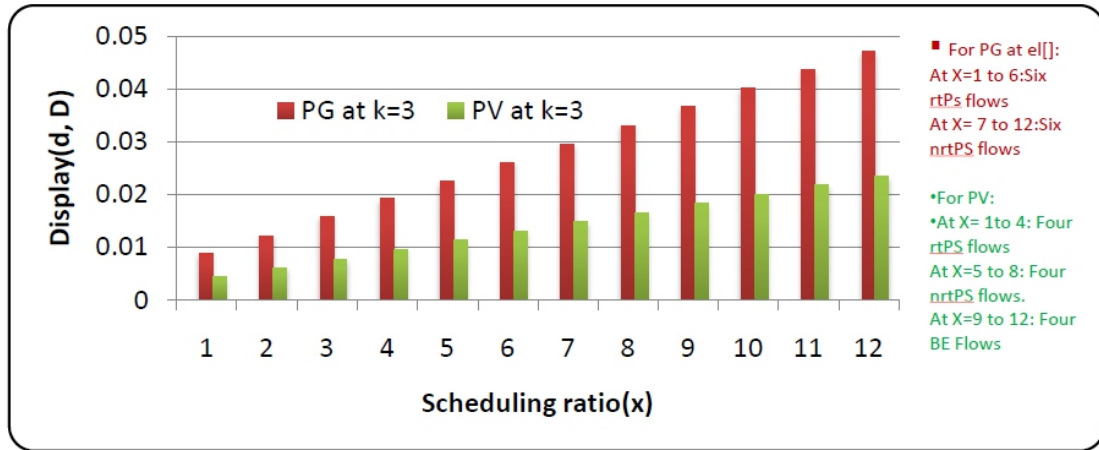
Table 1 for $C = 0.1$ and $C' = 0.2$ (intensity of delay for rtPS, nrtPS and BE connection)													
$k_1 \& k_2/x_i$	$\rightarrow$	1	2	3	4	5	6	7	8	9	10	11	12
$C=0.1$	K&L at $K=k_1=1$	0	0.004	0.008	0.012	0.016	0.02	0.024	0.028	0.032	0.036	0.04	0.044
$C'=0.1$	PG at $K=k_1=1$	0.0029	0.0064	0.0098	0.0133	0.0167	0.0202	0.0237	0.0271	0.0306	0.0341	0.0376	0.038
$C'=0.2$	PV at $K=k_2=1$	0.0014	0.0032	0.0049	0.0066	0.0083	0.0101	0.0118	0.0135	0.0153	0.017	0.0188	0.019
$C=0.1$	PG at $K=k_1=2$	0.0059	0.0093	0.0128	0.0162	0.0197	0.0231	0.0266	0.0301	0.0336	0.0371	0.0406	0.0441
$C'=0.2$	PV at $K=k_2=2$	0.0029	0.0046	0.0064	0.0081	0.0098	0.0115	0.0133	0.015	0.0183	0.0185	0.0203	0.022
$C=0.1$	PG at $K=k_1=3$	0.0088	0.0122	0.0157	0.0192	0.0226	0.0261	0.0296	0.0331	0.0366	0.0401	0.0436	0.0471
$C'=0.2$	PV at $K=k_2=3$	0.0044	0.0061	0.0078	0.0096	0.0113	0.013	0.0148	0.0165	0.0183	0.02	0.0218	0.0235
$C=0.1$	PG at $K=k_1=4$	0.0117	0.0152	0.0187	0.0221	0.0256	0.0291	0.0326	0.0361	0.0396	0.0431	0.0466	0.0501
$C'=0.2$	PV at $K=k_2=4$	0.0058	0.0076	0.0093	0.011	0.0128	0.0145	0.0163	0.018	0.0198	0.0215	0.0233	0.025



Here  $d$  delays representing P.G et al.[3] increase for first six rtPS connections over the same type and no. of flows of Kim and Lim[2] and decrease accordingly for the next six nrtPS connections. However,  $D$  delays representing the P.V's of four rtPS connections increase with the increase in  $x$  for first four connections and also increase for next four connections for nrtPs and again increase for last four connections of BE. However, from the above table we confirm that in view of greater BE delays we observe the mitigation of maximum starvation for the lower priority traffic. For  $k_2=k=1$ , we further analyze here that maximum delays of the rtPS of present first four connections to each of the connections corresponding to Kim and Lim [2] and P.G et al[3] have the variation of 50% and 48% respectively. Delays for next P.V of nrtPS flows with each of the two rtPS and nrtPS flows of Kim and Lim[2] and P.G et al [3] have variation of 54% and 33.3%. Maximum delays of the last four P.V of BE connections with each of the four nrtPS connections of Kim and Lim [2] and P.G et al [3] have variation of 56% and 50%. In conclusion this confirms that if we have to have a fair scheduling for support to better QoS then for mitigation of starvation for the least priority connection we have to meet the requirement of maximum delay to enhance varying between 33.3% and 56% as compared to maximum starvation of BE in K&L[2] and P.G[3].







B. For  $C=0.05$  and  $k_1=1 \dots 4$ .  $d$ , delays of 6 flows of rtPS traffic and 6 flows of nrtPS traffic

And

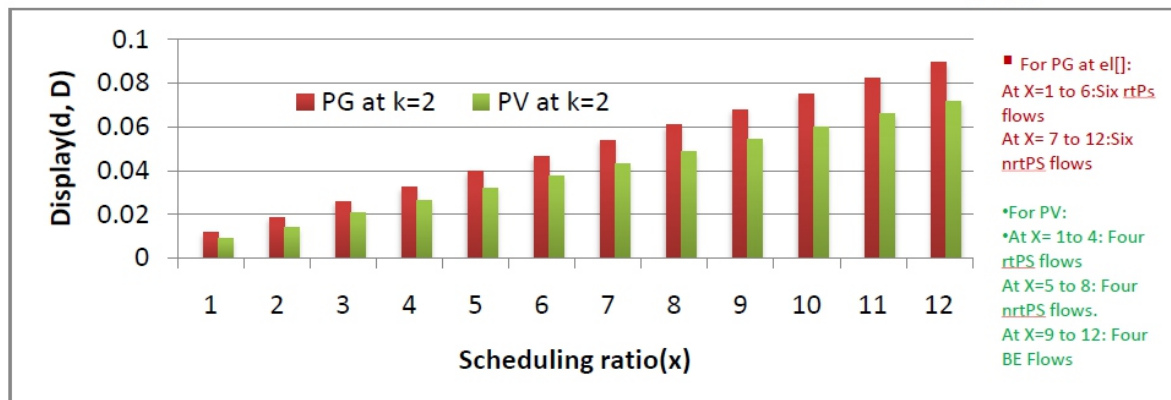
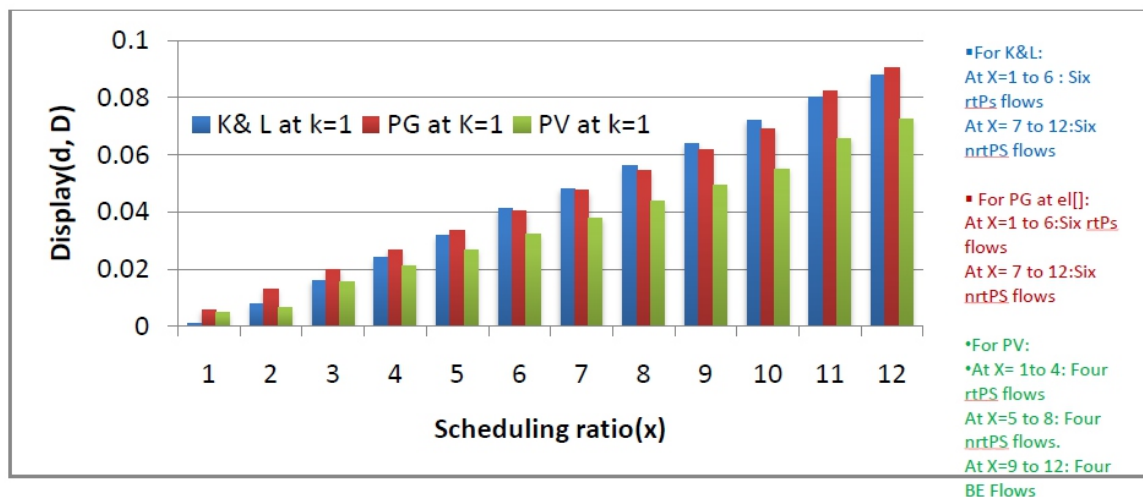
For  $C'=0.06$  for  $k_2=1 \dots 4$ .  $D$  delays of 4 flows of rtPS traffic, 4 flows of nrtPS traffic and 4 flows of BE traffic

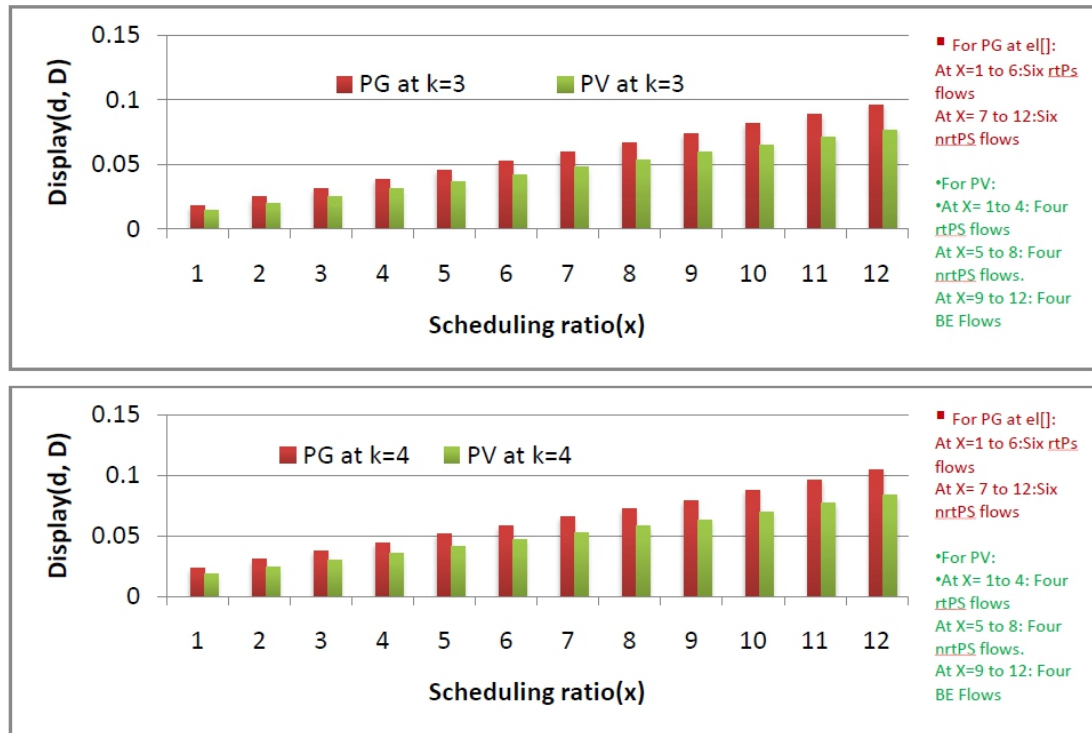
We note here that for P.G,  $k=k_1=1$  and P.V,  $k=k_2=1$ , etc.

In case of  $k=2, 3$  and  $4$  we here discuss the behavior of  $d$  delays of P.G et al [3] and  $D$  delays of P.V corresponding to the same set of values of  $C$  and  $C'$  as given by 6. In all the above three case we find that in case PG et al [3] with the increase of nodes  $x$  varying from 1 to 6, the corresponding rtPS flows in delay  $d$  gradually increase and then for the next values varying from 7 to 12, nrtPS flows in delay  $d$  also gradually increase. Similarly, in case of P.V we find the delay  $D$  steadily increase in first four rtPS flows, also delay  $D$  increase steadily for next four nrtPS flows and finally again delay  $D$  increase steadily for the last four values BE from  $x$  equals 9 to 12. From the above graph it is seen that there is maximum mitigation of starvation happening for nrtPS connections and mostly to the lower priority connection BE.

Table 2 for  $C = 0.05$  and  $C' = 0.06$  (intensity of delay for rtPS, nrtPS and BE connection)

$k1 \& k2 / x_i$	$\longrightarrow$	1	2	3	4	5	6	7	8	9	10	11	12
$C=0.05$	K&L at $K=k1=1$	0	0.008	0.016	0.024	0.032	0.041	0.0481	0.056	0.064	0.072	0.08	0.088
$C=0.05$	PG at $K=k1=1$	0.0059	0.0128	0.0197	0.0266	0.0336	0.0405	0.0475	0.0546	0.0617	0.0689	0.0823	0.0903
$C1=0.06$	PV at $K=k2=1$	0.0047	0.001	0.0157	0.0212	0.0268	0.0324	0.038	0.0436	0.0493	0.0551	0.0658	0.0722
$C=0.05$	PG at $K=k1=2$	0.0117	0.0186	0.0256	0.0325	0.0395	0.0465	0.0535	0.0606	0.0678	0.075	0.0822	0.0894
$C1=0.06$	PV at $K=k2=2$	0.009	0.014	0.0204	0.026	0.0316	0.372	0.0428	0.0484	0.0542	0.06	0.0657	0.0715
$C=0.05$	PG at $K=k1=3$	0.0177	0.0246	0.0315	0.0385	0.0455	0.0526	0.0596	0.0668	0.074	0.0813	0.0886	0.0959
$C1=0.06$	PV at $K=k2=3$	0.0141	0.0196	0.0252	0.0308	0.0364	0.042	0.0476	0.0534	0.0592	0.065	0.0708	0.0767
$C=0.05$	PG at $K=k1=4$	0.0236	0.0305	0.0375	0.0445	0.0515	0.0586	0.0658	0.0728	0.0792	0.0876	0.096	0.1044
$C1=0.06$	PV at $K=k2=4$	0.0188	0.0244	0.03	0.0356	0.0412	0.0468	0.0526	0.0582	0.0633	0.07	0.0768	0.0835





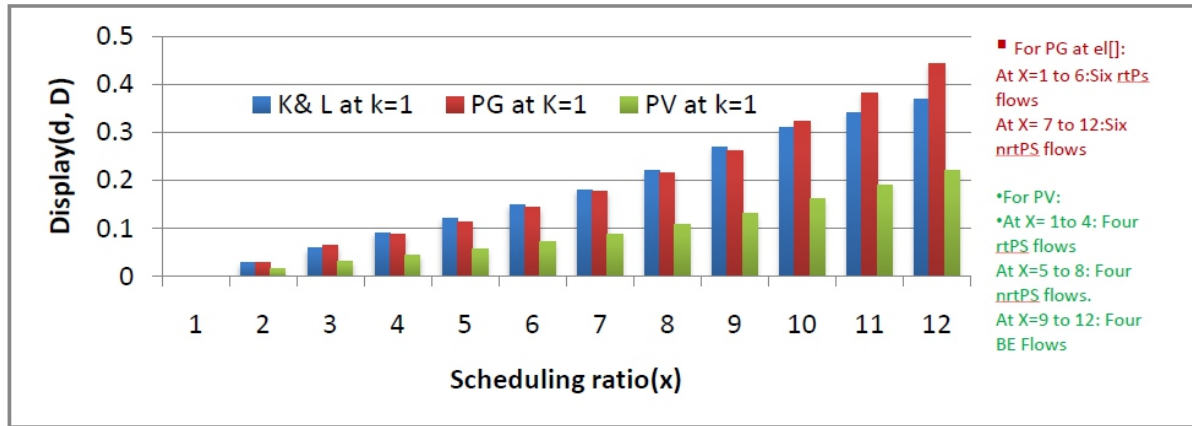
6.3  $C=0.01$  for  $k=1, \dots, 4$ . d, delays of 6 flows of rtPS traffic and 6 flows of nrtPS traffic

And

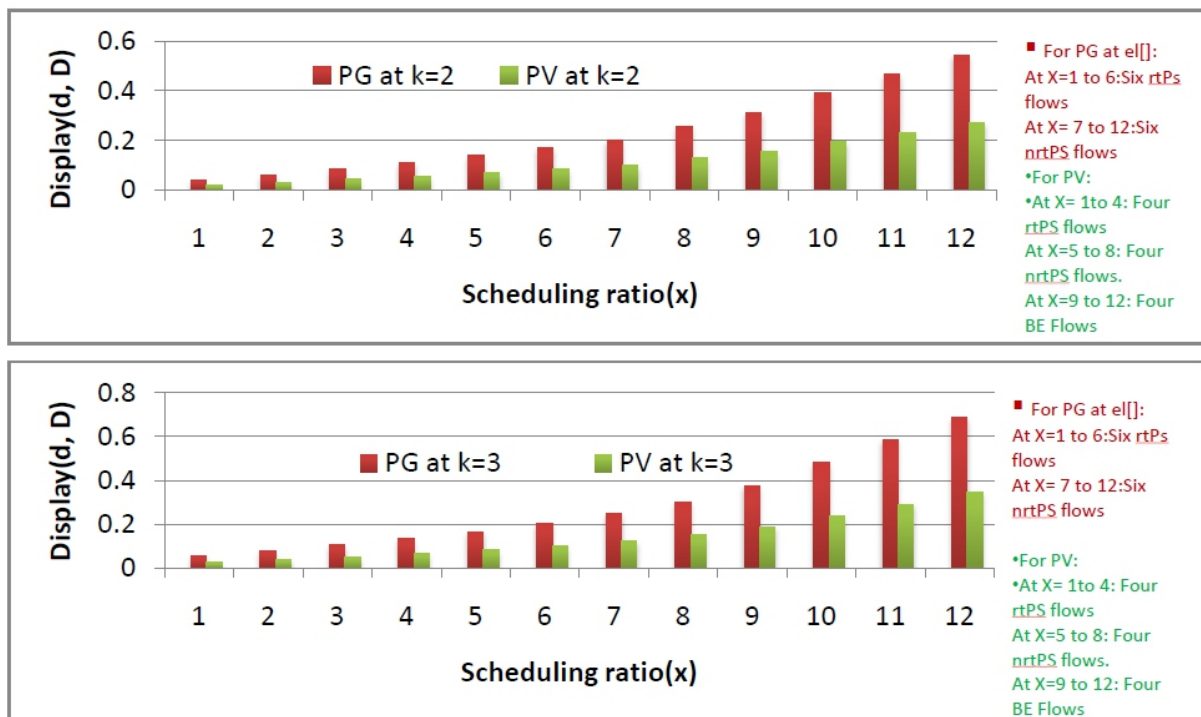
$C'=0.02$  for  $k=1, \dots, 4$ . D delays of 4 flows of rtPS traffic, 4 flows of nrtPS traffic and 4 flows of BE traffic

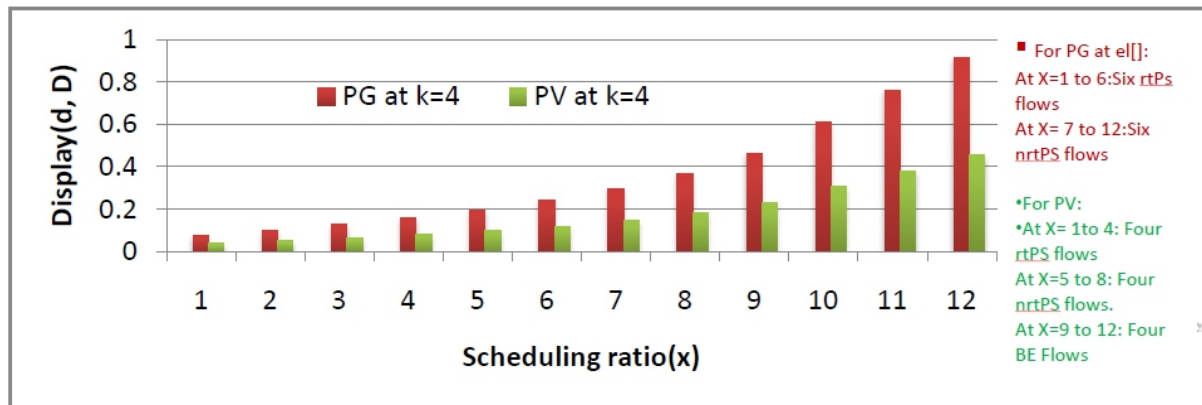
**Table 3 for  $C = 0.01$  and  $C'=0.02$ (intensity of delay for rtPS, nrtPS and BE connection)**

$k_1 \& k_2 / x_i$		1	2	3	4	5	6	7	8	9	10	11	12
$C=0.01$	K&L at $K=k_1=1$	0	0.03	0.06	0.09	0.12	0.15	0.18	0.22	0.27	0.31	0.34	0.37
$C=0.01$	PG at $K=k_1=1$	0.0002	0.03	0.0635	0.0877	0.1139	0.143	0.1761	0.215	0.2622	0.322	0.382	0.442
$C1=0.02$	PV at $K=k_2=1$	0.001	0.015	0.0317	0.0438	0.057	0.0715	0.088	0.1075	0.1311	0.161	0.191	0.221
$C=0.01$	PG at $K=k_1=2$	0.0374	0.0601	0.0841	0.11	0.1386	0.1711	0.1991	0.2548	0.3125	0.3892	0.4659	0.542
$C1=0.02$	PV at $K=k_2=2$	0.0187	0.03	0.042	0.055	0.0693	0.0855	0.0995	0.1274	0.1562	0.1946	0.2329	0.271
$C=0.01$	PG at $K=k_1=3$	0.0568	0.0806	0.1062	0.1343	0.1662	0.2032	0.2476	0.3034	0.3767	0.4803	0.5839	0.687
$C1=0.02$	PV at $K=k_2=3$	0.0284	0.0403	0.0531	0.0671	0.0831	0.1016	0.1238	0.1517	0.1883	0.2401	0.2919	0.343
$C=0.01$	PG at $K=k_1=4$	0.0771	0.1024	0.1301	0.1614	0.1975	0.2408	0.2946	0.3649	0.463	0.6131	0.7632	0.913
$C1=0.02$	PV at $K=k_2=4$	0.0385	0.0512	0.065	0.0807	0.0987	0.1204	0.1473	0.1824	0.2315	0.3065	0.3816	0.456



Again, in case of  $k=2,3$  and  $4$  we here discuss the behavior of  $d$  delays of P.G et al [3] and  $D$  delays of P.V corresponding to the same set of values of  $C$  and  $C'$  as given by 6.3. In all the above three case we find that in case PG et al [3] with the increase of nodes  $x$  varying from 1 to 6, the corresponding rtPS flows in delay  $d$  gradually increase and then for the next values varying from 7 to 12, nrtPS flows in delay  $d$  also gradually increase. In case of P.V we find the delay  $D$  steadily increase in first four rtPS flows, also delay  $D$  increase steadily for next four nrtPS flows and finally again delay  $D$  increase steadily for the last four values BE from  $x$  equals 9 to 12. From the above graph it is seen that there is maximum mitigation of starvation happening for nrtPS connections and mostly to the lower priority connection BE.





## 7. CONCLUSION

A novel proportional fair based QoS scheduling was designed under traffic conditions. The simulations are carried out and from that traffic includes majority of rtPs connections having excellent performance. When the traffic connections have nrtPS and BE as the majority requests then the performance of these traffics improve. The performance of rtPS and connections sharing fairness metrics has an improved performance with regard to the mitigation of starvation for nrtPS and BE traffics. The BE and nrtPS however hovers around 95% even though the rtPS traffic still has maximum utilization. We further notice here that the mitigation of lower priority schemes depend on respective suitable values of the intensity delays of C and C' as discussed in above fair scheduling.

Thus from the fairness scheduler it is inferred that the proposed rtpS traffic has almost 100% at all traffic nodes satisfying the WiMax QoS requirements. Using downlink scheduling, it does fair management of lower class services such as nrtPS and BE with the help of rtPS as BS.

## 8. ACKNOWLEDGEMENT

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# Multi Sink Scheduling Scheme For Wireless Sensor Networks

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## **ABSTRACT**

*In wireless sensor networks WSN increasing the network lifetime, where the information delay caused by moving the sink should be bounded. Some of the combinational complexity of this problem, most previous proposals focus on heuristics and provable optimal algorithms remain unknown. By build a unified framework for analyzing this joint sink mobility, routing, delay, and induced sub problems and present efficient solutions for them. We generalize these solutions and propose a polynomial-time optimal algorithm for the origin problem. Furthermore, we developing with multiple sink for the network and study the effects of different trajectories of the sink and provide important insights for designing mobility schemes in real-world mobile WNNs.*

**Index Terms**—Wireless sensor networks, delay-constrained mobility, network lifetime

## **1. INTRODUCTION**

In the past decades, wireless sensor network (WSN), one of the fastest growing research areas, has been attracted a lot of research activities. Due to the maturity of embedded computing and wireless communication techniques, significant progress has been made. Typically, a WSN consists of a data collection unit (also known as sink or base station) and a large number of sensors that can sense and monitor the physical world, and thus it is able to provide rich interactions between a network and its surrounding physical environment in a real-time manner.

The capacity-limited power sources of small sensors constrain us from fully benefitting from WSNs. Due to the unique many-to-one (converge-cast) traffic patterns, the traffic of the whole network will be converged to a specific set of sensor nodes (e.g., neighbouring nodes of the sink) and results in the hotspot problem . Much research effort has been dedicated to resolve this issue, for example, energy efficient communication protocols, multi-sink systems. However, as long as the sink and sensor nodes are static, this issue cannot be fully tackled. Therefore, there is arecent trend to exploit mobility of the sink as a promising approach to the hotspot problem.

By the way of using sink mobility, we can classify them into two categories: random mobility based and controlled mobility based. For the first category, the sink is designed to move randomly within the network. For example, Rahul et al. presented an architecture on which mobile entities (named MULEs) pick up data from sensors when in close range in sparse sensor networks. Schemes based on random mobility are straightforward and easy to implement. However, they suffer from shortcomings like uncontrolled behaviours and poor performance. Hence, recent research resorts to controlled mobility to improve the performance.

For the controlled mobility, the key problem is to deterministically schedule the sink to travel around the network to collect data. It is shown that by properly setting the trajectory even limited mobility would significantly improve the network lifetime. However, the mobility also brings new issue, i.e., the delay of the data delivery caused by the movement of the sink. Some previous proposals tried to avoid this issue by considering the so-called fast mobility, whereas the speed of the sink is sufficiently high so that the resulting delay can be tolerated. While others address this delay-bounded mobility problem by heuristics with little theoretical understanding.

To this end, we study the delay-bounded sink mobility problem (DeSM) of WSNs in this paper. We assume that WSNs are deployed to monitor the surrounding environment and the data generation rate of sensors can be estimated accurately. We constrain the mobile sink to a set of sink sites. First, we propose a unified framework that covers most of the joint sink mobility, data routing, and delay issue strategies. Based on this framework, we develop a mathematical formulation that is general and captures different issues. However, this formulation is a mixed integer nonlinear programming (MINLP) problem and is time consuming to solve directly. Therefore, instead of tackling the MINLP directly, we first discuss several induced sub problems, for example, sub problems with zero/infinite delay bound or connected sink sites (sink sites are connected if for any two sites there exists a path that connects them and each edge of that path meets the delay constraint). We show that these sub problems are tractable and present optimal algorithms for them. Then, we generalize these solutions and propose a polynomial-time optimal approach for the origin DeSM problem. We show the benefits of involving a mobile sink and the impact of network parameters (e.g., the number of sensors, the delay bound, and so on.) on the network lifetime. Furthermore, we study the effects of different trajectories of the sink and provide important insights for designing mobility schemes in real-world mobile WSNs.



Our main contributions are the following:

1. We provide a unified formulation of DeSM, which is general and practical. We discuss sub problems of DeSM and offer efficient algorithms for them to guide the design of our algorithm for the origin DeSM.
2. We generalize algorithms for sub problems and present an optimal algorithm with polynomial complexity for the DeSM.
3. We study the effects of different trajectories of the sink and provide important insights via extensive simulations.

## **2. RELATED WORK**

Mobility management is one of the most important issues in wireless networks, and it has received extensive research efforts in different areas of wireless networks such as mobile ad hoc network (MANET), wireless mesh network, vehicular ad hoc network.

Recently, there is a trend to investigate mobility as a means of relieving traffic burden and enhancing energy efficiency in WSN. We can classify sink mobility into two categories: random mobility and controlled mobility. Sinks in the first category move randomly within the network. Schemes based on random mobility are easy to implement, but they suffer from shortcomings like uncontrolled behaviors and poor performance. Recent research tends to use controlled mobility to improve the performance. The hardcore is to jointly schedule different issues (e.g., sink mobility, data routing, information delay, and so on.) to optimize the network lifetime.

For this paradigm, Gandham et al. first challenged this problem and proposed a heuristic algorithm. Wang et al. relaxed the problem by doing the sink scheduling and data routing separately, and their proposed routing scheme can work only in a grid network topology.

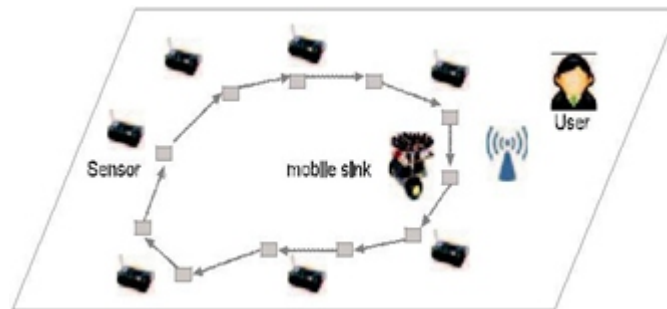
Recently, Shi and Hou developed the first algorithm with performance guarantee with a single sink. Liang et al. extended Shi's work by considering issues like multiple sinks and the maximum number of hops from each sensor to a sink. A three-stage heuristics has been developed to find high-quality trajectory for each sink as well as the actual sojourn time at each sojourn location. In our recent research, we proposed a generalized column generation-based algorithm that can be applied to a set of sink mobility problems with near-optimal performance.

In above proposals, they assume that sinks are high-speed so that information delay caused by moving the sink can be ignored. However, on the one hand, mobile sinks in physical worlds usually have limited speed. On the other hand, underlay applications like the real-time surveillance demand a delay upper bound. Therefore, it is natural to take the delay issue into consideration.

Keung et al. studied the message delivery capacity problem in delay-constrained mobile sensor networks where the sink nodes are static while sensor nodes are mobile. They focused on maximizing the percentage of sensing messages that can be successfully delivered to sink nodes within a given time constraint. Their network model is fundamentally different with ours and is somehow similar to the DTN. In our previous study, we also addressed the problem of lifetime maximization with delay bound in a mobile WSN. The major improvements of this paper over the previous one are twofold. First, we present some new theoretical results like the connectivity analysis of a WSN. Second, we design a set of new simulations to study the effects of different trajectories of the sink and provide important insights for designing mobility schemes in real-world mobile WSNs.

### 3 DeSM PROBLEM

Fig. 1 shows reference architecture for a WSN with a mobile sink (i.e.,  $s_0$ ). Sensor nodes, which are stationary, keep monitoring the surrounding environment and generating data. A mobile sink is used to gather sensed data by traveling around the network. We assume that



**Fig. 1.** Reference architecture for a WSN with a mobile sink.

Only at certain locations, the sink can communicate with the outside network and then deliver cached data to users. For example, due to interference and security issues, for a sensor network deployed in the battle field for the surveillance mission, it is reasonable that the sink can connect with the headquarters only at certain locations using wireless techniques like WiMAX or LTE. These locations are represented by squares in the figure. The sink has a maximum speed  $V_{\max}$  (in m/s). We assume that while the sink is moving, sensors<sup>1</sup> will buffer their newly generated data, as in. Only when the sink stays at one

sites, sensors will start transmitting data to the sink through multi-hop routing. This could potentially cause a high delay for data packets. Here, we define the delay of data as following,

**Definition 1** (Delay of data).

The delay of data is defined as the time spent by the mobile sink moving from one sink site to the next sink site.

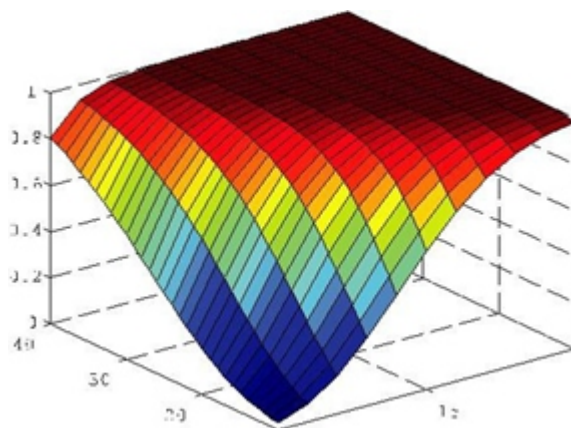
To limit such delay, a delay bound is set according to the underlay applications. Moreover, as pointed out in the previous study, whenever a sink has been relocated to a new site, it will take some time to rebuild the routes of sensors. Thus, we set  $\epsilon$  as the minimum residual time of any sink site.

## 4 EXTENDED SSDR (E-SSDR) ALGORITHM FOR DESM

### 4.1 E-SSDR Algorithm

To solve the origin DeSM problem, we prove the following conclusion: For an instance of DeSM, if its sink site graph  $G^0$  is not connected, we can divide  $G^0$  into connected subgraphs, each of which can be solved optimally by the SSDR. The overall optimal solution for this instance is the same solution of the sub graph with the longest network lifetime.

The proof is based on contradiction. Assume that for an instance of DeSM, we have an optimal solution which involves two sites from two different sub graphs. This means that we find a sink path including these two sites that meets the delay constraint. Thus, these two sites are connected and should be in the same sub graph.



**Fig. 2.** Probability of the full connection

we propose an E-SSDR approach to solve the origin DeSM optimally:

**Step 1.** Divide  $G^0$  into connected sub graphs.

**Step 2.** Apply the SDDR approach to each sub graphs and obtains the optimal sink path as well as corresponding routes.

**Step 3.** Choose the solution of the sub graph with the longest network lifetime as output.

## 5 NUMERICAL RESULTS

In this part, we evaluate the proposed algorithms using three typical trajectories of the sink, namely:

1. Linear trajectory. This case simulates that the sink travels along one predefined path, for example, a vehicle carrying a sink moves along the only path across the forest to gather sensed data daily.
2. Boundary trajectory. Luo and Hubaux suggested that it is the most efficient way to collect data in a dense network.
3. Arbitrary trajectory. In this case, we have little control over the distribution of sink sites, for example, in a battle field. Due to page limit, we prepare a supplement file, available online, for the simulation results the arbitrary trajectory.

## 6 CONCLUSION AND FUTURE WORK

We proposed a unified framework to analyze the sink mobility problem in WSNs with delay constraint. We presented a mathematical formulation that jointly considers different issues such as sink scheduling, data routing, bounded delay, and so on. The formulation is general and can be extended. However, this formulation is a MINLP and is time consuming to solve directly. Therefore, we discussed several induced subproblems and developed corresponding optimal algorithms. Then, we generalized these solutions and proposed a polynomial-time optimal approach for the origin problem. We show the benefits of involving a mobile sink and the impact of network parameters (e.g., the number of sensors, the delay bound, and so on.) on the network lifetime. Furthermore, we study the effects of different trajectories of the sink and provide important insights for designing mobility schemes in real-world mobile WSNs.

As for the future work, we plan on extending current work to accommodate networks with multiple sinks. Furthermore, using the centralized optimal algorithm developed in this paper as performance benchmark, we want to design distributed online algorithms for fast execution in large-scale networks and test them in real-world experiments.

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## Path Finding: A\* or Dijkstra's?

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### ABSTRACT

*It is well known that computing shortest paths over a network is an important task in many network and transportation related analyses. Choosing an adequate algorithm from the numerous algorithms reported in the literature is a critical step in many applications involving real road networks. In a recent study, a set of two shortest path algorithms that run fastest on real road networks has been identified. These two algorithms are: 1) the A\* algorithm, 2) the Dijkstra's algorithm. As a sequel to that study, this paper reviews and summarizes these two algorithms, and demonstrates the data structures and procedures related to the algorithms.*

***Keywords - A\* algorithm, Dijkstra's Algorithm, Heuristic Function, Geographic Information System, Transportation.***

### Introduction

With the development of geographic information systems (GIS) technology, network and transportation analyses within a GIS environment have become a common practice in many application areas. A key problem in network and transportation analyses is the computation of shortest paths between different locations on a network. Sometimes this computation has to be done in real time. For the sake of illustration, let us have a look at a situation in which a place has been hit with natural disaster. Now many persons may get stuck at different places but they have to be rescued anyhow. So, the main problem that rescue team face is that how they (rescuer) approach them i.e. which path they should opt from numerous available paths. Hence, the fastest route can only be determined in real time. In some cases the fastest route has to be determined in a few seconds in order to ensure the safety of a stuck people. Moreover, when large real road networks are involved in an application, the

determination of shortest paths on a large network can be computationally very intensive. Because many applications involve real road networks and because the computation of a fastest route (shortest path) requires an answer in real time, a natural question to ask is: *Which shortest path algorithm runs fastest on real road networks?* As there are many algorithm available to find shortest path so there is no clear answer. Here we have discussed about two algorithm, A\* and Dijkstra's.

### Description of Algorithms

Dijkstra's algorithm: Before going into details of the pseudo-code of the algorithm it is important to know how the algorithm works. Dijkstra's algorithm works by solving the sub problem  $k$ , which compute the shortest path from source to vertices among the  $k$  closest vertices to the source. For the Dijkstra's algorithm to work it should be directed-weighted graph and the edges should be non-negative. If the edges are negative then the actual shortest path cannot be obtained.

The algorithm works by keeping the shortest distance of vertex  $v$  from the source in an array, Dist. The shortest distance of the source to itself is zero. Distance for all other vertices is set to infinity to indicate that those vertices are not yet processed. After the algorithm finishes the processing of the vertices Dist will have the shortest distance of vertex from source to every other vertex. Two sets are maintained which helps in the processing of the algorithm, in first set all the vertices are maintained that have been processed i.e. for which we have already computed shortest path. And in second set all other vertices are maintained that have to be processed.

The above algorithm can be explained and understood better using an example. The example will briefly explain each step that is taken and how Dist is calculated.

Consider the following example:

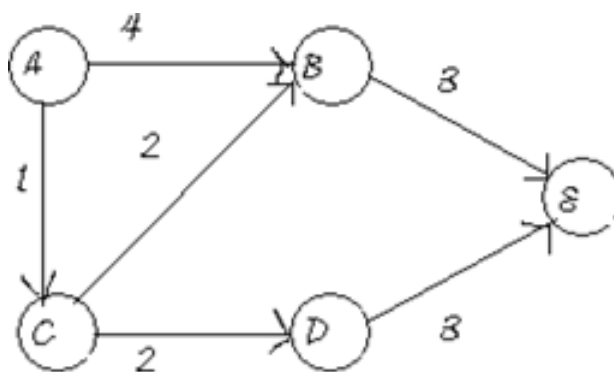


Figure 1(a) Step 1

The example is solved as follows:

### Initial step

$\text{Dist}[A]=0$ ; *the value to the source itself*

$\text{Dist}[B]=\text{infinity}$ ,  $\text{Dist}[C]=\text{infinity}$ ,  $\text{Dist}[D]=\text{infinity}$ ,  $\text{Dist}[E]=\text{infinity}$ ; *the nodes not processed yet*

### Step 1

$\text{Adj}[A]=\{B,C\}$ ; *computing the value of the adjacent vertices of the graph*

$\text{Dist}[B]=4$ ;  $\text{Dist}[C]=2$ ;

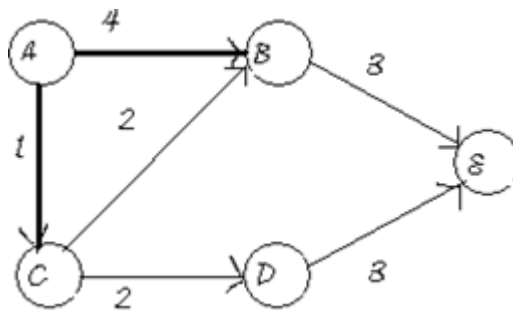


Figure 1(b) Step 2

*Figure: shortest path to vertices B, C from A*

### Step 2

*Computation from vertex C*

$\text{Adj}[C]=\{B,D\}$ ;

$\text{Dist}[B] > \text{Dist}[C] + \text{EdgeCost}[C,B]$   $4 > 1+2$  (*True*)

Therefore,  $\text{Dist}[B]=3$ ;  $\text{Dist}[D]=2$ ;

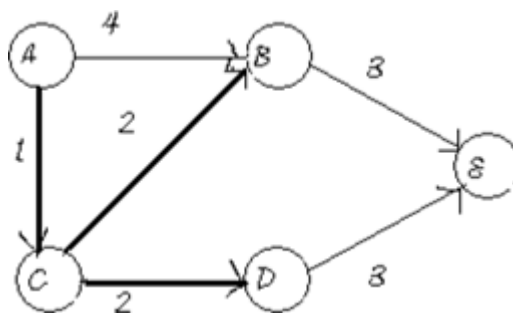


Figure 1 (c) Step 3

*Figure: Shortest path from B, D using C as intermediate vertex*

$\text{Adj}[B]=\{E\}$ ;



$$\text{Dist}[E] = \text{Dist}[B] + \text{EdgeCost}[B, E]$$

$$= 3 + 3 = 6;$$

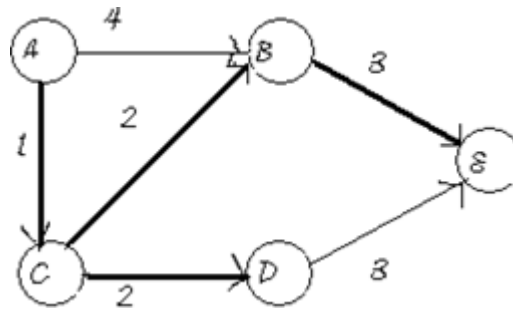


Figure 1 (d) Step 4

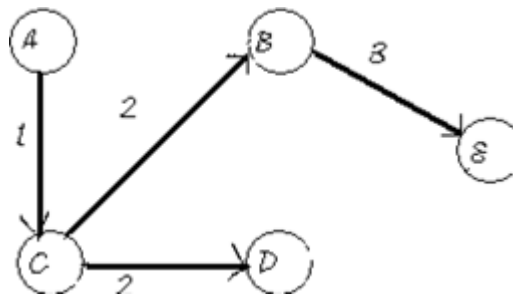


Figure 1 (e) Step 5

Figure 1 Shortest Path using Dijkstra's Algorithm

$$\text{Adj}[D] = \{E\};$$

$$\text{Dist}[E] = \text{Dist}[D] + \text{EdgeCost}[D, E]$$

$$= 3 + 3 = 6$$

This is same as the initial value that was computed so  $\text{Dist}[E]$  value is not changed.

### Step 3

$\text{Adj}[E] = 0$ ; means there is no outgoing edges from E

And no more vertices, algorithm terminated. Hence the path which follows the algorithm is ACBE.

**A\* Algorithm:** A\* algorithm is a graph search algorithm that finds a path from a given initial node to a given goal node. It employs a "heuristic estimate"  $h(x)$  that gives an estimate of the best route that goes through that node. It visits the nodes in order of this heuristic estimate. It follows the approach of best first search. The secret to its success is that it combines the pieces of information that Dijkstra's algorithm uses (favoring vertices that are close to the starting point) and information that Best-First

Search uses (favoring vertices that are close to the goal). In the standard terminology used when talking about A\*,  $g(n)$  represents the exact cost of the path from the starting point to any vertex  $n$ , and  $h(n)$  represents the heuristic estimated cost from vertex  $n$  to the goal.

Let's assume that we have someone who wants to get from point A to point B. Let's assume that a wall separates the two points.

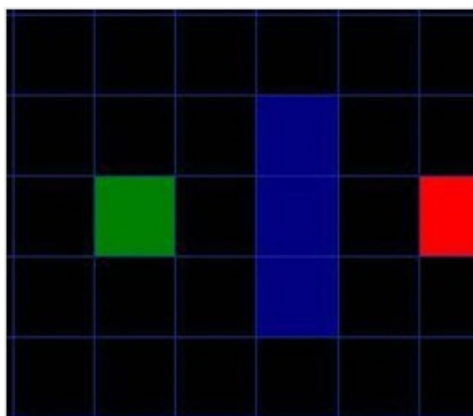


Figure 2 (a) Step 1

### Initial Step

The map has a starting point, an ending point and some obstacles. Green is the starting point, Red is the ending point and Blue are the obstacles.

### Step 1

We start by searching the 8 neighboring nodes of the Starting point.

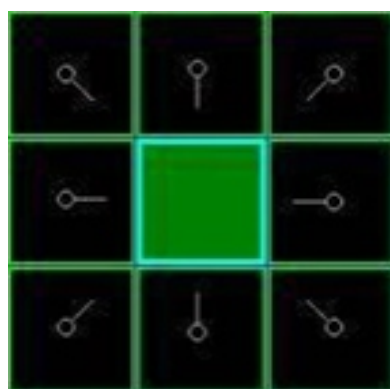


Figure 2 (b) Step 2

### Step 2

We calculate the Heuristics of those neighboring cells.  $F=G+H$ .

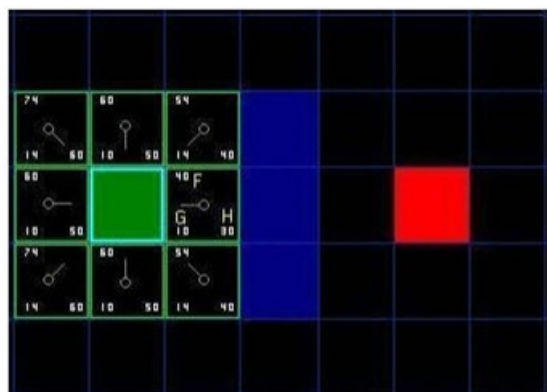


Figure 2 (c) Step 3

We proceed by choosing the nodes that are closer to the ending point than others.

Figure 2 (d) Step 4

The Red dots are the path that is chosen.

Figure 2 (e) Step 5

Heuristic Function: A heuristic is a technique that improves the efficiency of search process, possibly by sacrificing claims of completeness. While the almost perfect heuristic is significant for theoretical analysis, it is not common to find such a heuristic in practice. Heuristics play a major role in search strategies because of exponential nature of the most problems. Heuristics help to reduce the number of alternatives from an exponential number to a polynomial number. Heuristic search has been widely used in both deterministic and probabilistic planning. Heuristic functions generally have different errors in different states. Heuristic functions play a crucial rule in optimal planning, and the theoretical limitations of algorithms using such functions are therefore of interest. Much work has focused on finding bounds on the behavior of heuristic search algorithms, using heuristics with specific attributes.

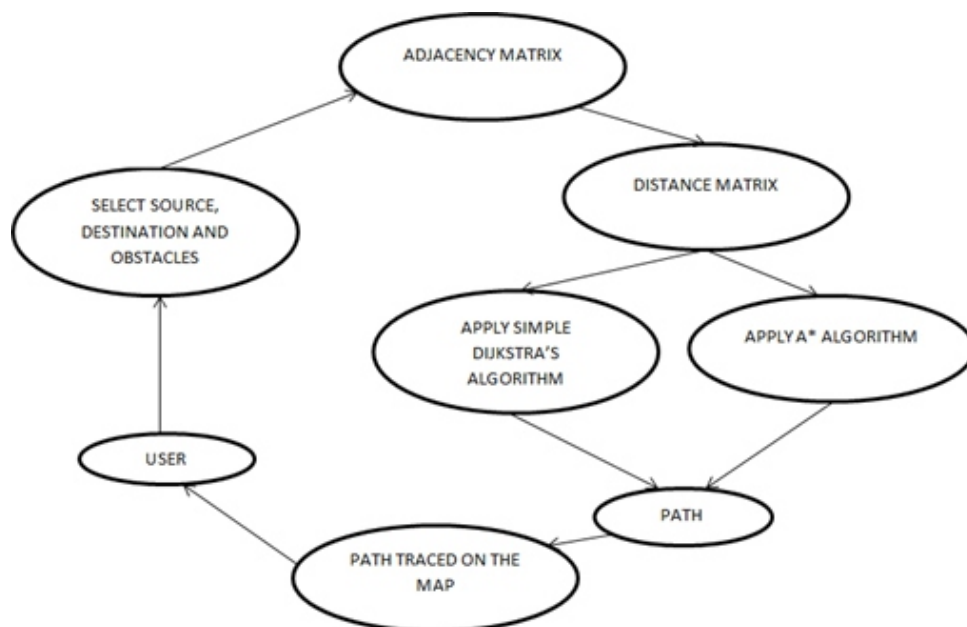


Figure 2 Flow Diagram of A\* and Dijkstra's  
Comparison of A\* and Dijkstra's

Both the algorithms find the shortest distance between two nodes. Now, we would try to implement A star and Dijkstra's on distributed system and compare them on running time and try and show that A star is a better than Dijkstra's.

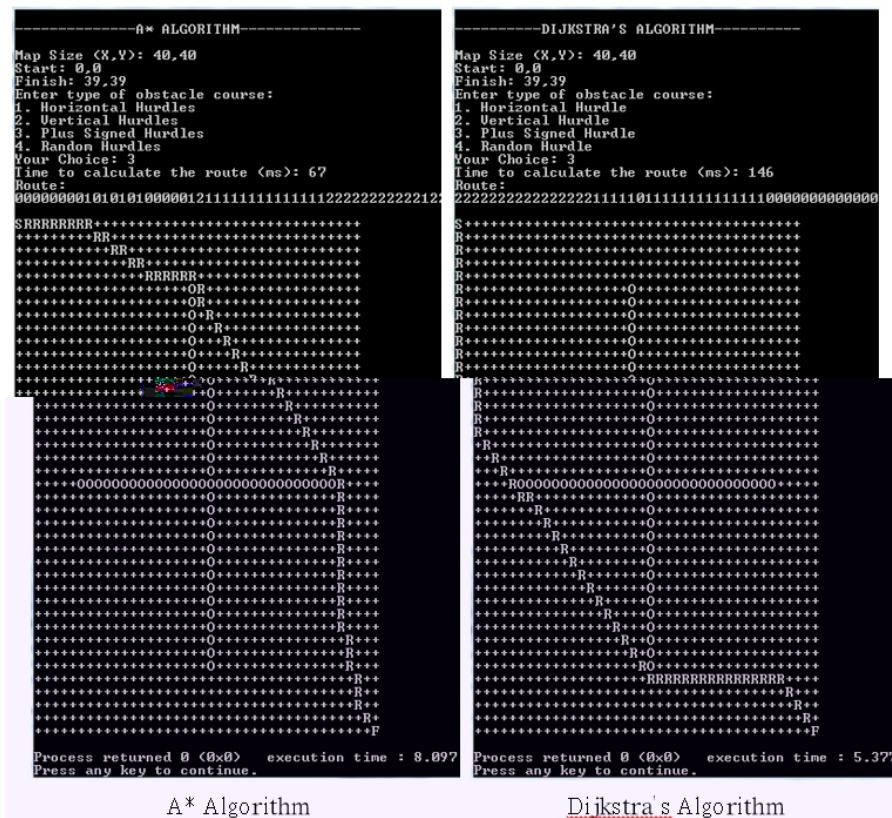
First of all let us cite the major differences between the two.

Note: Dijkstra's Algorithm is special case of A\* Algorithm, when  $h(n)=0$ .

**Table 1** Difference between A\* Algorithm and Dijkstra's Algorithm

Parameters	A* Algorithm	Dijkstra's Algorithm
Search Algorithm	Best First Search	Greedy Best First Search
Time Complexity	Time complexity is $O(n \log n)$ , $n$ is the no. of nodes.	The time complexity is $O(n^2)$ .
Heuristics Function	Heuristic Function, $f(n)=g(n)+h(n)$ , $g(n)$ represents the cost of the path from the starting point to the vertex $n$ . $h(n)$ represents the heuristic estimated cost from vertex $n$ to the $g$ .	$f(n)=g(n)$ , $g(n)$ represents the cost of the path from the starting point to the vertex $n$ . Dijkstra's Algorithm is the worst case of A star Algorithm.

We would implement A\* and Dijkstra's on distributed system and compare their running time, keeping the constraints same as much as possible. So that it can be proven that A\* is better than Dijkstra's.

**Figure 3** Case 1 of Comparison between A\* Algorithm and Dijkstra's Algorithm

In this scenario, as you can see, the time taken by A\* Algorithm to solve the problem is nearly half the time taken by Dijkstra's Algorithm to solve the same problem.

We are going to illustrate another example.

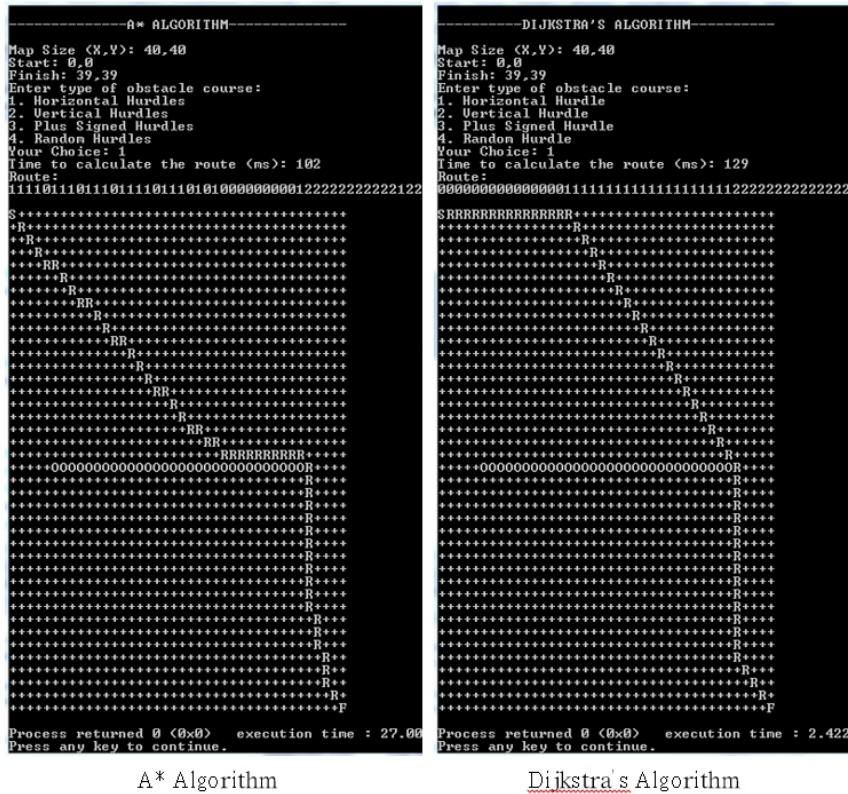


Figure 4 Case 2 of Comparison between A\* Algorithm and Dijkstra's Algorithm

Table 2

COMPARISION IN TIME (ms)			
A* ALGORITHM		DIJKSTRA'S ALGORITHM	
CASE 1	67	CASE 1	146
CASE 2	102	CASE 2	129
COMPARISION IN PATH LENGTH (NODES)			
A* ALGORITHM		DIJKSTRA'S ALGORITHM	
CASE 1	55	CASE 1	55
CASE 2	55	CASE 2	55

As it can be seen, the lengths of paths do not vary in both the cases, but the time varies largely. This implies that, though both the algorithms provide the shortest path of equal lengths, the Dijkstra's Algorithm scans more part of the map than the A\* Algorithm does. Hence in these two cases and some others, A\* Algorithm is better than the Dijkstra's Algorithm.

## Advantages and Disadvantages

A\* is faster as compare to Dijkstra's algorithm because it uses Best First Search whereas Dijkstra's uses Greedy Best First Search.

Dijkstra's is Simple as compare to A\*.

The major disadvantage of Dijkstra's algorithm is the fact that it does a blind search there by consuming a lot of time waste of necessary resources.

Another disadvantage is that it cannot handle negative edges. This leads to acyclic graphs and most often cannot obtain the right shortest path.

Dijkstra's algorithm has an order of  $n^2$  so it is efficient enough to use for relatively large problems.

The major disadvantage of the algorithm is the fact that it does a blind search there by consuming a lot of time waste of necessary resources.

## Conclusion

Dijkstra's is essentially the same as A\*, except there is no heuristic ( $H$  is always 0). Because it has no heuristic, it searches by expanding out equally in every direction, but A\* scan the area only in the direction of destination. As you might imagine, because of this Dijkstra's usually ends up exploring a much larger area before the target is found. This generally makes it slower than A\*. But both have their importance of its own, for example A\* is mostly used when we know both the source and destination and Dijkstra's is used when we don't know where our target destination is. Say you have a resource-gathering unit that needs to go get some resources of some kind. It may know where several resource areas are, but it wants to go to the closest one. Here, Dijkstra's is better than A\* because we don't know which one is closest. Our only alternative is to repeatedly use A\* to find the distance to each one, and then choose that path. There are probably countless similar situations where we know the kind of location we might be searching for, want to find the closest one, but not know where it is or which one might be closest. So, A\* is better when we know both starting point and destination point. A\* is both complete (finds a path if one exists) and optimal (always finds the shortest path) if you use an Admissible heuristic function. If your function is not admissible off - all bets are off.



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# QoS Issues in Implications of Voice over Venerable Networks

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## **ABSTRACT**

*The wireless based networking technology is an efficient way to disseminate data across the globe without physical connectivity between sender and receiver. Various type of wireless networking technologies viz. Ethernet, Wi-Fi and other IEEE 802 standard technologies have been deployed successfully for such data transmission over airing last two decades and have proved a huge success. Today, due to rapid proliferation in technology on hardware and software front both client demand pattern and data need change drastically. Now we need to deliver very large size data containing audio / video contents and other real time control applications simultaneously with time bar on delivery. Nevertheless, the above-mentioned wireless technologies lack integrated approach to deliver this entire thing simultaneously with standard quality and quantity. This lack in integrated approach and demand of huge data delivery within time motivated the networking community to design IEEE 802 standard for “time sensitive networks”. Such network provides vast number of connectivity applications but still have challenge on Quality of Service (QoS) in data delivery hence are more venerable in comparisons to previous networks. In this paper, we have discussed the issue of delivering voice data over such network and how to measure the Quality of Service (QoS) of voice over these networks.*

**Keywords**-time sensitive networks, venerable networks, wireless technology, QoS of voice in wireless network.

## **I. INTRODUCTION**

Wireless technology came in existence for the military applications that uses wireless data to connect the nodes in the region where cabling is not possible. Subsequently wireless network has twisted a network that covenants with the data transmission at radio frequencies. Wireless network entail of peer-to-peer networks where each computer openly interconnect with the other without wire. With the proliferation of cell phones, laptop and computers etc. demand of wireless network has grown up day by day. At inception, the wireless technology was slow, expensive and reserved for unreceptive environment

where cabling is difficult but now a complete range of IEEE 802 standard protocols is available and convenient in communicating audio, video and voice data that needs to interconnect within the assortment of the wireless network.

Video and voice are real time applications and with the augmentation of the internet more and more voice data sent over the wireless networks like voice over internet network, voice over ATM, voice over frame relay and voice over Ethernet. For all such real time applications, time is very important. Even the millisecond delay in information may cause huge catastrophe and information may lose its significance. Hence, the wireless network for real time application must be time sensitive. Time Sensitive Network for real time media is control stream that are used to automotive or industrial facilities. The main issues of the Time Sensitive network viz. firstly to identify the characteristics of the traffic of data, secondly to identify the hardware and software configuration details with their limitations and large network is scrabbling down into the small assessment facing many critical time synchronization issues. In real time systems, the information is processed within the specified time.

Real time system can broadly categorized in to two categories viz. *hard and soft real time systems*. Hard real time system must be completed within the time. The deadline imposed with the applications, so that any failure cannot be associated with the processed applications. Likewise soft real time system still functions where deadline not met. We have many protocols that are used in real time application but these protocol have some drawbacks that's why these protocols are not used in every field of real time application. So demand of huge data delivery in voice communication we to design IEEE 802 standards for Time Sensitive Networks. In order to make a wireless network time sensitive M.J. Taneer [1] has suggested three important additions to IEEE 802 architecture:

- (i) Time awareness of network by universal time synchronization.
- (ii) Time sensitive queuing of data with guaranteed timely forwarding and delivery of data.
- (iii) Bandwidth and latency reservation of wireless channel to provide dedicated path for time sensitive data.

In the primarily stage the time sensitive network will work from the identification of traffic characteristics so that time synchronization of devices routers and switches on the network. Large network consist of small test it will leads to the dropped packets in voice i.e. measured by the bandwidth that is reserved for the time sensitive queues. The rest of this paper organized as follows: section II gives details of the work done by various technocrats and scientists in the field of time sensitivity in wireless networks. Section III gives problem definition of the time sensitive network. Section IV gives of QoS

issues in time sensitive applications. Section V consists of implication of voice in data networks. Section VI follows how to measure the QoS of voice in networks and VII follows conclusion and future work.

## II. RELATED WORK

S. Soulhi [2] in his paper titled “Telephony over wireless networks” has analyzed and compares both traditional and packet based cellular telephone services. He established that traditional telephone services are based on circuit switching and a dedicated path is established that is reserved to the caller. Its major weakness is pulse code modulation (PCM) voice. But packet based data networks improved profitability and productivity in business communication by simultaneously transmitting voice and data packets on same channel. Despite that the many issues still left to consider like quality of service routing and security issues. J. Kim et al [3] studied characteristics of voice quality on the internet protocol network as per voice over internet protocols services. They provide a method to measure the quality of network not only with the internet protocol network but also with the gateway and gatekeeper also and concludes that continuous development in the wireless network need efficient measurement and performance of a network. L. Angrisani [4] measured the IPDV values and performance matrices, perceiving the quality at application layer by the user and applications of voice over internet protocols. Voice quality is measured at application layer of the ISO/OSI model. C. Mancas and M. Mocanu [5] considered performance of traffic types queuing and resource reservation protocol and issues in Quality of service (QoS) affected by delaying packets due to the parameters like latency, bandwidth, jitter, packet loss and echo. They found that when guaranteed bandwidth is not provided, it will lead to the congestion resulting in problem for the system. T. J. Kostas et al [6] provides architecture and technical viability of real time voice over packet switched networks. They examine the architecture for voice over internet protocol that measuring internet delay and loss characteristics of the data.

Z. Han et al [7] gives a user satisfaction factor (USF) that provide quality of service (QoS) on type of services such as audio voice data and multimedia data. High system performance for different data, modifying scheduling schemes also measured delay of sensitive application. They have also designed further generalized terms for the future wireless time sensitive network. Siddiqui [8] provides qualities of service mechanism to infrastructural solution that are implemented through Soft Phones in the interconnected campus through quality of service mechanism. B. Kim et al. [9] give the concepts, how packet loss and delay would be minimized in the internet network.

### III. PROBLEM DEFINITION

As envisage from above section, in order to ensure timely delivery of data over wireless network not a lot of work has been done and only a rare work is done on voice and QoS issues in Time Sensitive Networks (TSN). Since TSN are venerable in the sense that they are highly in demand for precise and time critical applications. Since many networks exist for real time applications but only those applications, which have implications related to voice over the Venerable networks, are critical and can be solved by the time sensitive network based on its advances. In the predominantly stage the time sensitive network works from the identification of traffic characteristics so that time synchronization of devices routers and switches on the network. Large network entail of small test it will leads to the dropped packets in voice i.e. measured by the bandwidth, which is reserved for the time sensitive queues.

Keeping all these factors in mind, this paper will try to calibrate these issues and measure QoS problems for them so that an inclusive plan to mitigate them can be designed.

### IV. ISSUES FOR TIME SENSITIVE APPLICATIONS

Quality of Service (QoS) is a major issue for time sensitive applications. The QoS issues may arise due to various factors like frame relay, internet protocol etc. Also many wireless networks lack integrated approach to deliver the data with standard quality are other relevant issues. So to get the standard quality of the data the improvement in the QoS is needed. It would be the major issue for this paper also. Firstly we will understand how the voice signals deteriorate when the voice is passes through the signal. It is a time sensitive issue that has to be considered in the wireless network. As per network application delay in voice data is very sensitive. When voice is delayed due to any reason it will degrade the quality of a network. Quality of service routing would be also considered as issue for a time sensitive application. When the data is routed sometimes an echo occurs which cause reflection of own voice and calls coming from the other places. It is a genuine problem; it leads to the disturbance for the users. Authentication is required to get the original data (voice), so security would be an issue for time sensitive applications. A critical issue is real time communication where provide service guarantees in the network. Loss of packets over the wireless network would be an issue for the time sensitive applications.

## V. IMPLICATIONS OF VOICE IN DATA NETWORKS

Over the decade huge amount of data is communicated over the internet. This data can be audio, video, voice, pictures, messages etc. But our main aim of this paper is to describe the implications of voice in data networks like frame relay, internet protocol, ATM, public switched telephone network, Ethernet etc. Now question arises, why voice is more important because voice provides effective communication medium for everyone so anyone can understand and take actions accordingly. If voice suffered with implications, so it becomes a genuine problem in time sensitive environment. This has created a motivation to design a “Time Sensitive network” to solve implications of voice.

Frame relay, Internet protocol and ATM are known cell switching technologies. Bandwidth is allocated dynamically when network needs to transmission. Public Switched Telephone Networks used dedicated path, hence inefficient use of bandwidth. Frame Relay and Internet Protocol delayed the voice transmission due to network congestion. So, dropped packets are increased and may leads to deteriorate the integrity of voice transmission. In frame relay to overcome the problem of dropped packets priorities are set for traffic. When priorities are set data packets are divided into the small fragments, so it increase the overhead and bandwidth efficiency reduced. When data is in queue it varies the arrival time of packets called jitter in Frame Relay Networks. Echo cancellation is also more important in voice over Internet Protocol, which suffers delays in the network. ATM services are costly and yet not universally available. AAL1 (ATM Adaptation Layer 1) and VOATM (Voice over ATM) produce waste bandwidth and voice transmission overhead increased. Frame Relay Access (FRA) not giving good voice quality because usually it uses data rates 56/64 kbps and used algorithms ITUG.723.1 and G.729A. But using RAD algorithm it increases high quality of voice. RAD is an internetworking strategy between Frame Relay and Internet Protocol.

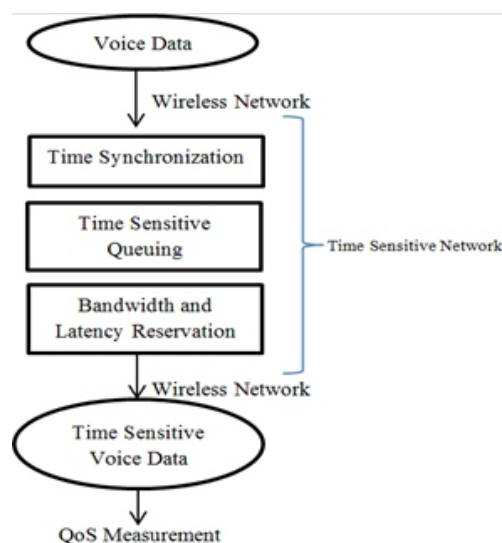


Figure 1: Block Diagram for Measuring QoS

Figure1 shows how QoS can be measured using voice data in the wireless network. Currently the problem would be solved using time sensitive synchronization, time sensitive queuing and bandwidth/latency reservation. So the time sensitive voice data would be come from the time sensitive network. Finally measure the Quality of services (QoS) of voice by the network.

## **VI. MEASUREMENT OF QoS OF VOICE BY THE NETWORK**

Thoroughly, we about the Quality of Service (QoS) and this section discuss how to measure the QoS by the network. Quality measurement of voice over internet uses PSTN, ISO/OSI model and mean opinion score. Functions and levels are used to systematically measurement of the Quality of service (QoS). To improve the quality of network it must include integrity, objectivity, and feasibility to design a measuring process. Integrity means data should be correct and accurate, objectivity means data should be in the either form and feasibility means analysis and evaluation of the proposed work.

To measure the QoS Catalina and Mithai have devised four parameters viz. bandwidth, latency, packet loss and jitter. Bandwidth means the amount of data carried out from one point to the other point in a given period of time, latency means delay how much time it take for a packet to get one designated point to another, packet loss means when data is travelled through the network but some of the packets would not reach to the destination and jitter means variations in the time interval between the arrival of package due to congestion and packet loss. Packet delay is minimized by suing the mechanism clock synchronization. Voice quality affected by packet loss rather mouth-to-ear delay over the wireless access network.

## **VII. CONCLUSION AND FUTURE WORK**

We conclude that this paper gives list of Implications of voice in Venerable Network that is used in time critical applications. Networks are prevailing for time critical applications, but some issues are over there, these issues measure the QoS of voice by the network. Future work will be elucidation of real time applications and how these problems can be solved, where time sensitive network can be used. Also simulation study of data transmission over TSN and identification of proper simulation plate form will provide a practical support to research in the field.



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# Study Of Power Electronic Converter And Modelling Of Pid Controller: An Analytical Study

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## **ABSTRACT**

*The utilization of DC-DC control converters is ceaselessly becoming both in control electronics items and frameworks. In a DC-DC converter application, it is constantly craved to get a directed yield voltage in spite of changes in input voltage, stack current and converter parts. To get controlled yield voltage analysts have utilized different straightforward ordinary to complex automatic control techniques. With the advancement of semiconductor manufacturing technology, effortlessness of configuration, size of gadgets, cost and better change efficiency have turned out to be critical outline criteria. This paper proposes the outline of a straightforward PID controller that can be connected to any DC-DC converter topology. The outlined PID controller is tried with buck and lift converter in MATLAB-Simulink environment. Recreation comes about demonstrate that the controller understands a superior yield voltage following and enhanced converter efficiency alongside the straightforwardness and effortlessness in outline.*

## **1. INTRODUCTION**

Power electronics manages an assortment of converters that are utilized at power level as opposed to the flag level. A power electronic framework comprises of at least one power electronic converters. A power electronic converter is comprised of some power semiconductor gadgets controlled by coordinated circuit. The exchanging attributes of power semiconductor gadgets allows a power electronic converter to shape the information power of one frame to yield power of some other shape.

DC-DC converters are a portion of the most straightforward power electronic converter circuits. They are generally utilized as a part of the power supply hardware for most electronic instruments and furthermore in particular high power applications, for example, battery charging, plating and welding. The wide assortment of circuit topologies ranges from the single transistor buck, lift and buck/support converters to complex arrangements containing two or four gadgets and utilizing a few strategies to control the exchanging misfortunes. The standard necessity of a control framework for the converter is



to keep up the yield voltage consistent independent of varieties in the DC source voltage  $V_{in}$  and the heap current. In any case, stack changes influence the yield voltage transitorily, conceivably causing huge deviations from the enduring state level. Furthermore, in a useful framework circuit misfortunes present a yield voltage reliance on relentless state stack current which must be made up for by the control system.

As of late, with the thriving of compact gadgets and advancement of semiconductor manufacturing technology, change efficiency, power utilization and size of gadgets have turned into the most vital outline criteria of exchanging power converters. It is basic to create exact exchanging power converters, which can lessen more squandered power vitality. For little applications it is imperative to direct the yield voltage of the converter with high accuracy and execution. Subsequently, a tradeoff among cost, efficiency and yield drifters ought to be considered.

This paper proposes the plan of a straightforward PID controller for power electronics DC-DC converter topologies. Cost, measure, exchanging pace, efficiency and effortlessness are the vital purposes of worry for the plan of proposed PID controller.

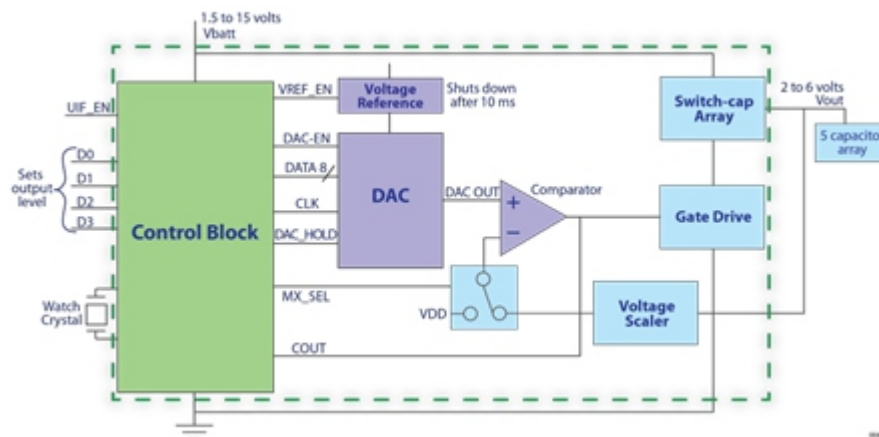
## 2. DC-DC CONVERTER CIRCUITS

The DC-DC converter has a few capacities. These are:

1. Convert a DC input voltage  $V_s$  into a DC yield voltage  $V_o$ .
2. Regulate the DC yield voltage against load and line variations.
3. Reduce the AC voltage swell on the DC yield voltage beneath the required level.
4. Provide isolation between the info source and the heap (if required).
5. Protect the provided framework and the info source from electromagnetic impedance

Figure 1 demonstrates a dc-dc converter as a black box. It changes over a dc input voltage,  $v_g(t)$ , to a dc yield voltage,  $v_o(t)$ , with an extent other than the info voltage. This transformation can be accomplished by an assortment of circuits in light of and utilizing exchanging gadgets. The generally utilized exchanging gadgets are diodes, thyristors, power MOS, etc. The converter regularly incorporates one (or a few) transistor(s) keeping in mind the end goal to control the yield voltage, utilizing the control flag  $(t)$ . It is attractive that the change be made with low misfortunes in the converter. To get low misfortunes, resistors are maintained a strategic distance from in the converters. Capacitors and inductors are utilized rather since in a perfect world they have no misfortunes. The electrical segments can be joined and

associated with each other in various ways, called topologies, every one having diverse properties. The buck, lift, and buck-boost converters are three fundamental converter topologies.



**Fig1.** Block diagram of a DC-DC converter

By utilizing pulse width modulation (PWM) control, direction of output voltage is accomplished by fluctuating the obligation cycle of the switch. Obligation cycle alludes to proportion of the period where power semiconductor is continued to the cycle time frame. Pulse width modulation (PWM) is a powerful procedure for controlling analog circuits. PWM is utilized in a wide assortment of applications, going from estimation and interchanges to power control and transformation. Control of PWM is normally affected by an IC, fundamental for regulating the output. The transistor switch is the most vital thing of the exchanged supply and controls the power provided to the load. It is likewise expressed that Power MOSFET's are more reasonable than BJT at power output of the request of 50 W. Picking of transistor additionally should consider its quick exchanging circumstances and ready to withstand the voltage spikes delivered by the inductor.

The proposed converter utilizes IGBT as the exchanging gadget. Utilization of IGBTs permits to construct less expensive and better converters. They have three appealing points of interest: higher exchanging frequency, simple and straightforward entryway control and no requirement for snubber circuits. IGBTs are constantly controllable amid turn on and kill. This makes overcurrent limitation significantly less demanding and permits  $dV/dt$  control to decrease and  $dV/dt$  stresses.

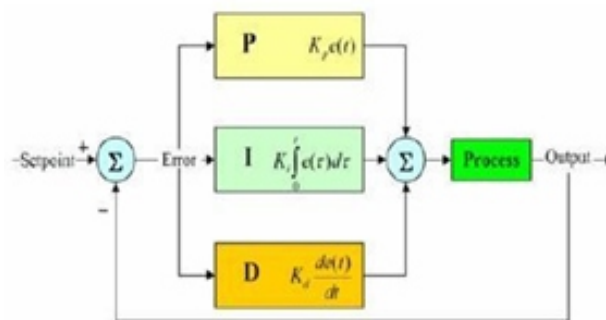
Numerous viewpoints must be considered for the situation where a converter is to be planned. One such angle is keeping the output voltage in the specified voltage interim. Here are a few cases of changes that can diminish the variety of the output voltage:

- Change the properties of a portion of the parts in the converter, e.g. increment the capacitance of the capacitor.
- Change the converter topology.
- Change to a more propelled controller.
- Increase the quantity of signs that are measured and utilized by the controller.
- Higher cost.
- Increased weight and volume.
- Lower unwavering quality.
- Lower efficiency.

Accordingly, the change or changes that are most appropriate depend to a huge degree on the converter particular close by. Converters can be enhanced as better segments are produced and more learning winds up noticeably accessible. This rouses inquire about in the zones of segments, converter topologies and controllers for instance. To acquire elite control of a framework, a great model of the framework is required.

### 3. PID CONTROLLER

Relative Integral-Derivative (PID) controller has been utilized for a very long while in enterprises for process control applications. PID includes three separate parameter, the relative, the fundamental and subsidiaries. By tuning the three constants in PID controller calculation, the controller can give control activity intended to particular process necessity.



*Fig2. Block diagram of a PID Controller*

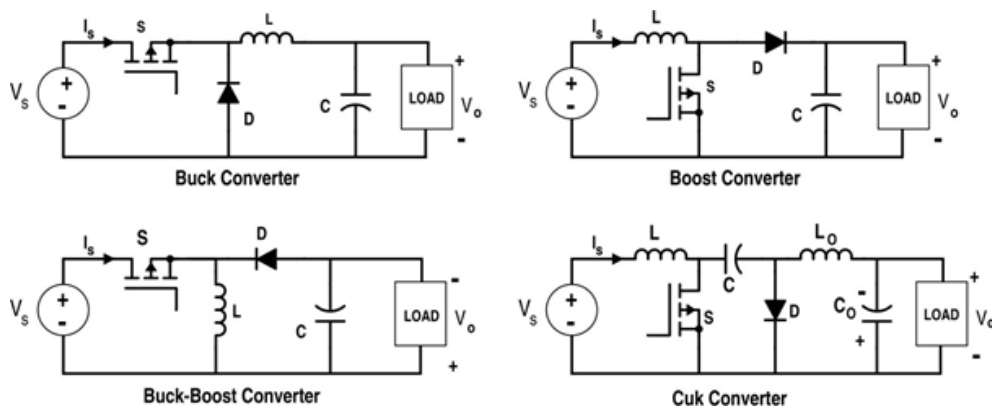
There are different strategies accessible for the tuning of PID controller. In any case, for comfort reason experimentation technique is for the most part utilized.

To begin with the PID controller works in a shut circle framework appeared in Figure2. The variable (e)

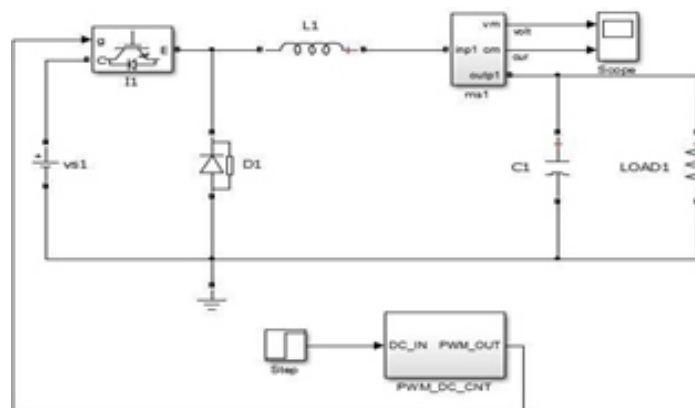
speaks to the following mistake, the distinction between the coveted info esteem (R) and the genuine yield. This blunder flag (e) will be sent to the PID controller, and the controller figures both the derivative and the necessary of this mistake flag. The flag (u) simply past the controller is currently equivalent to the proportional pick up ( $K_p$ ) times the size of the blunder in addition to the indispensable pick up ( $K_i$ ) times the necessary of the mistake in addition to the derivative pick up ( $K_d$ ) times the derivative of the blunder where, this flag (u) will be sent to the plant, and the new yield will be gotten. This new yield will be sent back to the sensor again to locate the new mistake flag (e). The controller takes this new mistake flag and registers its derivative and it's essential once more. This procedure continues forever, this flag (u) is gotten as takes after.

#### 4. DC-DC BUCK CONVERTER CIRCUIT

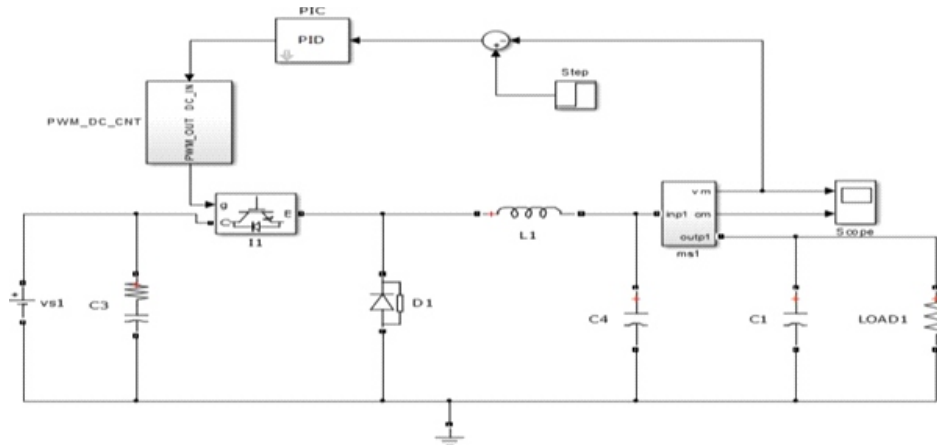
The operation of basic buck converter for mathematical modeling and analysis is represented in figure below,



**Fig3.** Basic Buck Converter circuit (open loop)



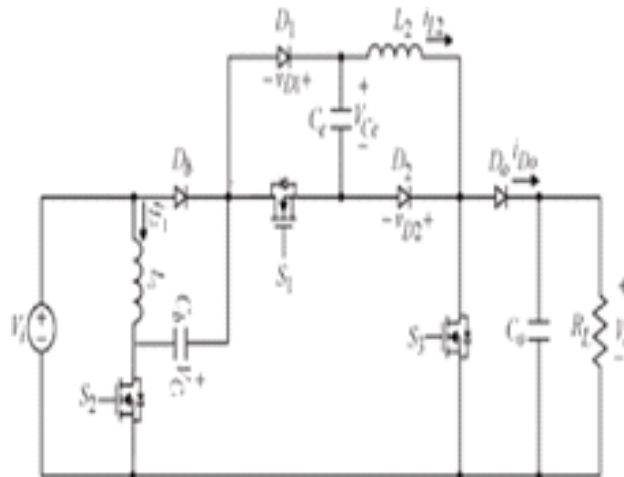
**Fig 4.** Buck Converter circuit with PWM



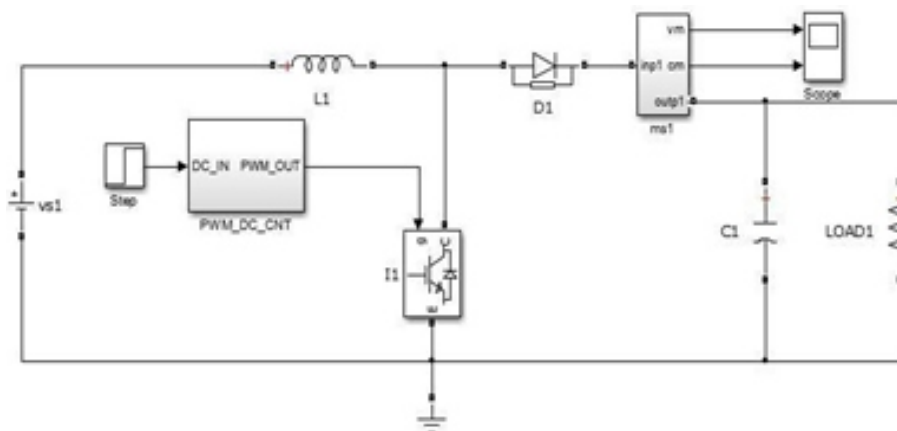
**Fig 5.** Buck Converter circuit with PID controller(closed loop)

## 5. DC-DC BOOST CONVERTER CIRCUIT

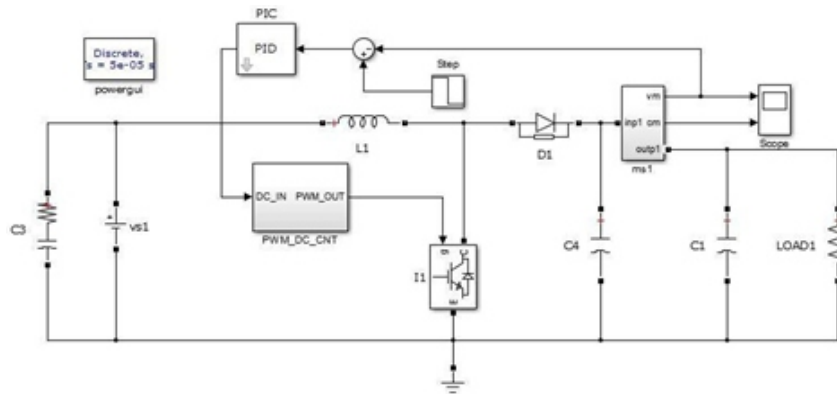
The operation of basic boost converter for mathematical modeling and analysis is represented in the figures 6,7 and 8 below.



**Fig 6.** Basic Boostloopconverter circuit (open loop)



**Fig7.** Boost converter circuit with PWM

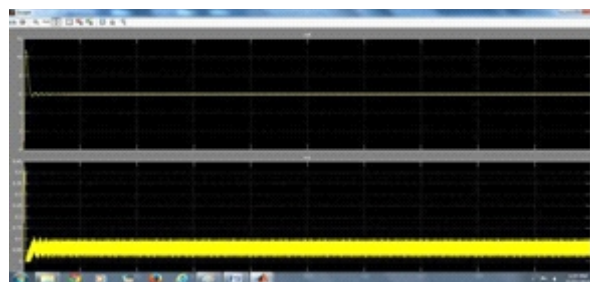


**Fig 8.** Boost converter with PID controller (closed loop)

Two topologies of DC-DC converter to be specific ,buck and lift converters have been outlined and examined. Figures 3, 4, 5 indicate buck converter in open circle and shut circle. The PID controller controls the obligation cycle of PWM flag connected to semiconductor switch IGBT according to the yield necessity. For buck converter the yield DC voltage is not as much as the info DC voltage. A similar PID controller is connected to support converter appeared in figures 6,7,8. The outline and reproduction is completed in MATLAB-Simulink environment.

## 6. EXPERIMENTAL ANALYSIS RESULTS

The proposed PID controller for buck and lift converter circuit is outlined and recreated utilizing MATLAB Simulink environment. The PID parameters are resolved in light of the basic control designing information that transient exhibitions can be enhanced if the P and I picks up are huge and the D gain is small at the beginning. For both the topologies input voltage is taken as 12V. The operation of the converters can be best checked with step input reference voltage. Reenactment has been done for different changes in stack esteems and info variations. Results demonstrate that the composed PID controller has better yield voltage following capacity, accordingly enhancing yield voltage direction. The plan is exceptionally straightforward with little size and decreased cost



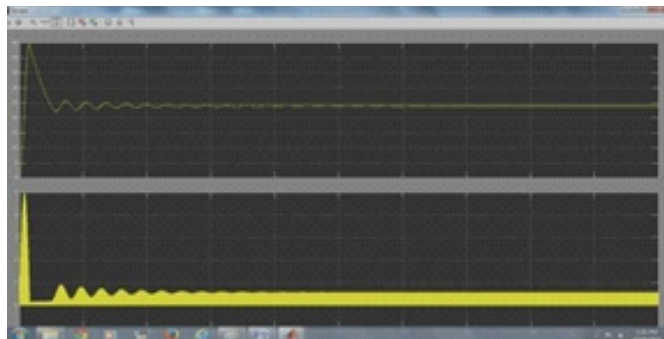
**Fig 9.** Output voltage waveform of basic buck converter



**Fig 10.** Output voltage waveform of buck converter with PWM



**Fig 11.** Output voltage waveform of buck converter with PID controller

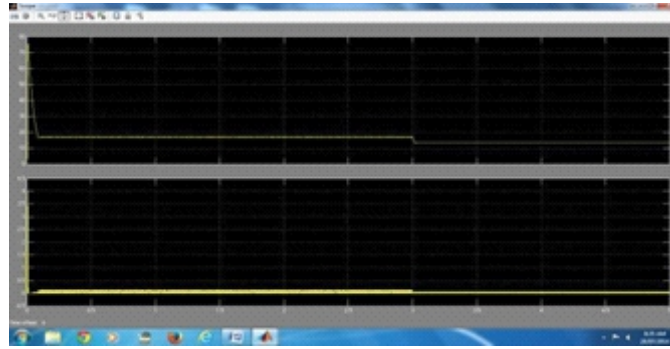


**Fig12.** Output voltage waveform of basic boost converter



**Fig13.** Output voltage waveform of boost converter with PWM





**Fig14.** Output voltage waveform of boost converter with PID controller

## 7. CONCLUSION

The composed buck and lift converter works successfully when PID controller is utilized. The controller understands a superior yield voltage following with insignificant overshoot, little relentless state errors, short settling time and enhanced converter efficiency. The outline is straightforward and simple with lessened size and cost.

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