A Hybrid Scheme with Impressive QoS over Telecommunication Networks

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***Abstract*— This paper introduced and analyzed a Hybrid scheme which is designed to improve the performance of back bone telecommunication networks. It is compared to the regular traffic shaping and policing schemes which are designed to maintain the quality of service (QoS) based on ITU-I.350 standard. The varieties of information such as voice, video and data are served to explore and evaluate the system for classical and hybrid schemes. The Asynchronous Transfer Mode network (ATM) was selected to evaluate and analyze because ATM's properties are provided on high reliability and high speed data rate, also ATM network was deployed as back bone telecommunication network in many countries. The results are represented in the form of mathematics and supported by simulations.**

Index Terms— ***QoS, Traffic shaping, Traffic policing, Hybrid scheme.***

I. Introduction

Quality of Service (QoS) in the field of computer networking and other packet-switched telecommunication networks was described in [1]. In point of view, the traffic engineering term for QoS was referred as a resource reservation control mechanisms rather than the achieved service quality. QoS is the ability to provide different priority to different applications, users, or data flows, or to guarantee a certain level of performance to a data flow. For example, a required bit rate [2], delay, jitter [3], packet dropping probability [4] and/or bit error rate [5] may be guaranteed. QoS guarantees are important if the network capacity is insufficient, especially for real-time streaming multimedia applications such as voice over IP [6], online games and IP-TV [7], since these often require fixed bit rate and are delay sensitive, and in networks where the capacity is a limited resource, for example in cellular data communications. In the absence of network congestion, QoS mechanisms are not required.

Manuscript received May 15, 2011. This review article was supported by Department of Telecommunication Science, Faculty of Science & Technology Assumption University, Thailand.

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Many researchers perform research to solve the congestion problem which is more critical in the telecommunications network since it highly serves a huge amount of information nowadays and much more in the future.

In the field of telephony, quality of service was defined in the ITU standard X.902 as "A set of quality requirements on the collective behavior of one or more objects". Quality of Service comprises requirements on all the aspects of a connection, such as service response time [8], loss, signal-to-noise ratio [9], cross-talk [10], echo [11], interrupts [12], frequency response, loudness levels, and so on. A subset of telephony QoS is the Grade of Service (GOS) requirements, which comprises aspects of a connection related to capacity and coverage of a network, for example the guaranteed maximum blocking probability and the outage probability [13].

Section II explained why Asynchronous Transfer Mode (ATM) network is selected to analyze and evaluate as our concerned network. Section III described the differences of methods to improve the QoS in ATM network. Section IV described the regular traffic shaping based on queuing network model with mathematic formulas. Section V described the hybrid scheme which is performed in queuing network model to obtain better performance than regular traffic shaping based on its algorithm, and mathematic formula. Section VI gave the definition of comparison for both mechanisms. Section VII showed the results and discussion. Last is conclusion.

II. Asynchronous Transfer Mode

ATM network provides a unique function when compared to another network technology with the following characteristics:

- It is scalable and flexible. It can support megabit-to-gigabit transfer speeds and is not tied to a specific physical medium.
- It efficiently transmits video, audio, and data through the implementation of several adaptation layers.
- Bandwidth can be allocated as needed, lessening the impact on and by high-bandwidth users.
- It transmits data in fixed-length packets, called cells, each of which is 53 bytes long, containing 48 bytes of payload and 5 bytes of header.

- It is asynchronous in the sense that although cells are relayed synchronously, particular users need not send data at regular intervals.
- It is connection oriented, using a virtual circuit to transmit cells that share the same source and destination over the same route.

Moreover, ATM network provides a number of native features to support QoS:

- Fixed-size cells (as opposed to IP's variable-length packets) provide predictable throughput. As an analogy, if all boxcars on a train are the same size, you can predict how many will pass a certain point if you know the speed of the train.
- Predictable behavior allows for bandwidth management and the creation of guaranteed service-level agreements.
- ATM is also connection oriented and delivers data over virtual circuits that deliver cells in order, an important requirement for real-time audio and video.
- ATM supports admittance control and policing, which monitor traffic and only allow a new flow if the network will support it without affecting the bandwidth requirements of other users.
- ATM networks "police" traffic to prevent senders from exceeding their bandwidth allocations. If traffic exceeds a certain level, the network may drop packets in that circuit. Packets are classified, with some being more "drop eligible" than other.

Based on the above characteristics and feature’s ability, ATM will be considered as target network to analyze and evaluation.

III. Method to Maintain QoS

Figure 1. illustrated the key difference between traffic policing and traffic shaping. When the traffic rate reaches the configured maximum rate, excess traffic is dropped (or remarked) which is traffic policing mechanism. The result is an output rate that appears as a saw-tooth with crests and troughs. In contrast to policing, traffic shaping retains excess packets in a queue and then schedules the excess for later transmission over increments of time. The result of traffic shaping is a smoothed packet output rate.

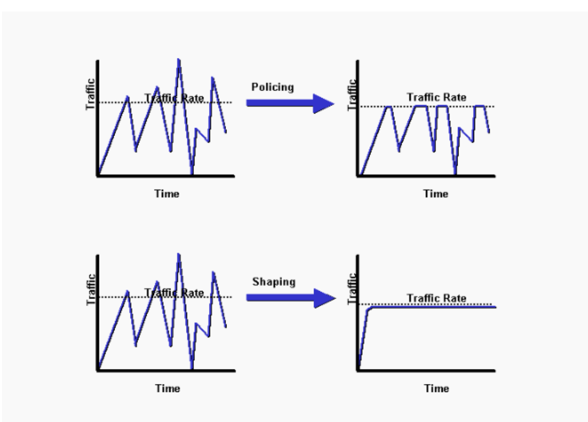

Fig. 1. Policing Versus Shaping [14]

The shaping implies the existence of a queue and of sufficient memory to buffer delayed packets, while policing does not. Queuing is an outbound concept; packets going out of an interface get queued and can be shaped. Only policing can be applied to inbound traffic on an interface. Memory size must be sufficient when enabling shaping. In addition, shaping requires a scheduling function for later transmission of any delayed packets. This scheduling function allows traffic engineers to organize the shaping queue into different queues. Examples of scheduling functions are Class Based Weighted Fair Queuing (CBWFQ) [15] and Low Latency Queuing (LLQ) [16].

However, traffic policing is the process of monitoring network traffic for compliance with a traffic contract and taking steps to enforce that contract. Traffic sources which are aware of a traffic contract may apply traffic shaping to ensure that their output stays within the contract and is thus not discarded. Traffic exceeding a traffic contract may be discarded immediately, marked as non-compliant, or left as-is, depending on the administrative policy and the characteristics of the excess traffic.

IV. Regular Traffic Shaping

Reference[17] introduced the basic traffic shaping which deploy over ATM network. This back bone network provided a high speed of transfer information rate which varies from 48 Mb/s till Tb/s. Since this network is served with multimedia traffic then collision or packet loss will exist as described in [18]. Then [19] simply used a mechanism to solve this problem as concluded in Figure 2., and Figure 3. represents this problem in the queuing model.

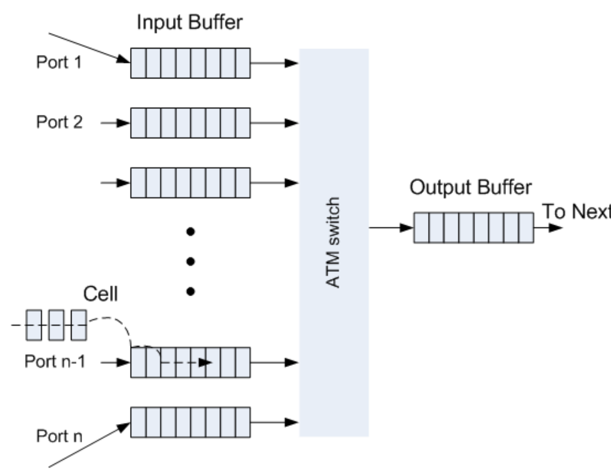

Fig. 2. Analytical model

Then the basic formula which is shown below was applied in [19]. The math formula was described as Derivation of Steady State Probability.

In [19], ATM switches were referred to the 3COM CELLplex 7000 switch specification for this investigation. Each switch will require a processing time of 10 microseconds. 10 microseconds was applied as exponential service time. For exponential distribution, probability density function (pdf) is

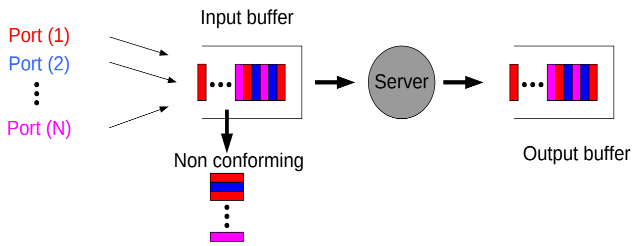

Fig. 3. Queuing model

given by

$$f(x) = \begin{cases} \frac{1}{\mu} e^{\frac{-x}{\mu}}, x \geq 0 \\ 0, \quad x < 0 \end{cases}$$

The cumulative distribution function (cdf) is given by

$$F(x) = \int_{-\alpha}^{x} f(t)dt = \begin{cases} 1 - e^{\frac{-x}{\mu}}, x \geq 0 \\ 0, \quad x < 0 \end{cases}$$

The parameter m can be interpreted as the mean number of occurrences per time unit, For example, if interarrival times X1, X2, X3 , ….. have an exponential distribution with rate μ then μ could be interpreted as the mean number of arrivals per time unit, or the arrival rate. Notice for any i

$$E(X_i) = \frac{1}{\mu} \qquad i = 1,2,3,\ldots$$

The inverse transform technique can be utilized, for any distribution, but is most useful, when the cdf, F(x), is simply inverse, (F^{-1}), are computed. Thus, for the exponential distribution, the cdf is

$$F(x) = 1 - e^{-\mu x}, x \geq 0$$

Set F(X) = R on the range of X. For the exponential distribution, it becomes $1 - e^{-\mu x} = R$ on the range $X \geq 0$. Thus,

$$1 - e^{-\mu x} = R$$

$$e^{-\mu x} = 1 - R$$

$$-\mu x = \ln(1 - R)$$

$$x = (-\frac{1}{\mu})\ln(1 - R) \qquad (1)$$

Generating uniform random numbers R1, R2, R3, we can compute the desired random variables by

$$X_i = F^{-1}(R_i)$$

For the exponential case, from equation (1), then

$$F^{-1}(R) = (-\frac{1}{\mu})\ln(1 - R)$$

so that

$$X_i = (-\frac{1}{\mu})\ln(1 - R) \qquad (2)$$

for i = 1,2,3,…. One simplication that is usually employed in equation (2) is to replace 1- Ri by Ri to yield

$$X_i = (-\frac{1}{\mu})\ln(R_i)$$

which is justified since both Ri and $1-R_i$ are uniformly distributed on (0, 1). Then we apply exponential service time equal to -1 divied by (μ * (ln (R_i))) where μ equal to 10 and R_i can be computed by random function.

Reference[19] used the standard specification of fiber optic link (node 2 and 4). For a 155.52 Mbps link, the cell slot rate is 366,792 cell/s and the service time per cell is 2.726 microsecond. Then the cell slot rate available for traffic can be computed as 26/27 * 366792 = 353208 cell per second. This will give a result of 1/353208 = 2.831 microsecond per cell. Thus we will consider a constant bit rate of 2.831 microsecond for any individual cells over the link.

Input source node will generate cell, with corresponding traffic type, non priority level, and arrival process. Thus, in simulation we assume Poisson distribution function as arrival process for each cell. These cells may be described by counting function N(t) defined for all $t \geq 0$. The counting function will represent the number of cells that occurred in[0,t]. For each interval [0,t], the value N(t) is an observation of a random variable where the only possible value that can be assume by N(t) are integers 0,1,2,….

So probability that N(t) is equal to n is given by

$$P[N(t) = n] = \frac{e^{-\lambda(t)} (\lambda t)^n}{n!} \quad \text{for } t \geq 0 \text{ and } n = 0,1,2,\ldots \qquad (3)$$

The Poisson probability mass function is given by

$$P(x) = \begin{cases} \frac{e^{-\lambda}\lambda^x}{x!}, & x = 0,1,2,3\ldots\ldots \\ 0, & otherwise \end{cases} \qquad (4)$$

where $\lambda > 0$.

Comparing equation (3) to equation (4), it can be seen that N(t) has the Poisson distribution with parameter $\alpha = \lambda t$. Thus, its mean and variance are given by

$$E[N(t)] = \alpha = \lambda t = V[N(t)]$$

For arbitrary s and t satisfaction s < t, the assumption of stationary increments implies that the random variable N(t) – N(s), representing the number of arrivals in the interval [s, t] is also Poisson distributionwith mean λ (t-s). Thus,

$$P[N(t) - N(s) = n] = \frac{e^{-\lambda(t-s)} [\lambda(t-s)]^n}{n!} \quad \text{for } n = 0,1,2,\ldots$$

and

$E[N(t) - N(s)] = \lambda (t - s) = V[N(t) - N(s)]$

Now, consider the time at which cell arrivals occur in a Poisson process. Let the first arrival occur at time A1, The second occur at time A1+A2, and so on. Thus, A1,A2, are successive interarrival times. Since the first arrival occurs after t if any only if there are no arrivals in the interval[0,t], it is seen that

$$\{ A_1 > t \} = \{ N(t) = 0 \}$$

and, therefore,

$$P(A_1 > t) = P [N (t) =0] = e = e^{-\lambda t}$$

which is the cdf for an exponential distribution with parameter λ. Hence, A_1 is distributed exponential with mean $E(A_1)$equal to $1/\lambda$ Thus, arrival time of each cell equal to $- 1/\lambda . \ln (R_i)$.

Result from [19] based on math analysis was shown that the waiting time in queue is high when compared with the QoS maximum delay time which is issued in ITU-I.356.

Then [20] introduced the Hybrid scheme to overcome this situation.

V. HYBRID SCHEME

The draw back of regular system is the drop cells which discarded from the system by collision, buffer overflow, and issued low throughput. Then [20] proposed the hybrid scheme as presented in Figure 4.

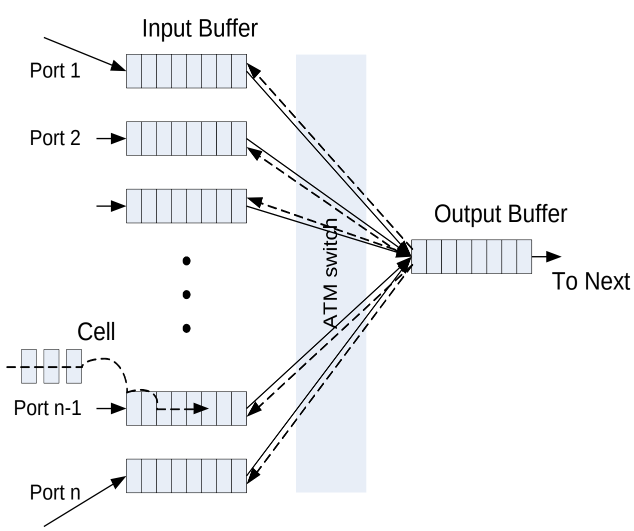

Fig. 4. Hybrid scheme.

The Hybrid scheme will utilize the bandwidth by adjusting the window size of the buffer based on the statistical scheme. The available slot that is empty will be served to the other which is heavy load due to input traffic. However this scheme must meet the QoS standard which is described in RFC 1024, traffic policing will deploy over this scheme as show in Figure 5.

Reference[20] proposed the Hybrid scheme in the name of Adaptive Rate Control (ARC) algorithm as described in next paragraph.

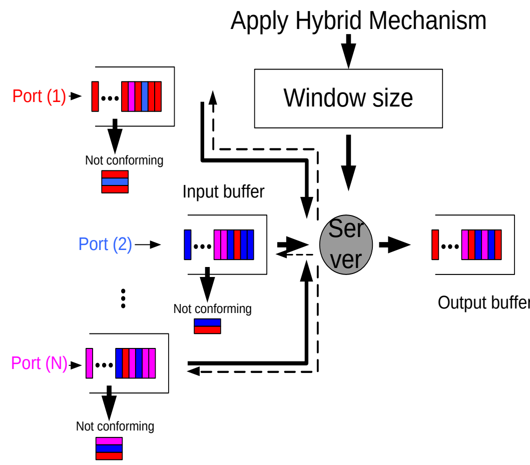

Fig. 5. Hybrid scheme in Queuing Model.

/*--------------------- Hybrid Algorithm -------------*/
Start:
Find number of port (P_all)
Find current allocate rate for each port(Ai)
P(id)(Ai)
For(i=0 and I <P_all so i=i+1)
{
If traffic type of P(id)[i] == voice
P(id)_type[i] = voice
Else if traffic type of P(id)[i] == video
P(id)_type[i] = video
Else P(id)_type[i] = data
}// Classified traffic characteristic for each port as Voice, Video, or Data //
Find current window size (Wi)
Priorityport = Sort (P(id)_type) base on Voice, Video,and Data
Do {
IF packet drop **THEN** {
for(i=0 and i < P_all so i = i + 1) {
Find New allocate rate for Priorityport[all-i]
current allocate rate P(id)(Ai) = New allocate rate P(id)(Ai) }
Find New windows size.
current window size = New window size }
ELSE {
for(i=0 and i < P_all so i = i + 1){
current allocate rate P(id)(Ai)
=current allocate rate Priorityport[i] }
current window size = current window size }
}**While** (Cell is transmitting)
END:
where
Find current allocate rate (Ai)

$$Source\_rate(t) = \frac{WindowSize\_(t) * Cell\_size}{\left(\sqrt{Cell\_drop(t-1) * Cell\_delay\_time}\right)} \quad [20]$$

Cell_delay_time = Cell traveling time between source to destination

Find current window size (Wi)

$$WindowSize(t) = \frac{Bandwidth(t-1) * Cell\_delay\_time}{P\_all} \quad [20]$$

This algorithm will help the drop cells to retransmit based on priority to maintain QoS.

VI. Method of Investigation

In this section two schemes will be compared by loaded multimedia traffic to ATM network based on three scenarios. First is medium loaded capacity of ATM switch, second is nearly fully loaded capacity of ATM switch and last is exceeded loaded capacity of ATM switch. This paper will present the extreme case which is the last scenario for our consideration.

The Peak Rate (*PR*) equal to λa and that equal to *1/T* which is in the units of cell/second, where T is the minimum inter-cell spacing in seconds (i.e., the time interval from the first bit of one cell to the first bit of the next cell)

$PR = \lambda a = 200$ Mbps (471698 cells/s)

Means input traffic to ATM switch will receive cells equal to 471698 cells in 1 second. Hence,

Voice = 33% of total voice cells (157233 cells/s),
Video = 33% of total video cells (157233 cells/s),
Data = 33% of total data cells (157232 cells/s).

The 3COM CELL plex 7000 was equipped with 24 ports, then 24 is deployed as P_all parameter in hybrid scheme.

VII. Result and Discussion

The comparison results for both mechanisms are shown in four graphs as throughput, number of dropped cells, mean queue delay time, and utilization of the bandwidth.

From Figure 6, Hybrid scheme allows more voice traffic than regular shaping scheme and also video traffic but in vice versa hybrid scheme performs worse for data traffic. This circumstance is based on priority which is deployed over hybrid scheme as voice video and data. However, the overall throughput which hybrid scheme performs is 365900 cells/second, better than regular shaping scheme of 346700 cells/second. 19200 cells are the number of cells that hybrid scheme performed better.

With reasons above, number of dropped cells which hybrid scheme performs will be less than regular traffic shaping scheme as shown in Figure 7. However, Hybrid scheme performs longer delay for data traffic by allocated data traffic's cell and gave it to voice traffic and video traffic as shown in Figure 8. This means that data traffic's cells have to wait in the queue buffer and cause each cell not to be processed immediately due to that hybrid scheme applied prioritization.

Figure 9 represented the work load on the ATM switch in the percentage units. Hybrid scheme will force switch to be busy all of the time unlike regular shaping scheme which was given some idle time.

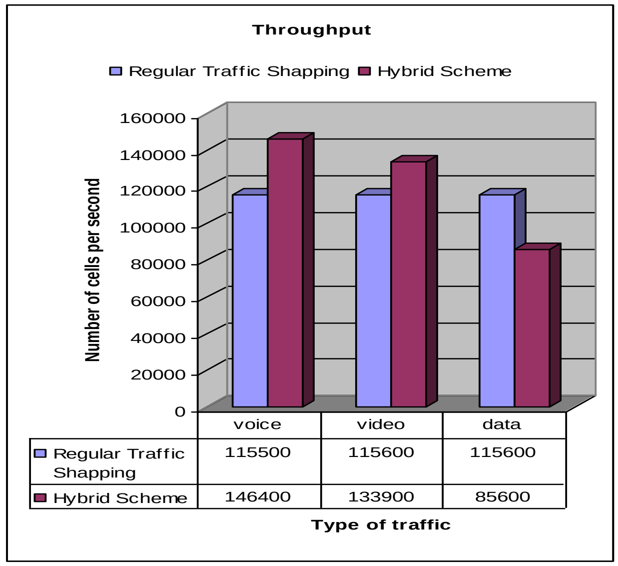

Fig. 6. Throughput

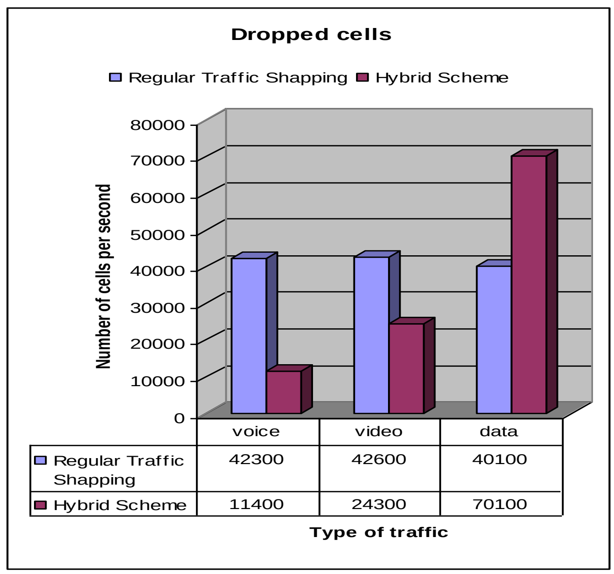

Fig. 7. Number of dropped cells

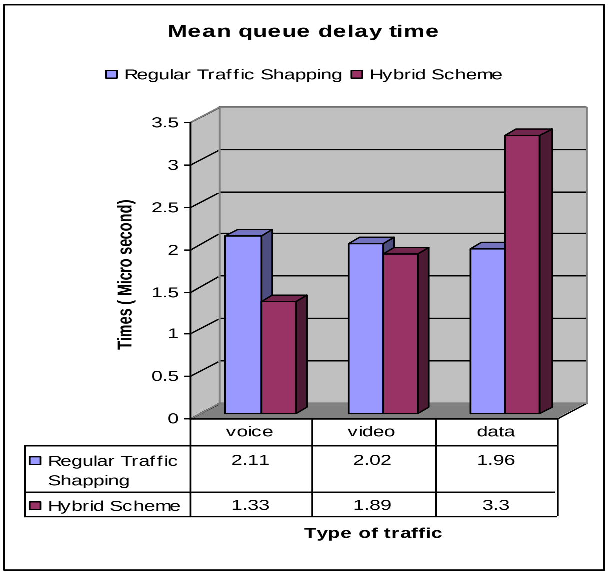

Fig. 8. Mean queue delay time

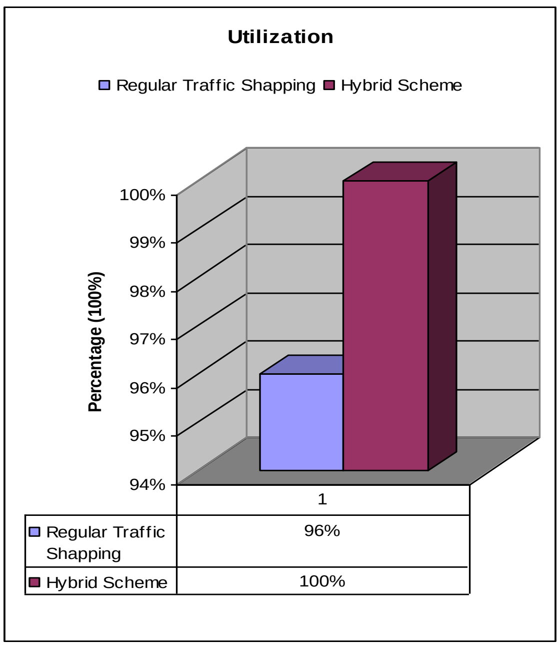

Fig. 9. Utilization of the ATM switch

VIII. CONCLUSION

Hybrid scheme allows more on voice traffic and video traffic to transmit through ATM switch. It controls the QoS base upon ITU-I.356 standard. Figure 6 shows the number of cells for each scheme performed, but data traffic type was dropped due to that their occupied cells were assigned to voice and video traffic in order to maintain the QoS of the multimedia traffic with priority base in sequence. However if we observe the mean queue delay time as shown in Figure 8, the voice traffic and video traffic spend shorter time in the system than regular shaping scheme because they are not waiting for their own slots. Then as priority setting, data traffic spends longer time in the system. Besides, this hybrid scheme performs less dropped cells on voice traffic and video traffic. Then the performance in multimedia traffic will be improved which is shown in Figure 6 and Figure 7.

Besides, if all telecommunications network deploys hybrid scheme, it will increase the throughput through the system when compared to the regular system. The reason that voice traffic and video traffic were increasing, is the priority which is assigned from the hybrid scheme. However the increasing delay time for data traffic cells does not exceed the required standard [21]. Also the beneficial would be more to service provider as well as more impression from the users.

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