Multimedia Traffic Over VP-Based ATM Ring Network

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Abstract- Asynchronous Transfer Mode (ATM) installations will reach their peak in the next decade. It is time to begin to consider what technology should follow ATM and how best to get there. In this paper, we address some of the most important issue regarding recent advances in ATM research, including video, voice, and data integration. The performance of integration video, voice, and data on the proposed VP-Based ATM Ring network has been studied. A suggested control method provides fairness among the traffics, and confirms its effectiveness for integrating of multimedia traffic on the ATM network.

index terms: VP-Based Ring ATM network, integration multimedia traffic.

1 Introduction.

The Broadband integrated Services Digital Network (B-ISDN) is intended to meet the transport needs of the future. It will provide an integrated high speed service for all types of traffic- image, graphics, data, teephony, broadcast TV, and interactive video. All applications, and interactive multimedia in particular, will benefit from a single network that can provide the required quality of service (QoS). These networks will use digital switching and transmission, and ATM is the chosen technology [1,2]. ATM networks have the capability to provide a wide range of services and guarantee various end-to-end QoS [3]. The basic transmission unit for ATM is a cell with 53 bytes. Each cell with a fixed length simplifies the process of segmentation and reassembly. When the cell routes through a switch node, Virtual Path Identifier (VPI) and Virtual Channel Identifier (VCI) in the cell header are used for looking up switching tables to quickly decide the output VPI, VCI, and port number for completion of switching. Several Virtual Paths (VPs) and Virtual Channels (VCs) can be multiplexed to share a link bandwidth. By fast switching technology and flexible multiplexing, ATM has the advantages of both circuit switching and packet switching. One important characteristic of ATM network is that the ATM networks support various services types to meet application with different QoS [4,5,6]. This paper investigates the performance of the ATM ring network for carrying multimedia (video, voice, and data) traffic.

2 The Proposed Network

The proposed network is a VP-Based ATM Ring Network based on a ring topology modified to the architecture in [7]. In our study the VP in the ring is defined as pair for each node, which preassigned a duplex VP. Figure 1 illustrates (Synchronous Optical NETwork) SONET/ATM Ring Architecture using point-to-point VP's (denoted by SARPVP). The VP is used in the point-to-point VP Add-Drop multiplexing (VP-ADM) scheme carries VC between the same two ring nodes. As mentioned in the VP-Based architecture, each ring's node pair is preassigned a duplex VP. For example, from Figure 1, VP#2 and VP#2` (not shown in the Figure) carry all VCs from Node 1 to 3 and from Node 3 to Node 1 respectively. The physical route assignment for the VP depends on the type (unidirectional/bi-directional) of the considered SONET ring.
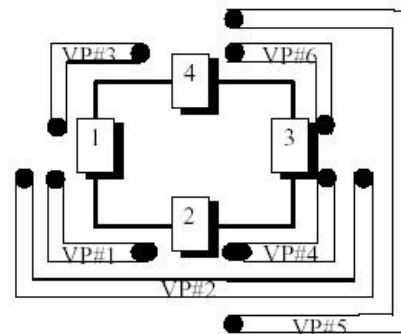

Figure 1 VP-Based ATM Ring network.

In a unidirectional ring, two diverse routes which form a circle are assigned to each VP as shown in Figure 1. Two physical routes 1-2-3 and 3-4-1 are assigned to VP#2 and VP#2' (not shown in the Figure). In a bi-directional, only one route is assigned to each duplex VP (e.g., route 1-2-3 is assigned to both the VP#2 and VP#2`), and demands between Node 1 and 3 are routed through Route 1-2-3 bidirectionally for more details on SONET unidirectional and bi-directional ring architecture see [8,9]. In order to avoid the VP translation at intermediate ring nodes of VP connection, the VPI value is assigned on a global basis. The ATM cell add-drop or pass-through at each ring node is performed by checking the cell's VPI value. Since the VPI value has global significance and only one route is available for all outgoing cells. It need not be translated at each intermediate ring node. Thus no VP cross-connect capability is needed for the ATM ADM of this SARPVP ring architecture. The global VPI value assignment presents no problem here, since only one route exists for all outgoing ATM cells. The number of nodes supported by a ring is usually limited to 91 nodes depending on VP field size in bits. If the point-to-point VP ring is used to support present traffic such as Digital Signal type 1 (DS1) services (via circuit emulation), each DS1 comprises a VC connection and is assigned a VPI/VCI based on its addressing information and the relative position of the DS1 within all the DS1's terminating at the same source and destination on the ring. For example, VPI=2 and VCI=3 represents DS1 that is the third DS1 of the DS1 group terminating at Node 1 and Node 3.

2.1 Routing in the VP-Based ATM Ring Network

As mentioned above, the physical route assignment for the VP depends upon the type (unidirectional or bi-directional) of the considered SONET ring. VPI on the node defines it either transit node, which just pass the cells to the next node, or terminated node, which just drop the coming cells, according to the value of VPI. Each node has VPI value for each node else, and no two VPIs have the same value in the same network. Figure 2 illustrates the two types (unidirectional and bi-directional) of SONET Ring. So, the solid line alone indicates the unidirectional type, meanwhile both solid and dotted lines indicate the bi-directional type. In the unidirectional SONET ring, for N nodes, the maximum number of physical hops is (N-1). For example, node 1 transmits cells to node 2 via the route 1-2 and node 2 transmits cells to node 1 via route 2-3-...-N-1. Obviously that the route in the reverse direction is longer than that in the forward direction, resulting in the required time between the pair conversation is not equal. In the Bi-directional SONET ring, for N nodes, the maximum number of physical

hops is |N/2|. So for forward direction the path is represented as (1-2-3-...-N), and for reverse direction the path is represented as (N-(N-1)-...-3-2-1). If the value of N is odd, the number of VPIs for both forward and reverse directions are equal. Meanwhile, if the value of N is even, the number of VPIs in forward direction is more than the number of VPIs in reverse direction by one. Table 1 shows functions to determine the direction and the number of hops to send cell from source to destination. It is to be noted here that the queue for each direction makes the bi-directional SONET Ring work like two separately unidirectional SONET Ring. The following results in the unidirectional case. To be in bi-directional the number of sources is the duple of the number of sources in the case of unidirectional.
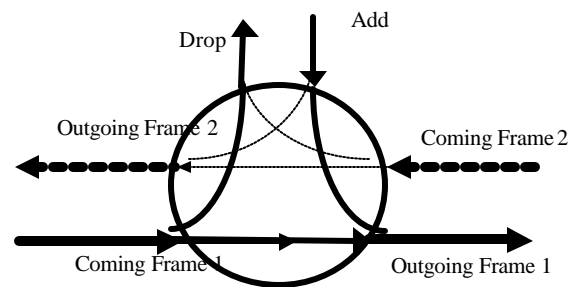

Figure 2   ATM/ADM Node

Table 1

Function 1: hop count
hop_count function(S: source, D: destination)
$min = MINMUM(S,D)$
$max = MAXIMUM(S,D)$
$if (|N/2| >= (max - min)$
$\qquad hop\_count = max - min$
$else$
$\qquad hop\_count = N - (max - min)$

Function 2: direction flag
dir_flag function (S: source , D: destination)
$min = MINMUM(S,D)$
$max = MAXIMUM(S,D)$
$hop = hop\_count(S,D)$
$if (hop = max - min)$
$\quad if (S < D)\ dir\_flag = forward$
$\quad else\ dir\_flag = reverse$
$\quad if (hop = N - (max - min))$
$\qquad if (S<D)\ dir\_flag = reverse$
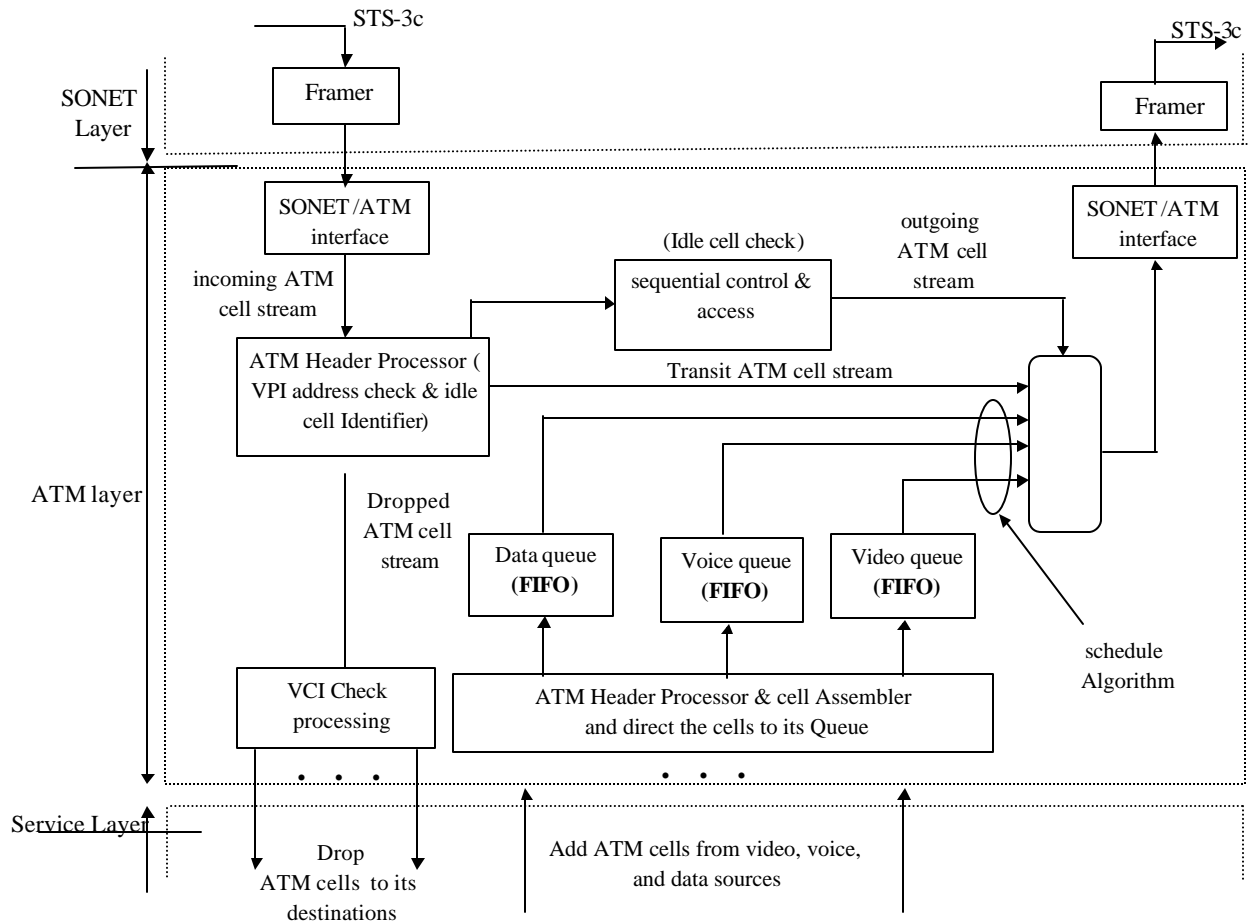$\qquad else\ dir\_flag = forward.$

Figure 3 An STS-3c Add-Drop Hardware Configuration

2.2 ATM Add-Drop Multiplexer (ADM) for VP-Based Ring

A possible Add-Drop Multiplexer (ADM) design for the VP-Based ATM Ring implementation is shown in Figure 3. The main goal of this ADM design is to minimize the cost of evolution from the previous SONET ADM's to ATM ADM's. The ATM ADM for the SARPVP architecture can be implemented in different ways depending on physical SONET STS-Nc terminations. The most commonly proposed ATM SNc termination are STS-3c, STS-12c, and STS-48c [7]. The ADM configuration with STS-3c terminations for a VP-Based ATM Ring implementation supports lower-level digital signals (DS1, DS2, AND ITU-T standards) services via circuit emulation [7,10]. The ATM VP add-drop function which is performed at the STS-3c level requires three major modules. The first module is the ATM/SONET interface,

which converts the STS-3c payload to an ATM cell stream and vice versa. The functions performed in this module include all delineation, self synchronization, and scrambling. The scramble process here is to increase the security and robustness of the cell delineation process against malicious users or unintended simulations of a cell header followed by a correct Header Error Control (HEC) in the information field. The second module is to perform header processing, which includes cell addressing (VPI in this case) and HEC. In order to perform cell add-drop/pass-through, this module checks VPI value of each cell to determine if it should be dropped or passed through. This module also identifies idle cells which can be used to insert cells from the considered office (i.e. signal adding) via a simple sequential access protocol. This sequential access protocol can be implemented by the third module that passes through each nonidle cell and inserts the added cells from each queues into outgoing idle cells in a sequential order. The

third functional module includes a service lower-level digital signal card based on VPI/VCI values of ATM cells. This service mapping module first demultiplexes all ATM cells from STS-3c payload and then drops ATM cells to its destinations according to the VCI values.

2.3 The Proposed Control Method to Allocate Cells

There are some drawbacks with the proposed control method in [11], in which when the offered load of the data traffic is higher than the video and voice traffics, unfairness among the traffics is involved, that is because the ratios among the traffics are fixed and the offered load of each traffic along with the time is not fixed. In our proposed control method [12], we tackle the unfairness problem between the whole traffic, that is by counting the number of cells from the queue which corresponding to each offered load by the following steps: 1) Assume that the ratios of video, voice, and data traffics are M: N: D respectively. These ratios are supposed to fill up a complete transmission frame. 2) For fairness services between the traffics we have to follow the following steps:

(a) Get Min_{ol} (Minimum offered load of all the applied traffics).

(b) Set $M = \dfrac{video\,offer\,ed\,load}{Min_{ol}}$. -----------------(1)

(c) Set $N = \dfrac{voice\,offer\,ed\,load}{Min_{ol}}$. ------------------(2)

(d) Set $D = \dfrac{data\,offer\,ed\,load}{Min_{ol}}$. ------------------(3)

The above mentioned proposed control mechanism tackled the problem of fairness between the traffics. For example, suppose that the offered load of each traffic are the same 0.333, 0.333, and 0.333 this means that the ratio will be 1:1:1, but if the offered load is different as (0.60, 0.30, 0.10), this means the ratio will be 6:3:1, and vice versa.

3 Traffic Model
3-1 Video traffic.

The video traffic is defined as the number of sources (N_{vi}) and video encoding rate (R_{vi}). We have considerd that video stream are encoded according to a coding standard such as H.261[13] or MPEG[14].

Figure 4 shows the block diagram of a station which encodes and sends a video stream over an ATM network. A frame is taken in the video camera, and sent as an analog signal to the frame grabber, where it is digitized, then it is compressed by the encoder. The data produced by the encoder is passed to the host computer.

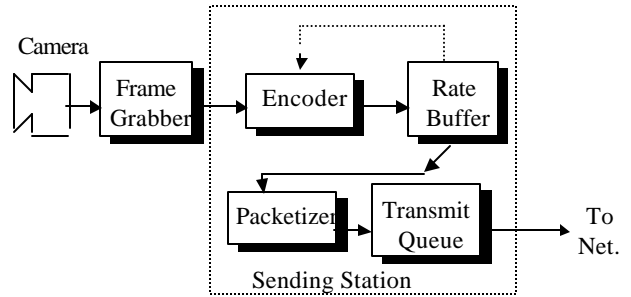The host computer packetizes the encoded video stream, and sends it over the network[14].

Figure 4 Block Diagram of a Station which Encodes and Sends a Video Stream..

A delay constraint comes from the need to support interactive communications, it is a well known fact that human beings can tolerate up to 200-250 ms of delay in two-way conversation. In a communications system using compressed video, there are delays both in the encoder and in the decoder that can be as high as 100 ms, as well as delays in the local networks to which the video stations are attached [15]. Therefore, a reasonable constraint for the wide-area component of the delay would be 40 ms. In our study, the video encoding rate is a fixed data rate of 1.5 Mbps.

3.2 Voice Traffic

The voice source is modeled as shown in Figure 5. The voice cell delay time of end-to-end have to be in the range of 250-600 ms. Figure 5 demonstrates the connection of voice sources to the ATM network. At each voice source, a continuos time, analog signal is digitized by a coder [16]. The generated samples are accumulated in a packetizer. When the number of samples in the packetizer reaches the pre-determined cell length, header is attached and an voice cell is generated. The voice cell generation process may be synchronized to an external timing.
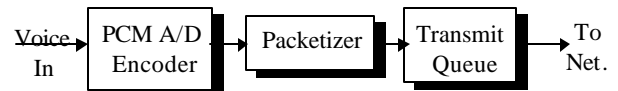

Figure 5 Block Diagram of a Station which Encodes and Sends a Voice Stream.

The generated cells are stored in the transmit buffer in the order of their generation waiting for transmission. Note that, in some LAN, the voice samples are transmitted directly using an assigned TDM channel on the network. In most other LAN protocols, the voice is transmitted in the form of packets where each packet consists of a number of voice samples within a

packetization interval [17]. The information from a single voice source can be divided into voiced and unvoiced periods. In the voiced period, a cell is generated when the information for a cell is ready. In the unvoiced period, the cell is generated as an empty cell without information which will be deleted at the user-network interface. and we can also say that the voiced period is talkspurt and unvoiced period is silent period. Each period is approximated by an exponential distribution of means $1/a$ and $1/b$ s respectively. Let T be the cell interarrival time, then the average arrival rate per source (S in cells/sec) is given by the equation (4).

$$S = \frac{1/a}{T(1/a+1/b)} \text{. ----------------------------(4)}$$

It is noted that the randomness introduced by replacing the deterministic time by an exponential one. The first burstness parameters was set $1/a$ =352ms and $1/b$ =650 ms which corresponds to a 35 % activity factor [18]. We have assumed that, all the voice sources has the same talk period at the same time in the worst case the offered load is talk periods.

3.3 Data Traffic

The data traffic is defined as message size (M_{siz}) in cells and mean interarrival time (m). We selected a fixed M_{siz}, and interarrival time has an exponential distribution with mean value of 5 ms. We suppose that the data traffic is a message arrived in specified duration. The message comes in instant (in the case of heavy load sometimes during the transmission of long message which takes more than the specified duration another message arrives) of times is sent through the network. It is to be mentioned here that the data traffic can arrive to the destination with a long delay time since the data traffic slightly effects with the long delay.

4   Performance Results
4.1 Simulator Check Point

We have considered that the network consists of a single ATM ring ADM video, voice, and data traffics. The transmission frame is fixed at 44 cells as payload depending on the channel speed at 155.52 Mbps according to the SONET standard. Simulation check point is presented here by adjusting the simulation model closed to the analytical model, the service time assumed as effective service time (0.0625ms) as shown in Figure 6. The simulation model works as the server services cells out from the queue by the rate of 352 cell/ms through the time interval of 0.0625 ms, then no cell serviced through the next interval of 0.0625 ms, and so on. The comparison between the results of the simulation and the queue models is showin Figure 7. The simulation and analytical

results, indicate that as the Offered Load (OL) increathe Mean Waiting Time (MWT) and Buffer Size (B_{siz}) are also increase up to saturation limit. Beyond the saturation limit, MWT and B_{iz} rapidly increase due to the large number of cells and queuing delay.
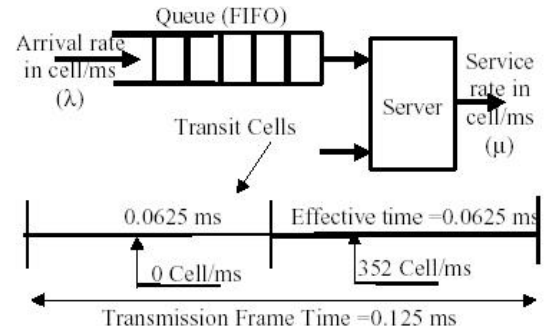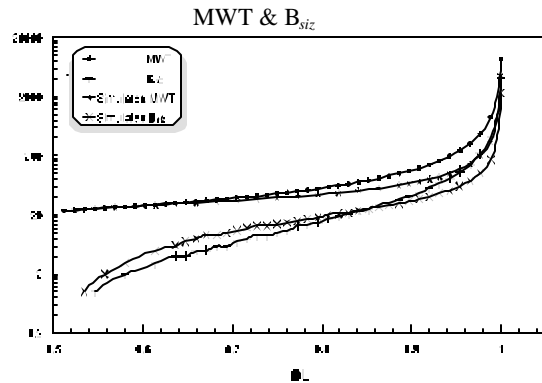

Figure 6 Simulation Queue Model for ADM/ATM Node

Figure 7 Simulation and Analytical Results

The performance study here is to find the effect of the load on the mean waiting time (MWT) and maximum buffer size (MBS). Then finger the maximum load capacity that be carried by the proposed network under multimedia traffic. The simulation programs are implemented in C language.

4.2 ATM Ring ADM Node under Multimedia traffic

The generated rate of video and data (GR $_{vi}$ + GR $_{vi}$) is depending on the generated rate of video traffic (GR$_{vi}$), and maximum number of video sources(N $_{vi}$). The ideal Maximum number of voice sources (N$_{vo}$) gets from equations (6) and (7) respectively.

$$GR_{vi} + GR_{da} = (GR_{da} + N_{vi} \times GR_{vi}) \text{ -----------------------(6)}$$

$$Max.N_{vo} = \frac{[352(cell/ms) - transit\_rate(cell/ms)] - GR_{vi} + GR_{da}]}{GR_{vo}} \text{ --(7)}$$

Figure 8 shows MWT of video and voice versus N_{vo}, and Figure 9 shows MWT of Voice and Data versus N_{vo}, for $R_{vi}=$ 1.5 Mbps, $R_{vo}=$192 Kbps, N_i sets to various values such as 10 and 20, M_{siz} sets to 100 and 300 cells, with $\mu = 5$ ms. From Figure 8 and 9, obvious that Voice MWT slightly increases up to saturation limit. Both Video MWT and Data MWT approximately remain constant up to saturation limit. Beyond the saturation limit, Video MWT, Voice MWT, and Data MWT rapidly increase due to the large number of cells of all traffics with a fixed bandwidth. Also the increases in fixed generated cells for video and data traffics, reduces N_o. Obviously that video MWT is smaller than voice MWT and voice MWT is smaller than data MWT. That is because the video traffic has highest priority of service, followed by voice traffic and then Data traffic. Table 2 shows the different fixed generated rates, the integration of video/data, and corresponding N_{vo}, video MWT, voice MWT, and data MWT.

Table 2

Parameters	Case 1	Case 2	Case 3
N_{vi}	10	10	20
M_{siz}	100	300	100
GR_{vi+da}	60	100	100
Ideal N_{vo}	227	149	149
N_{vo}	210	140	140
Video MWT	175.58	35.29	68.45
Voice MWT	304.98	950.02	532.56
Data MWT	491.49	1869.61	1003.73

Figure 10 illustrates video MBS and voice MBS versus N_{vo}, and Figure 11 illustrates voice MBS and data MBS versus N_o, for $R_{vi}=$ 1.5 Mbps, $R_{vo}=$192 Kbps, N_{vi} sets to various values such as 10 and 20 sources. M_{siz} sets to 100 and 300 cells, with $\mu = 5$ ms. Figure 10 and 11, indicate that with the increasing of N_{vo} the voice MBS slightly increases, video MBS and data MBS approximately remain constant up to saturation limit, beyond the saturation limit, video, voice, and data MBS sharply increase because of the increasing of N_{vo} increases the number of voice cells, resulting in increases in video, voice, and data MBS. The video MBS remains constant for short interval and then increases up to saturation limit. That is because the ratio of each traffic is suddenly change (N.B. the ratio of each traffic ignore the fraction so the ratio increment or decrement suddenly). Table 3 shows the different fixed generated rates of the video/data integration, and corresponding N_o, video MBS, voice MBS, and data MBS.

Table 3

Parameters	Case 1	Case 2	Case 3
N_{vi}	10	10	20
M_{siz}	100	300	100
GR_{vi+da}	60	100	100

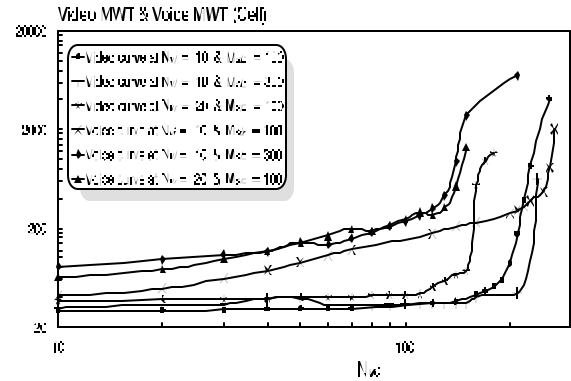| Ideal N_{vo}	227	149	149
N_{vo}	210	140	140
Video MBS	485	38	206
Voice MBS	296	1024	494
Data MBS	708	4997	1125

Figure 8 depicts MWT of video & voice vers us N_{vo} with R_{vi} =1.5 Mbps, R_{vo}=192 Kbps and μ =5 ms.
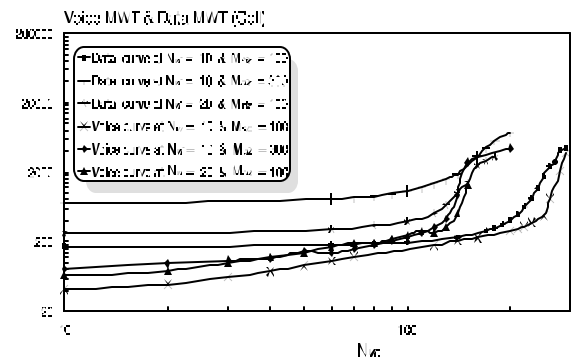

Figure 9 depicts voice MWT & data MWT versus N_{vo} with R_{vi} =1.5 Mbps and R_{vo} = 192 Kbps and μ = 5 ms.

5 Conclusion

We have presented the performance of VP-Based ring architecture based on ATM network and ADM for carrying multimedia traffics. A proposed method bears an acceptable performance for the integration of multimedia traffic, and the simulation results confirm its efficient to adapt the ratios of all traffics and provides fariness among traffics which depend on the applied offered load of each traffic, it is promising enough.
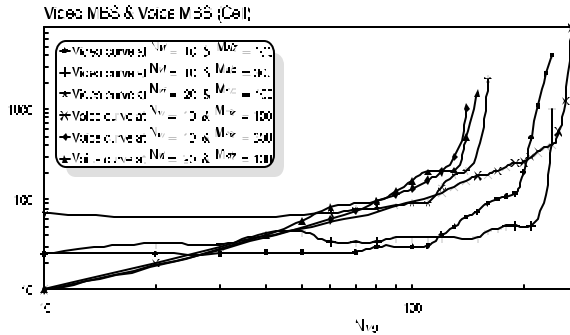
Figure 10 depicts video MBS & voice MBS versus N_{vo} with R_{vi} =1.5 Mbps and R_{vo} = 192 Kbps and $\mu = 5$ ms.
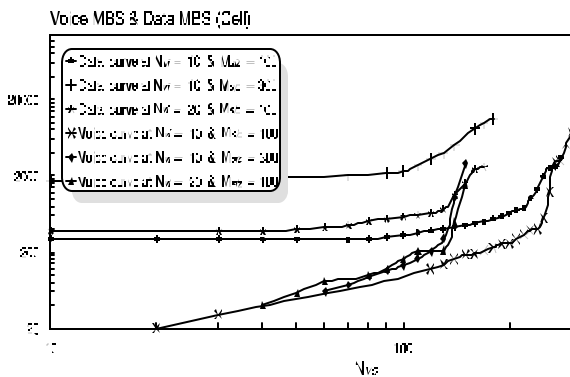

Figure 11 depicts voice MBS & data MBS versus N_{vo} with R_{vi} =1.5 Mbps and R_{vo} = 192 Kbps and $\mu = 5$ ms.

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