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Studies on Jitter Behavior and Its Reduction Scheme for Multimedia Communication in ATM Network

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Abstract

In the transmission of multimedia contents such as video and audio over ATM network, jitter is one of the most important characteristics for supporting the quality of service (QoS) of multimedia communication. The jitter is defined as the variation of interarrival time of cells at destination nodes and a large jitter may cause buffer overflow or underflow at destination nodes and, resulting in QoS degradation at application level. Therefore it is critical in multimedia communication over ATM to characterize the jitter process and to reduce the jitter to a certain level.

This thesis firstly studies the jitter behavior of MPEG2 stream on the experimental network for the project of interconnecting CATV in Hyogo Prefecture, Japan. In this network, ATM switches are connected serially and cells with two types of requirement for QoS are multiplexed; cells for MPEG2 which require real-time transmission and those for Internet packets which are much more sensitive for the cell loss ratio. A simulation model of the experimental network is constructed and the jitter process is examined under some scenarios.

Secondly, traffic characteristics of the sampled data from the experimental network are investigated in terms of long-range dependency (LRD) and self-similarity. The traffic data measured at the ATM backbone of the experimental network is analyzed statistically and it is shown that the experimental network possesses a certain self-similar nature, which is mainly due to the Internet traffic.

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Then a simulation model where the self-similar nature is taken into consideration is constructed and a relation between the self-similar traffic and the jitter of MPEG2 cells is investigated.

Finally a jitter reduction scheme for multimedia communication over ATM-ABR service is proposed. The ATM-ABR service category is originally designed for data transmission and not sufficient for real-time transmission. Although the rate control algorithm proposed recently by using queue control function provides a small delay transmission over ATM-ABR, the effect of the jitter is not taken into consideration. In this thesis, we propose a simple scheduling scheme which enforces the source node to control the cell stream upon arrival of critical cells conveying the end part of the data packet. In the proposed method, critical cells are delayed intentionally and it is expected that the packet stream at application level becomes smooth. The effectiveness of the proposed algorithm is verified by both of queueing analysis and simulation.

Keywords:

ATM Network, Jitter reduction, Multimedia communication, Queueing model, Self-similarity, Long-range dependency

ATM ネットワーク上でのマルチメディア通信における ジッタ特性とその抑制に関する研究[*]

安達 直世

内容梗概

ATM ネットワーク上でリアルタイムビデオや音楽などのマルチメディア通信を行う場合、サービス品質 (QoS) を保証する上でジッタは重要な特性のひとつである。ジッタは送信先におけるセルの到着間隔の変動として定義され、ジッタの値が大きくなるとデコーダでのバッファオーバフローやアンダーフローが発生し、結果としてアプリケーションレベルでの品質が低下する。従ってジッタ特性を把握し、アプリケーションレベルの QoS が保証される範囲内にジッタを抑制することは、 ATM ネットワーク上でマルチメディア通信を実現する上で重要となる。

本論文では、まず、兵庫県 CATV インターコネクトプロジェクトにおける実験ネットワーク上の、MPEG2 ストリームに対するジッタ特性をシミュレーションにより評価する。実験ネットワークでは ATM スイッチが多段に接続され、2 種類の異なったサービス品質を持つセルが多重化される。ひとつは MPEG2 に関するセルで実時間転送が要求される。もう一方のセルはインターネットに関するセルであり、セル損失率に関して厳しい制約を持つ。ここでは実験ネットワークに対して複数のシナリオ下でシミュレーションを行うことにより、 ATM スイッチの段数、インターネットトラヒック、他の MPEG2 トラヒックのジッタ特性に対する影響を明らかにする。

次に、長期依存性および自己相似性の観点から、実験ネットワークで測定されたトラヒックデータを統計的に解析する。具体的には実験ネットワークのバックボーンである ATM ネットワーク上での多重化セル流に対して統計解析を行い、あるレベルの自己相似性が見られること、およびそれはインターネットトラヒックに起因することを示す。さらに、自己相似性を考慮したシミュレーション

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モデルを構築し、自己相似性と MPEG2 トラヒックのジッタとの関連性を明らかにする。

最後に、ATM-ABR サービス上でマルチメディア通信を実現するためのジッタ抑制手法を提案する。本来 ATM-ABR はデータ通信用のサービスカテゴリであり、実時間転送には適していない。それに対して近年、 ATM-ABR サービスカテゴリ上でボトルネックスイッチのバッファ状態を反映させたレート制御手法が提案された。そこでは低遅延特性が実現されているがジッタの抑制については考慮されていない。そこで本研究では、データパケットの末尾セルの到着点に着目した、アプリケーションレベルのジッタ抑制を目的とするセル送出手法を提案する。本提案手法では、パケットの最終セルが同期を取るように送出されるため、アプリケーションレベルでのパケット流の平滑化が期待できる。本手法に対して待ち行列による近似解析およびシミュレーションを行い、提案手法の有効性を検証する。

キーワード

ATM ネットワーク、ジッタ抑制、マルチメディア通信、待ち行列モデル、自己相似性、長期依存性

List of Publications

Journal Papers

1. N. Adachi, S. Kasahara and Y. Takahashi, "Simulation Study on Multi-hop Jitter Behavior in Integrated ATM Network with CATV and Internet," *IEICE Transactions on Communications,* vol. E81-B, no. 12, pp. 2413-2422, December 1998.

2. N. Adachi, S. Kasahara and Y. Takahashi, "Application-Level Jitter Reduction Scheme for Multimedia Communication over ATM-ABR Service," Submitted.

International Conference Papers

1. N. Adachi, S. Kasahara and Y. Takahashi, "Simulation Study on ATM Network Integrating CATV and Internet: Experimental Testbed in Kobe for Constructing Robust Information Network against Destructive Disaster," in *Proc. 6th International Conference on Telecommunication Systems: Modeling and Analysis,* Nashville, TN, USA, March 5-8, 1998, p. 257.

2. K. Yamada, N. Adachi, S. Kasahara and Y. Takahashi, "On the Characteristics of Cell Stream in ATM Network Integrating CATV and Internet," in *Proc. 7th International Conference on Telecommunication Systems: Modeling and Analysis,* Nashville, TN, USA, March 18-21, 1999, pp. 306-315.

3. N. Adachi, S. Kasahara and Y. Takahashi, "Jitter Behavior of MPEG2 Stream in Self-similar Traffic of ATM Network Integrating CATV and Internet," in *Proc. IFIP 7th Workshop on Performance Modeling and Evaluation of ATM & IP Networks (IFIP ATM'99),* Antwerp, Belgium, June 28-30, 1999.

4. N. Adachi, S. Kasahara, and Y. Takahashi, "Active Rate-Management Algorithm for Jitter Reduction over ATM ABR Service," *INFORMS-KORMS 2000, Seoul, Korea,* June 18-21, 2000.

Other Conference Papers

1. N. Adachi, S. Kasahara, and Y. Takahashi, "Simulation Study on Jitter Evaluation for MPEG2 Cells of ATM Network Integrating CATV and Internet," in *Abstracts of the 1999 Autumn National Conference of ORSJ,* September 20-21, 1999, pp. 102-103. (in Japanese)

2. N. Adachi, S. Kasahara, and Y. Takahashi, "An Active Rate Control Algorithm for Application Level Jitter Reduction over ATM ABR Service," in *Proc. 2000 Communications Society Conference of IEICE,* B-6-1, June 30, 2000. (in Japanese)

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Chapter 1

Introduction

1.1. Background of Communication Network

The prologue of the communication network over long distance goes back to the telegraph system by Morse in 1837 and telephone communication by Alexander Graham Bell in 1876. Now telecommunication networks extend over the world with digital voice systems [32].

On the other hand, in the later 1940s the classical computer was born as a high-speed calculator. Although 60 years have passed from the first cry of new-born, the architecture of the computer has not changed until now. However, by the recent rapid progress of the semiconductor technology, we obtain an extraordinary power of calculating speed and the extensive memory capacity. This progress enables us to deal with multimedia contents such as audio and video, which require a large amount of computer resources. In addition, by the fast spread of data communication networks and Internet computers are used as the communication tool [32, 36].

In 1990 the epoch-making invention for the coding method of digital movie was published by the Advanced Television (ATV) project in the United States. By this invention the demand of multimedia communication with high speed network increases and that leads to the construction of broad-band integrated services digital network (B-ISDN) which incorporates functions and features of circuit and packet switching networks to provide existing and new services in an integrated manner in a single network [36].

1.2. Overview of ATM Network

Asynchronous Transfer Mode (ATM) is a promising technology as the backbone for the future B-ISDN. ATM technology enables us to use the high speed network and to transmit the data of the various applications with different types of traffic characteristics in a single network [15, 29]. The key concepts of ATM network are the following:

1. Virtual circuit.

   ATM network is a connection-oriented circuit switched network. We must setup the end-to-end virtual connection between source and destination nodes before sending data to the destination party. This connection setup phase is called signalling and this is done in an effort to ensure the circuit a particular quality of service corresponding to the characteristics of traffic which is transmitted.

2. Fixed-size cell.

   ATM network uses a fixed-length cell of 53 bytes. ATM cell consists of the header of 5 bytes and the payload of 48bytes, and the routing of ATM cell is decided by the index of Virtual Channel Identifier (VCI) and Virtual Path Identifier (VPI) included in the ATM header with hardware switching. The main reasons that ATM network uses fixed and small sized cells are:

(a) It is easier to implement the fixed cell switching mechanism with simple buffer hardware than the case of variable size packet.

(b) The processing time to transmit one cell is constant and there is no getting around the fact that some cells will have to wait to finish processing the large size packet. The small fixed-size cell realizes a small transmission delay by reducing the processing time at each node.

In many cases, the size of a data packet generated at higher-level layer is larger than 48 bytes, and it is not possible to fit in the payload of an ATM cell. In this case, higher-level messages are fragmented into ATM cells at source, ATM transmits the cells over network and then reassembles the fragmented cells back to higher-level messages. In ATM terminology, this is called segmentation and reassembly.

3. Statistical multiplexing

   ATM enables the demand-based multiplexing of multiple data sources over a shared link. From this effectiveness it is possible to transmit data cells efficiently.

In ATM networks any type of data which has various traffic characteristics is multiplexed in a single network and it is expected that multimedia application accounts for a large part among user applications. Especially transmitting the real-time video application will be important in the future. For transmitting the various types of traffic, the concept of Quality of Service (QoS) is very important.

1.3. Concept of QoS

ATM has a function for supporting the QoS corresponding to traffic characteristics and transmits various data in a general network. There are following two basic applications with different types of traffic characteristics [5]:

1. Voice, video and audio transmission.

   These types of traffic are sensitive to transmission delay. The transmission delay for a natural communication should not exceed 200 msec. So the network must provide the assurance for small transmission delay. On the other hand, if these types of traffic are transmitted with certain noise, the communication is arranged correctly between the parties because the mutual context supports the conversation.

2. High-speed data communication.

   The data communication is not sensitive to transmission delay and delay variation. However, it is sensitive for cell loss since in the case of the computer program, the loss of even a small part of a message causes a damage of the whole contents and transmitted data become useless.

1.3.1 Source Traffic and QoS Descriptors

Source Traffic Descriptors

The source traffic descriptors indicate various characteristics of the traffic such as above examples. The source traffic descriptor consists of following four parameters [3, 27].

- PCR (Peak Cell Rate).

  PCR traffic parameter is the maximum rate at which the source is supposed to inject traffic into the network. The PCR is the inverse of the minimum time between two cell submission to the network.

- CDVT (Cell Delay Variance Tolerance).

  CDVT is defined how much deviation from the specified minimum cell interarrival time is allowed at the interface of network.

- SCR (Sustainable Cell Rate).

  SCR is defined as an upper bound on the average rate of the conforming cells of an ATM connection, over time scales which are long relative to those for which the PCR is defined.

- BT (Burst Tolerance).

  BT is the duration of the period in which the source is allowed to submit traffic at its peak rate.

With these parameters, the maximum number of cells that can arrive at the switch at the PCR, the maximum burst size (MBS), is given as follows,

$$MBS = \lfloor 1 + BT/[(1/SCR) - 1/PCR)] \rfloor,$$

where $\lfloor x \rfloor$ is the largest integer which is smaller than or equal to x.

QoS Descriptors

The ATM provides the service categories for supporting various QoS corresponding to the traffic with above parameters. ATM service categories are defined with the following QoS parameters which describe the network performance [3].

- CTD (Cell Transfer Delay).

  The CTD is defined as the elapsed time of transmitting cells from source to destination node.

- p-pCDV (peak-to-peak Cell Delay Variation).

  The p-pCDV is the $(1 - \alpha)$ quantile of CTD minus fixed CTD that could be experienced by any delivered cell on a connection during the entire connection holding time. The value of α has not been specified yet.

- maxCTD (maximum Cell Transfer Delay).

  The max CTD is the $(1 - \alpha)$ quantile of the CTD. That is, $100(1 - \alpha)\%$ of cell delays are required to be less than or equal to the specified value.

- CLR (Cell Loss Ratio).

CLR is maximum percentage of lost cells and defined for a connection as:

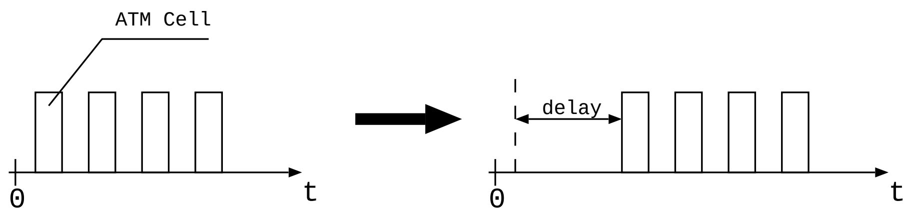
$$CLR = \frac{Lost\ Cells}{Total\ Transmitted\ Cells}.$$

1.3.2 QoS for Multimedia Communication

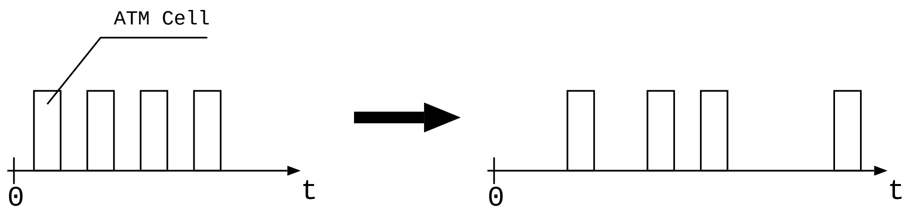
In the transmission of multimedia contents, the cell-transfer-delay CTD is one of the important elements for supporting the QoS. In addition, cell-delay-variation (CDV), or equivalently the jitter, is also an important element for supporting the QoS of multimedia communication.

The jitter is defined as the variation of the interarrival time of cells at destination node. Figure 1.1 shows the ideal cell-transmission and the real cell-transmission. When data cells are transmitted to the destination without any other traffic, the cells of traffic with constant interval are delayed with same transmission delay. But when there are any other background traffic, the interarrival time of cells transmitted to the destination varies due to other traffic cells.

A large jitter value causes buffer overflow or underflow at the destination node and results in the QoS degradation at application level. Therefore it is crucial for multimedia communication over ATM to characterize the jitter processes and to reduce the jitter to a certain level.



Ideal Transmission



Real Transmission

Figure 1.1. Jitter.

1.4. Service Category

In ATM network, there are five types of service class categories for supporting the various QoS [3].

- Constant Bit Rate (CBR).

  The CBR service category is used for the connections which require the static amount of bandwidth characterized by a PCR value. Once the connection is established, the source reserves a constant amount of bandwidth which is indicated by the PCR during the connection lifetime. When all cells are conforming to the relevant conformance tests, the negotiated QoS is assured to all cells.

- Real-Time Variable Bit Rate (RT-VBR).

The RT-VBR service category is intended for the real-time applications such as the voice and video applications which require tightly constrained delay and delay variation.

- Non Real-Time Variable Bit Rate (NRT-VBR).

  The NRT-VBR service category is used for the transmission of non-real-time applications which have bursty traffic characteristic and expect a low cell loss ratio.

- Unspecified Bit Rate (UBR).

  The UBR service category is intended for non-real-time applications which do not require tightly constrained delay and delay variation such as the traditional computer communications applications, i.e., file transfer and email.

- Available Bit Rate (ABR).

  The ABR service category is designed for data transmission which does not require the stringent transmission delay and delay variation. In this service category the available bandwidth may vary, but does not become less than MCR. The feature of this service class is the feedback loop control mechanism. As for this feedback loop control mechanism, we describe later.

The CBR, RT-VBR and NRT-VBR service classes are high-priority classes for supporting QoS of the real-time application data such as the video streaming and voice application, which require the strict restriction for the transmission time. Since these service categories reserve larger bandwidth than the actual

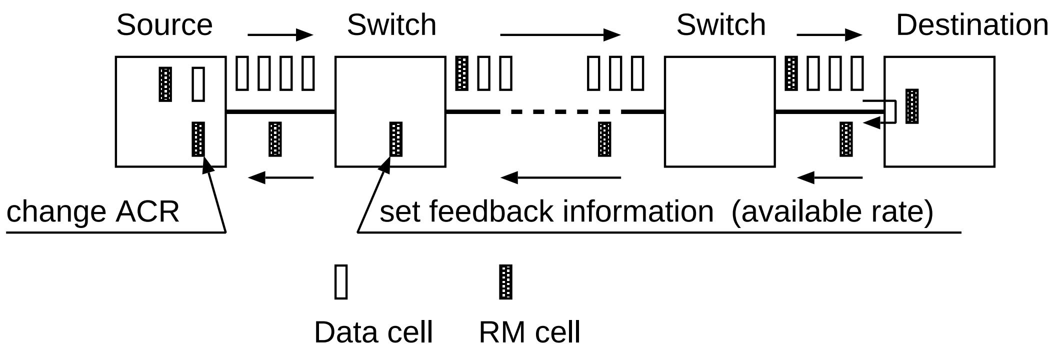


Figure 1.2. Feedback Scheme of ABR Service Category.

demand for supporting the stringent requirement for transmission time, it causes the wastefulness of the link capacity.

ABR and UBR service category classes achieve the high bandwidth utilization. The UBR service category does not guarantee any QoS such as cell loss or transmission delay. This service class is used for the best effort communication.

The ABR service class is designed for data transmission. The difference of ABR service class from UBR is that ABR guarantees only the CLR and does not provide any other QoS guarantees such as the CTD and CDV. The mechanism for supporting CLR in ABR service category is based on the feedback control where the allowed cell rate (ACR) of the source node is dynamically adjusted by the resource management (RM) cells which are the feedback signal with the information of the congestion state of ATM network (see Figure 1.2). By this mechanism, ACR varies according to the congestion state of the network and hence ABR service category can achieve low CLR. There are a number of previous works for the rate control algorithm of ABR service category such as EFCI, ERICA and ERICA+ [12, 18].

1.5. Long Range Dependency and Self-similarity

For the appropriate design of network services, it is necessary to evaluate the network performance carefully. Here we note the concept of long range dependency and self-similarity, which will cause degradation of the network performance.

Packet arrivals are often modeled as the Poisson process for analytic simplicity, even though a number of traffic studies have shown that packet intervals can not be modeled with exponential distribution. Markovian arrival processes would have a characteristic burst length which would tend to be smoothed by averaging over a long enough time scale. However, the importance of long-range dependence (LRD) in network traffic has been observed in studies such as [7, 16, 19], which show aggregating streams of network traffic typically intensifies the self-similarity instead of smoothing. Afterwards, the significant traffic variance is present over the wide range of time scales. Markovian arrival processes can not afford to describe the phenomenon sufficiently, yet the notion of self-similar processes can describe these burst and correlation natures on time series.

In following two subsections, we summarize mathematical definitions of LRD and self-similarity.

1.5.1 Long Range Dependency

Let X_t denote the discrete stochastic process which satisfies the following condition

$$\lim_{k\to\infty} \frac{\rho(k)}{C_\rho k^{-\alpha}} = 1, \quad 0 < \alpha < 1, \tag{1.1}$$

where C_ρ is a positive constant and $\rho(k)$ is the auto correlation function of X_k defined by

$$\rho(k) = \frac{CovX_t X_{t+k}}{VarX_t}. \tag{1.2}$$

Then it is said that the process X_t possesses the LRD property.

On the other hand, when X_t satisfies the following condition

$$\lim_{k\to\infty} \frac{\rho(k)}{C_\rho \alpha^k} = 1, \quad 0 < \alpha < 1, \tag{1.3}$$

it is said that the process X_t is the short range dependent process.

From now, we use the parameter $H = 1 - \alpha/2$ instead of α. When we use the parameter H, the condition of LRD property is $0.5 < H < 1$. The parameter H is called Hurst parameter and this is also the index of self-similarity.

In general the traffic with LRD causes the growth of queue length at buffer in ATM switch and degration of network performance.

1.5.2 Self-similarity

Let X_t denote the stochastic process and define the aggregated series $\{X_k^{(m)} : k \geq 1, m = 1, 2, \cdots\}$ by summing the original series X_t over non-overlapping blocks of size m. That is,

$$X_k^{(m)} = \frac{1}{m} \sum_{i=1}^{m} X_{m(k-1)+i}, \quad k \geq 1. \tag{1.4}$$

Let σ^2 denote the variance of X_k, $\rho(k)$ and $\rho^{(m)}(k)$ the auto correlation function of X_k and $X_k^{(m)}$, respectively.

There are following two definitions of the self-similar process.

- If for all $m = 1, 2, \cdots$,

$$VarX^{(m)} = \sigma^2 m^{2H-2}, \quad 0.5 < H < 1, \tag{1.5}$$

then X_k is called the rigorous second order self-similar process with Hurst parameter H.

- If

$$VarX^{(m)} \sim L(m) m^{2H-2}, \quad as \quad m \to \infty, \tag{1.6}$$

then X_k is called asymptotically second-order self-similar with the Hurst parameter H. In (1.6), $L(m)$ is a slowly varying function at infinity and $a(x) \sim b(x)$ means

$$\lim_{x \to \infty} \frac{a(x)}{b(x)} = 1. \tag{1.7}$$

1.6. Related Works

The jitter in ATM networks has been studied extensively and several analytical models were proposed. One of pioneering works was done by [31], where the jitter is characterized by a random delay component. The departure process was analyzed by assuming the Markovian behavior of the delay component. [20, 21, 22] analyzed the departure process of tagged cells under which the tagged cell stream is multiplexed with the background stream. In [9], the authors considered the message transmission where messages consist of consecutive cells. They analyzed the message delay process which is defined as the time elapsing from the arrival epoch of the first cell of the message to the epoch when the last cell is transmitted.

[9] and [21] considered the jitter behavior under the multiple-node case, where the tagged cell stream is superposed with the background stream. They analyzed it by the approximation technique or performing simulation due to an analytical difficulty.

On the other hand, the statistical characteristics of self-similarity and long-range dependency have been shown to apply to all sorts of networks such as Ethernet LAN, WAN, and ATM network [7, 16, 19], where the authors showed aggregating streams of network traffic typically intensifies the self-similarity instead of smoothing. In general, the self-similar traffic with LRD degrades the network performance and hence it is important to capture the traffic characteristics in terms of LRD for investigating the performance measures such as cell loss probability, transmission delay, and jitter process.

As for the QoS supporting, [37] and [38] proposed a design method with the queue control function which is used for calculating the bandwidth allocated to the source node. Using the queue control function, it is possible to control the transmission delay and to achieve low CLR by adjusting the allowed cell rate (ACR) according to the queue length of the bottleneck switches along the path. The queue control function algorithm is quite attractive since the ABR service category can support the multimedia communication with small delay. However, the jitter is not taken into consideration in their algorithm. The jitter is also important for the real-time video transmission where the jitter affects the quality of the decoded video at the destination node.

1.7. Purpose of This Study

As we describe in the previous section, the jitter is an important element for supporting the QoS of multimedia communication. It is crucial for multimedia communication over ATM to characterize the jitter processes and to reduce the jitter to a certain level. In addition, it was reported that the self-similar traffic degrades the network performance and hence it is also important to investigate how the self-similar nature affects the jitter of tagged cells.

Hence, the objective of this study is to investigate the jitter processes and to propose a scheduling scheme at the source node to reduce the jitter at application level. The outline of this dissertation is as follows.

In Chapter 2, we examine the jitter behavior of MPEG2 stream on the experimental network for the project of interconnecting CATV in Hyogo Prefecture, Japan, by simulation. In this testbed network, ATM switches are connected serially and the cells with two types of requirement for QoS are multiplexed; cells for MPEG2 which require real-time transmission and those for Internet packets which are much more sensitive for the cell loss ratio. Constructing a simulation model of the experimental network, we investigate the jitter processes under some scenarios and show how the jitter process is affected by the Internet traffic and the other cell streams of MPEG2. Furthermore, we study the effect of the number of ATM switches on the jitter process when more CATV networks are added serially.

In Chapter 3 we investigate the traffic characteristics of the experimental network in terms of the long-range dependency (LRD) and self-similarity. We analyze the traffic data measured at the ATM backbone of the experimental network statistically and show that the experimental network possesses a certain self-similar nature which is mainly due to the Internet traffic. Then we construct the simulation model where the self-similar nature is taken into consideration and investigate the relation between the self-similar traffic and the jitter of MPEG2 cells.

In Chapter 4, we propose a scheduling scheme at the source node to reduce the jitter at application level under the ATM-ABR service class. The ATM-ABR service category is originally designed for data transmission and not sufficient for the real-time transmission. In our proposed scheme, we focus on the departure point of critical cell corresponding to the last part of the data packet. The critical cell is intentionally delayed until the next data packet generation and transmitted at the beginning of the next cycle of packet generation. According to this scheme, the departure points of critical cells at the source node are like the CBR traffic and therefore the reduction of the jitter at application level is expected. Since the points of sending the critical cells are intentionally delayed, we call our proposed scheme intentionally delayed transmission (IDT). The strong point of IDT scheme is that we need not change the existing ATM facilities except the source node. We verify the effectiveness of the proposed algorithm by the analytical model with queueing theory and simulation.

Finally we conclude the dissertation in Chapter 5.

Chapter 2

Jitter Behavior of MPEG Stream

2.1. Introduction

In this chapter, we present the wide area project "The Project of Interconnecting CATV in Hyogo Prefecture, Japan" aiming at the inter connection of presently existing CATV networks in Kobe, Nishinomiya and Amagasaki through three ATM switches. In this testbed network, ATM switches are connected serially and the cells with two types of requirement for QoS are multiplexed; cells for MPEG2 which require real-time transmission and those for Internet packets which are much more sensitive for the cell loss ratio. We construct a simulation model of the experimental network and investigate the jitter processes under some scenarios. We show how the jitter process is affected by the Internet traffic and the other cell streams of MPEG2. Furthermore, we study the effect of the number of ATM switches on the jitter process when more CATV networks are added serially.

This chapter is organized as follows. In Section 2.2, we present "The Project of Interconnecting CATV in Hyogo Prefecture, Japan" in detail. In Section 2.3, we show our simulation model and give the explanations of modules developed for the simulation. In Section 2.4, we present the simulation results.

2.2. The Project of Interconnecting CATV in Hyogo Prefecture, Japan

After the destructive disaster of Hanshin-Awaji earthquake, we have recognized that it is significantly important to construct the robust information infrastructure for transmitting large volume of emergent information on rescue calls, supply schedules, safe refuge places, and so on. It has been reported that the coaxiable cable for CATV was less damaged even by the earthquake of magnitude seven and that the optical fiber cable laid on the rail way was not damaged [30] . From these reasons, the project of interconnecting CATV and Internet has been planned in Hanshin area since 1996. The main objective of the project is to construct the robust interactive information network with optical fibers laid on the rail way, and with coaxial cables which have been deployed in the wide area.

On the other hand, recently, there exists rapidly growing interest in the interconnection of CATVs which integrates broadcasting and communication services aiming to support multimedia traffic. ATM technique and optical fiber trunk lines are promising for this purpose and adopted in this project, which is the first trial in Japan to provide both broadcasting and communication services.

The pioneering challenge in this project is to efficiently integrate these two kinds of traffic stream of entirely different characteristics. The effectiveness of traffic management and control mechanisms in the integrated circumstance of broadcasting and communication data traffic through ATM networks is the crucial matter to be studied. The efficient use of available bandwidth is also an important subject to be covered in the project.

The construction of the testbed for the project was completed at the end of February, 1998 and the experiment has been started since March, 1998. This research project has been named as "Research and Development on Traffic Modeling and Management/Control Techniques for the Integration of Broadcasting and Communication Services through Optical Fiber Trunk Lines Cascading CATV Networks".

The research has been performed by Kobe Multi-node Integrated Connection Research Center (KOMIC [41]), which has been established by Telecommunications Advancement Organization of Japan (TAO). KOMIC research center carries out advanced research and development towards the effective integration of broadcasting and communication traffic through ATM network connecting three existing CATVs with optical links.

Furthermore, the application of the outcome of this project enables the sharing of information pertinent to everyday life as well as the broadcasting of local programs produced by one of the CATVs. It also functions as a reliable infrastructure corresponding to a variety of local needs to mutually back up the CATVs through inter-connection by optical fiber trunk lines, in emergency cases when the facilities of one of the CATVs is damaged by natural disaster such as earthquakes.

The research and development areas of this project are as follows.

1. Simulation Technology for ATM Network.

   The qualitative evaluation of the effect of hops through ATM network on the performance measures such as transmission delay and jitter is carried out by simulation [14].

2. Image Transmission Technology through ATM network.

   (a) Image transmission experiment in emulated traffic environment

   (b) Image transmission experiment in real traffic environment

3. Robust Image Transmission Technology for Transmission Delay and Jitter. Based on performance measurements of image transmission experiments in the real traffic environment and simulation studies on ATM networks, robust technology will be discussed and developed.

In this chapter, we examine the jitter processes of decoding points by the simulation. According to [21], we consider the inter-arrival time of tagged MPEG cells to the destined decoder. Here we define the jitter as the variance of inter-arrival time of tagged MPEG cells.

In our testbed, cells with two types of requirement for QoS are multiplexed: cells for MPEG2 which require the real time transmission and those for Internet packets which are much more sensitive for the cell loss ratio. In the following, we refer to the cells for MPEG2 as the MPEG cells and the cells for Internet packet as the Internet cells. In our simulation model, each MPEG stream is connected under CBR class and it is copied and sent in accordance with the multicast function of the ATM switch. All connections for MPEG2 and Internet streams are established with PVC. The tagged MPEG stream contends for the output port with the other MPEG stream and the Internet packet stream. We investigate the jitter processes under some scenarios and show how the jitter

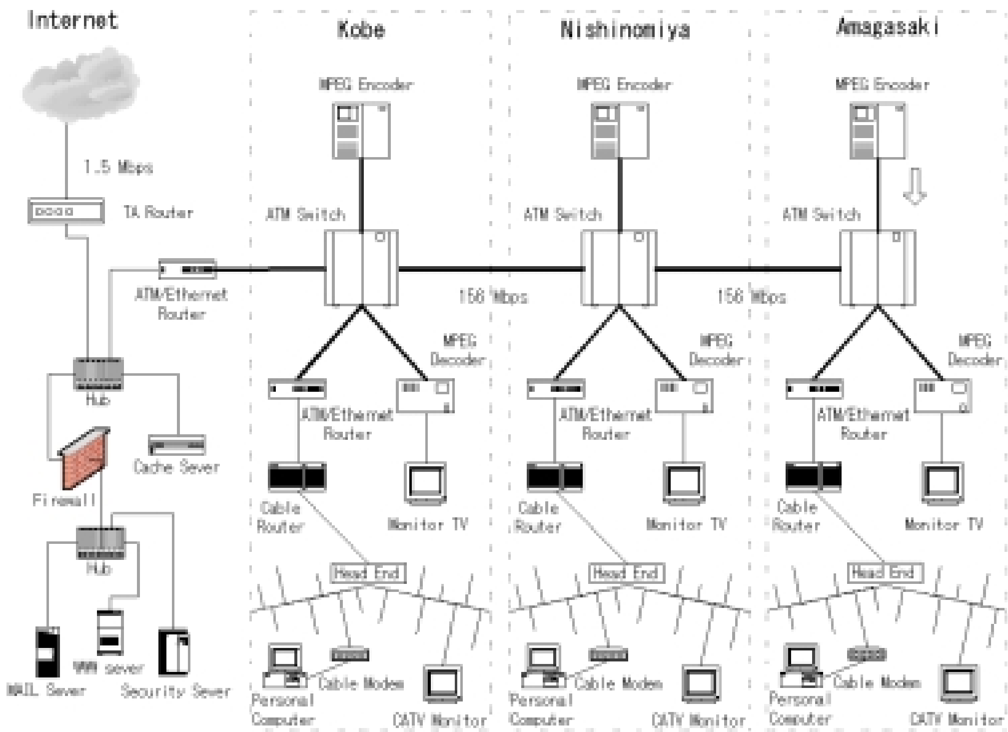


Figure 2.1. Testbed Network in Hyogo Pref.

process is affected by the Internet traffic and the other cell streams of MPEG2. Furthermore, we study the effect of the number of ATM switches on the jitter process when more CATV networks are added serially.

2.2.1 Testbed Network and Its Equipments

The outline of the testbed system is as follows (see Figure 2.1).

There are three CATV companies in Hanshin area; Kobe, Nishinomiya and Amagasaki. An ATM switch is equipped in each CATV and these CATVs are connected serially in the above order. Each company provides the video service to the rest of companies using the MPEG2 over ATM. The video service includes not only the standard video but the TV program broadcasting. Each MPEG2 stream is sent to the other two CATV companies according to the function of multicast implemented in ATM switch. In the area where companies provide the CATV service, fifty subscribers receive the standard CATV broadcasting. In addition, those subscribers are able to access the Internet using the cable modem. That is, Internet services such as E-mail, ftp and Web-browsing are supported through the CATV network. In the coverage of each CATV, fifty subscribers utilize Internet connection using cable modems as well as standard CATV broadcasting service.

Now we present the details of our testbed network in the following subsections.

Optical Transmission Path

In order to transmit massive volume of broadcasting and communication data at a high speed through Kobe, Nishinomiya and Amagasaki, optical fiber cables have been laid along the railway of the Hanshin Railroad Co. for major portions and electric poles for the rest of them.

Transmission Facilities

In order to achieve high speed transmission in the network which integrates broadcasting and communication data, an ATM switch fabric having high speed switching capability and easy scalability has been installed at each CATV company. MPEG2 encoder/decoder system enables transmission of images and visual data of broadcasting TV programs in a digital compressed form through ATM networks. Owing to this technology, broadcasting programs as well as local programs produced by one of the three CATVs can be shared among them, which makes it possible to develop an advanced image transmission system. Furthermore, network monitoring and analyzing equipment including ATM analyzer is provided to facilitate the transmission experiments effectively.

Internet Facilities in CATV System

Approximately fifty households are selected from among CATV subscribers and provided with cable modems for Internet access so that the characteristics of Internet traffic are clarified.

Internet Server Facilities

Internet related servers such as a WWW server for information retrieval and a mail server for mail delivery are also equipped in the network so that the qualitative evaluation of the effect of Internet traffic on other types of traffic, especially on moving image traffic can be performed in a real environment. WWW server is also utilized to announce the outline and outcome of the project.

Monitoring and Measuring Equipments

Simulation tools and high performance workstations are prepared to develop simulation models and to execute simulation programs. An ATM analyzer is also equipped to monitor and analyze traffic streams from multimedia sources in the real system, which makes it possible to monitor and to measure the QoS of transmission in the integrated environment from theoretical and practical points of view.

Remark: Note that our ATM network has a serial topology. For the future plan, about ten CATV companies will be connected serially since those are located serially along the Hanshin railway. From the economical point of view, the serial connection of CATV companies seems to be reasonable. If we focus our attention on the construction of more robust network against the disaster, we have to discuss the network topology from various points of views. However our main purpose here is to investigate the effect of the multi-hop jitter behavior of MPEG stream multiplexed with the Internet cells.

2.3. Simulation Model

In our simulation study, we used OPNET[25], a comprehensive software environment for modeling, simulating, and analyzing the performance of communications networks, computer systems and applications, and distributed systems. In this section, we describe the modules originally implemented for our simulation model and also show the parameters used in those modules.

2.3.1 Modules

Though OPNET provides the extensive modules for modeling the communications networks, it was needed to develop the original modules for realizing the certain features in our integrating network system. In our simulation model, we have implemented the following modules.

1. MPEG2 encoder

2. MPEG2 decoder

Table 2.1. Parameters of MPEG2 Encoder

Parameter Name	Description
Inter-arrival time	Mean inter arrival time of MPEG cell
Inter-arrival pdf	PDF of inter arrival time of MPEG cell
source_number	Identifier of module

3. ATM switch

4. Client module which generates Internet cells

5. Module of Internet Server

In the following, we show above modules in detail.

MPEG2 encoder

MPEG2 encoder module generates MPEG cells according to a certain probability distribution function. When MPEG2 encoder generates MPEG cells, it sets the specified flag in the header of each cell, according to which the ATM switch functions for the multicast transmission. In Table 2.1, the parameters used in this module are presented.

MPEG2 decoder

This module accepts MPEG cells transmitted from the MPEG2 encoder. It also measures inter-arrival times of the MPEG cells generated by each MPEG2 encoder. The data are obtained by the function of Probe of OPNET.

Table 2.2. Parameters of ATM Switch

Parameter Name	Description
ATM Switch Fabric Delay	Switching time in ATM switch
sw_number	Number of copies of MPEG cell and index for routing
copy_port	Port number to which MPEG cell and copied cell are sent.
server_port	Port number for Internet server
client_port	Port number for the destination client corresponding to sw_number
path_port	Port number for Internet cells

ATM Switch

The module of ATM Switch simulates the switching function of arriving cells and performs the multicast if it is requested. The procedure of the multicast is presented in Subsection 2.3.2. In Table 2.2, the parameters set in the module are presented.

The standard module of ATM switch provided by OPNET supports SVC and does not support PVC. Thus, we have developed the new module of ATM switch which realizes the pseudo PVC. We also have implemented the function of multicast on the switch module. The multicast process is performed at each ATM switch module when the cell with multicast request arrives. According to this module, routing for both MPEG and Internet cells is performed and copying of MPEG cell is processed.

Client module which generates Internet cells

This module generates Internet cells whose inter-arrival times are exponentially distributed. When the module generates cells, it sets the flag with which each cell is identified as a Internet cell. On the other hand, if the client module accepts cells sent by the Internet server, accepted cells are discarded and lost. In Table 2.3, the parameters set in the module are presented.

Table 2.3. Parameters of Client Module (IC: Internet Cell)

Parameter Name	Description
Inter-arrival time	Mean inter-generating time of IC
Inter-arrival pdf	PDF of inter-generating time of IC
source_number	Identifier of module

Module of Internet Server

When this module receives Internet cells generated by the client, the module returns them to the client.

Remark: In our simulation, we do not implement the TCP/IP protocol which is provided by OPNET as a default module. Since the standard ATM modules of OPNET support only SVC, we tune the parameters for realizing the pseudo PVC of MPEG stream. In this situation, the simulation time on SUN-4 workstation is about three to five seconds while the run time for a simulation program is over two hours. When we use the TCP/IP module, it is hard to set the SVC for Internet path and to transmit Internet cells within five seconds of simulation time. From this reason, we implement an Internet-cell generator which generates cells directly

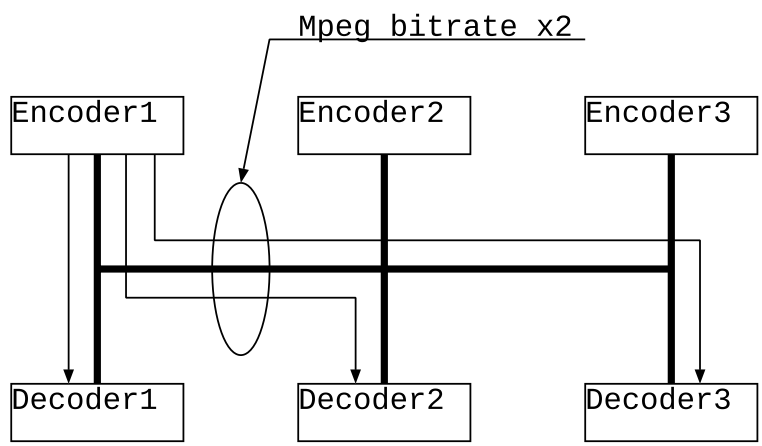


Figure 2.2. Transmission without Multicast

at ATM layer. However, this does not simulate the real TCP/IP application and hence implementing the TCP/IP modules on our simulation program is one of the important issues to be improved for future study of our testbed network.

2.3.2 Multicast of ATM Switch

In this subsection, we present the multicast procedure implemented on our simulation model.

If multiple copies of MPEG cells are sent to each CATV station without using multicasting function, it wastes the bandwidth of the link between ATM switches as shown in Figure 2.2. On the other hand, if MPEG cells are multicasted, it achieves the effective usage of the bandwidth of each link (see Figure 2.3). It also reduces the number of cells to be switched at the ATM switch. This load reduction of the ATM switch is more effective when the number of ATM switches increases.

There have been a number of researches concerning the ATM multicast [6,

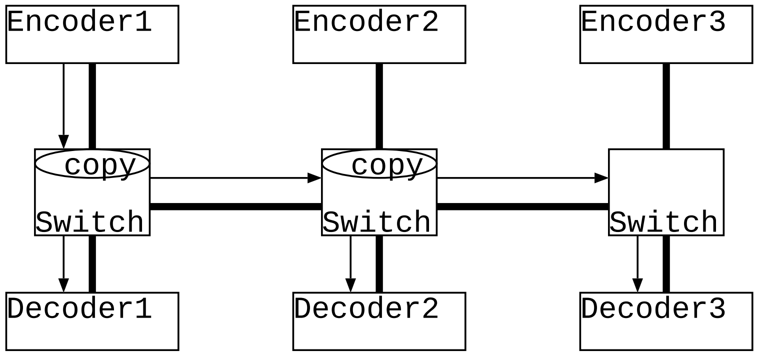


Figure 2.3. Transmission with Multicast

10]. We have implemented the following multicast procedure on the ATM switch module (see Figure 2.4). The copy network is equipped in front of the ATM switch and this network copies the required amount of cells from the original MPEG cell.

In this procedure, the cell which is not for the multicast also goes through the copy network and hence it increases the delay of the cell transmission. The strong point of this procedure is that it is easy to implement on OPNET.

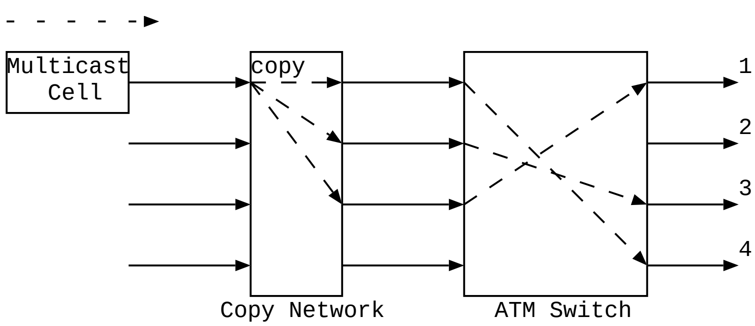


Figure 2.4. Copy Network Implemented on Simulation

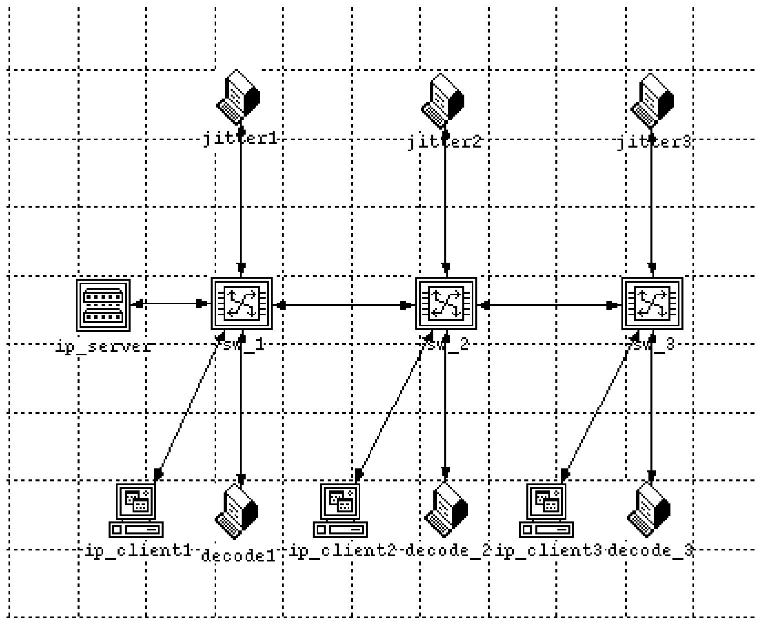


Figure 2.5. Network Model on OPNET

2.3.3 Simulation Model

Using the modules described in the previous subsection, we finally construct the simulation model of the testbed network as shown in Figure 2.5. Here, assumptions and parameters set in the simulation are all based on the technical specification prescribed by Hyogo Prefecture.

In Figure 2.5, jitters 1 to 3 are modules of MPEG2 encoders. Jitter 1 is corresponding to Kobe, jitter 2 to Nishinomiya, and jitter 3 to Amagasaki. As for the cell-generating rate of the MPEG2 encoder, λ, the range 3 to 14 Mbps is specified. Hence we investigate the jitter process with the three values of the generation rate: 4, 8, and 13 (Mbps), respectively.

The tagged MPEG stream is generated with the constant bit rate λ and the others are generated according to the Poisson process with rate λ. This avoids the dependence of the initial time of the MPEG cell generating process. Since the size of a cell is equal to 53 bytes, we set the values of Inter-arrival time for MPEG encoder module as shown in Table 2.4. In the real MPEG encoder, the raw data stream is considered as bursty because of the different sizes of I, P, and B pictures. As for the MPEG encoder with the function of MPEG2 over ATM, which is equipped in our testbed network, the output stream is realized as the CBR owing to the feedback control within the encoder. Hence we don't need to consider the GOP level for modeling the MPEG stream.

Table 2.4. Values of Bit Rate of MPEG Encoder

Case	Value (second)
1	$53 \times 8/(4 \times 10^6) \simeq 1.0 \times 10^{-4}$
2	$53 \times 8/(8 \times 10^6) \simeq 5.0 \times 10^{-5}$
3	$53 \times 8/(13 \times 10^6) \simeq 3.0 \times 10^{-5}$

In Figure 2.5, decodes 1 to 3 are the modules of MPEG decoder. sw 1 to 3 are the modules of ATM switch. In the testbed network, the switching capacity is assumed to be at least 5 Gbps. Hence we choose 6Gbps as the value of the switching capacity and set ATM Switch Fabric Delay in Table 2.2 equal to $53 \times 8/(6 \times 10^9) \simeq 7.0 \times 10^{-8}$ (sec.).

The capacity of all links is equal to 156Mbps and we assume no propagation delay of the link. The links for MPEG and Internet streams are all established under CBR class.

In Figure 2.5, ip_server is the module of the Internet server. ip_clients 1 to 3 are the modules of Internet client. As for the bit rate of Internet cells, let N denote the number of ATM switches. We assume that all ip_clients generate

Internet cells with the same bit rate. In addition, the arriving Internet cell to the server is returned to the client. Thus, the maximum bit rate with which the client generates Internet cells, W, is given by $156/2N$ (Mbps). For example, if $N = 3$, the maximum bit rate is given by

$$W = \frac{156}{2 \times 3} = 26 \quad \text{(Mbps)}.$$

To investigate how Internet cells affect the jitter process, we set their bit rate equal to 0, $156/4N$, and $156/2N$ (Mbps), respectively. We assume that all Internet clients generate cells according to exponential distributions with the same rate.

Finally, we investigate the jitter process when N is equal to 3, 4, and 5, respectively. In the cases of $N = 4$ and 5, the ATM switches are added serially to the right direction. That is, the 4th ATM switch is equipped on the right side of Amagasaki. Similarly, the 5th ATM switch is equipped on the right side of 4th one.

2.4. Simulation Results

We show our simulation results in Figures 2.6 to 2.16. In all figures, the horizontal axis represents the location of the CATV company and the vertical axis means the jitter value, i.e., the variance of inter-arrival time of tagged MPEG cells (second^2). Note that the scale of the vertical axis is in the order of 10^{12}. In Table 2.5, we summarize the parameters of Figures.

First, we show the mean inter-arrival time of MPEG cell at each MPEG2 decoder in Table 2.6. From this table, it is observed that inter-arrival times of

Table 2.5. Parameters List of Simulation (MC: MPEG cell, IC: Internet cell)

Figure	N	MC (Mbps)	IC (Mbps)
2.6	3	4	0
2.7	3	4	13
2.8	3	4	26
2.9	3	8	0
2.10	3	8	13
2.11	3	8	26
2.12	3	13	0
2.13	3	13	13
2.14	3	13	26
2.15	4	8	13
2.16	5	8	13

Table 2.6. Mean Inter-arrival Time of MPEG Cell (10^{-5}s, MPEG cell = 13Mbps, Internet Cell = 26Mbps)

From / To	Kobe	Nishinomiya	Amagasaki
Kobe	$3.00 \pm (7.29 \times 10^{-4})$	$2.99 \pm (3.75 \times 10^{-3})$	$3.00 \pm (4.06 \times 10^{-3})$
Nishinomiya	$3.00 \pm (1.53 \times 10^{-3})$	$2.99 \pm (7.70 \times 10^{-4})$	$3.00 \pm (1.32 \times 10^{-3})$
Amagasaki	$3.00 \pm (1.97 \times 10^{-3})$	$3.00 \pm (1.56 \times 10^{-3})$	$3.00 \pm (7.34 \times 10^{-4})$

MPEG cells are almost same.

In Figures 2.6 to 2.8, we show the jitter values with MPEG cell rate equal to 4Mbps when the Internet cell rate is set to 0, 13, and 26 (Mbps), respectively.

In Figure 2.6, we observe that the jitter values are symmetric at the decoder point of Nishinomiya. In this case, there are no Internet cell streams in the network and hence the MPEG traffic are symmetric at Nishinomiya. We also observe that the jitter values of MPEG data generated at Kobe and Amagasaki become large as the number of ATM switches increases.

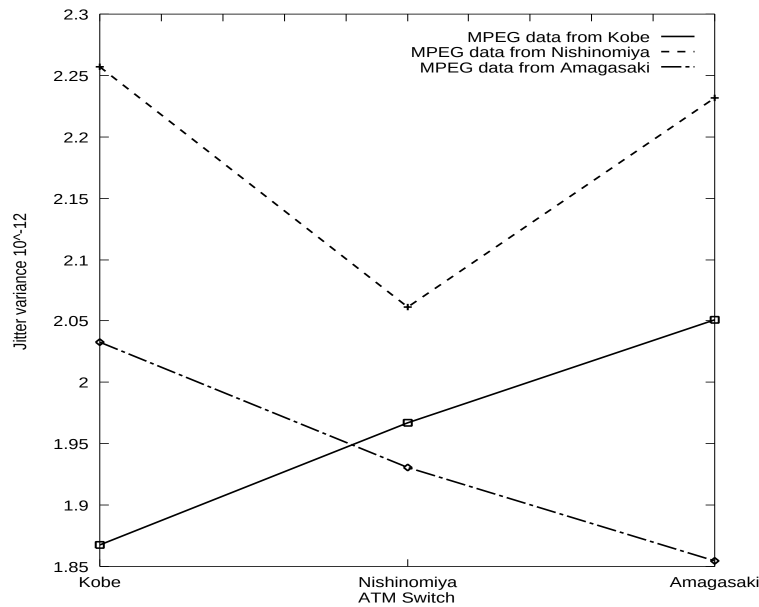


Figure 2.6. Jitter Values: MPEG = 4Mbps, IP = 0bps

In Figure 2.7, the jitter value of MPEG cell from Kobe becomes large as the number of ATM switches increases. The jitter value of MPEG cell from Nishinomiya to Kobe is larger than that to Amagasaki. This is because the Internet traffic is mainly directed to Kobe. Also, the jitter value of MPEG cell from Amagasaki is becoming large as the number of ATM switches increases. In particular, the increase of the jitter at Kobe is larger than that at Nishinomiya because of the aggregation of the Internet traffic at Kobe.

Figure 2.8 is the case that the bit rate of Internet cells is largest in the three figures. Figure 2.8 shows the same tendency of Figure 2.7. Note that at the decoder of Kobe, the jitter values are quite larger than those of Figure 2.7. This means that the Internet cell with high bit rate largely affects the jitter at the decoder of Kobe.

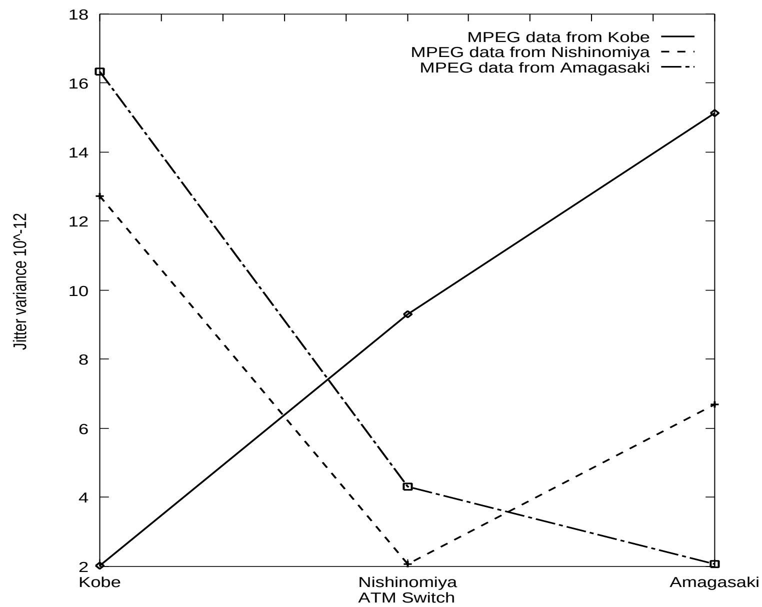


Figure 2.7. Jitter Values: MPEG = 4Mbps, IP = 13Mbps

Figures 2.9 to 2.11 are with the bit rate of MPEG cells equal to 8 Mbps under Internet cell equal to 0, 13, and 26 (Mbps), while Figures 2.12 to 2.14 are with 13 Mbps of MPEG cell under 0, 13, and 26 of Internet cell. We can observe the same tendencies of Figures 2.6 to 2.8.

Now we compare above figures under the same bit rate of Internet cell. Figures 2.6, 2.9 and 2.12 are under the case that there is no Internet stream in the network. When the bit rate of MPEG cells becomes large, the jitter value of MPEG cell from Kobe (Amagasaki) is getting large at the decoder at Nishinomiya but becoming small at Amagasaki (Kobe). (Figure 2.12 is remarkable.) That is, the jitter does not always become large when the number of ATM switches increases. One of the reasons is that when the MPEG cell traverses the cascaded ATM switches under the homogeneous traffic environment, the jitter tends to

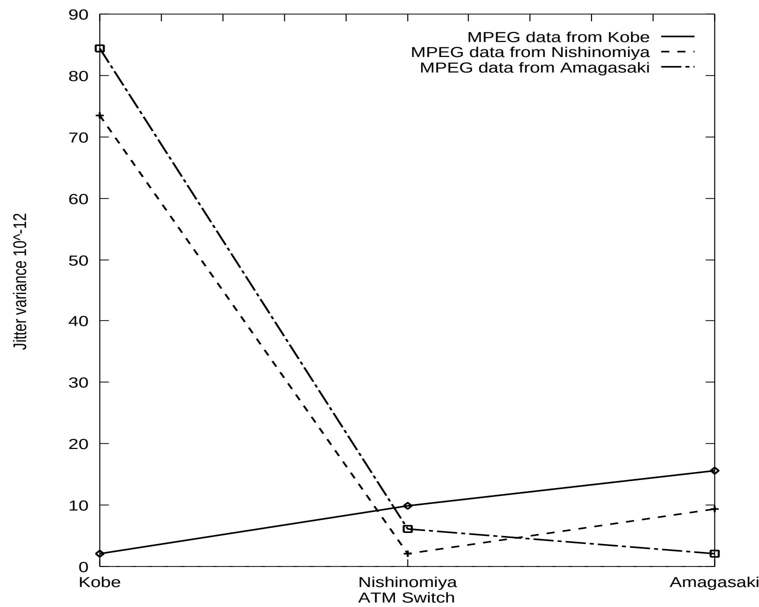


Figure 2.8. Jitter Values: MPEG = 4Mbps, IP = 26Mbps

decrease (see [2] in details).

Figures 2.7, 2.10 and 2.13 are under the case that the bit rate of Internet cells equals 13 Mbps. The tendency of the variations of curves is same. Note that the ranges of the jitter value under three figures are not so different when the bit rate of MPEG cell is getting large. This means that the Internet cell affects the jitter value more largely than the MPEG cell. We can observe the same tendency in Figures 2.7, 2.10 and 2.13.

Figures 2.15 and 2.16 represent the jitter values under N, the number of ATM switches, equal to 4 and 5, respectively. In both figures, the MPEG bit rate is equal to 8Mbps and the bit rate of Internet cell equal to 13 Mbps. From the figures, we can see the same tendencies of previous figures where the stream of Internet cell exists. In particular, jitter values at the decoder of Kobe become

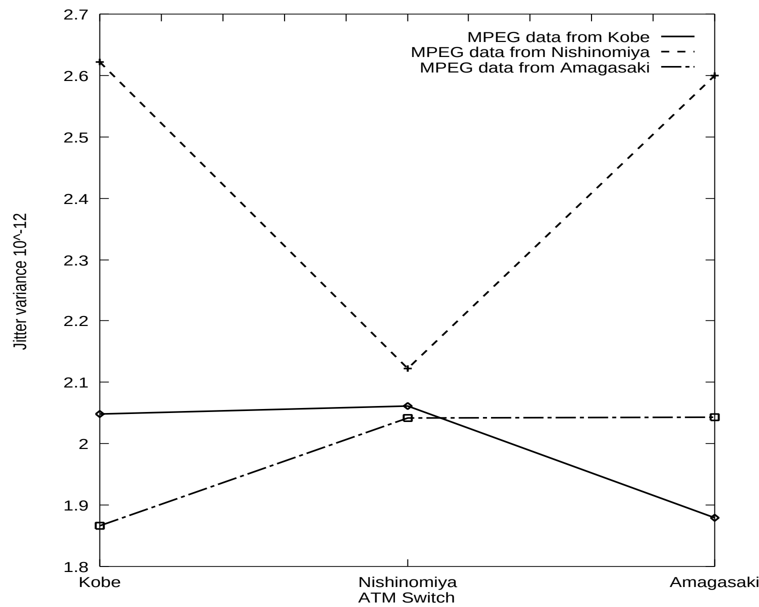


Figure 2.9. Jitter Values: MPEG = 8Mbps, IP = 0bps

quite large when $N = 5$. This suggests that the Internet servers or gateways must be distributed in certain areas if the small jitter values at all decoder points is desired. Otherwise, we have to develop a certain scheme, such as the new protocol for MPEG2 transmission or the new function of the ATM switch, for making the jitter small.

2.5. Conclusion

In this chapter, we have presented the wide area project in Hyogo Prefecture and showed the outline of the testbed network constructed in Kobe, Nishinomiya and Amagasaki. We also investigated the jitter processes with the simulation study and presented the effects of the bit rates of Internet and MPEG cells and the number of ATM switches.

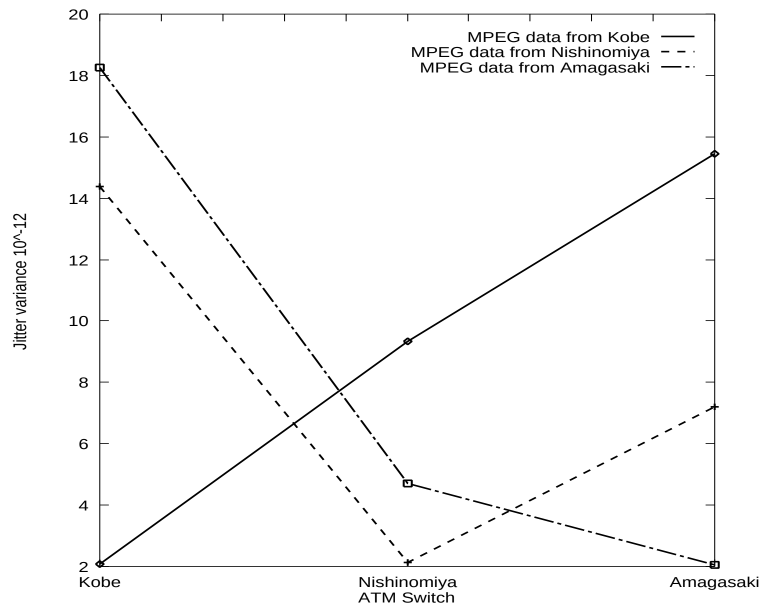


Figure 2.10. Jitter Values: MPEG = 8Mbps, IP = 13Mbps

It is an important issue to evaluate the quality of decoded pictures under our simulation results. [27] reported that the tolerated value of cell delay variation (CDV) for 1.5-Mbps MPEG NTSC video is 6.5ms while that of 20Mbps HDTV video is 1ms. But the tolerated value of MPEG2 is not clear. The worst deviation observed in Figure 2.16 equals about $\sqrt{1.8 \times 10^{-10}} \simeq 13.4\mu$s and it looks to satisfy the QoS for MPEG2. However more researches on the QoS requirement for MPEG2 in terms of the jitter value are needed. Here we have following issues related to the QoS for our future research:

- Investigate the QoS requirement of MPEG2 video in terms of the jitter values.

- How does the number of ATM switches affect the jitter of MPEG cells?

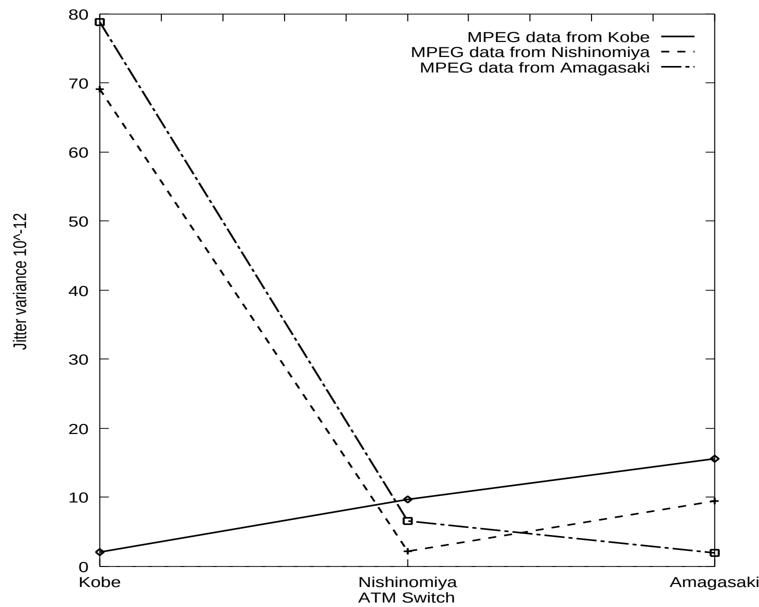


Figure 2.11. Jitter Values: MPEG = 8Mbps, IP = 26Mbps

- What size of the buffer of MPEG2 decoder is needed for absorbing the jitter effects? This issue is also related to the cost problem.

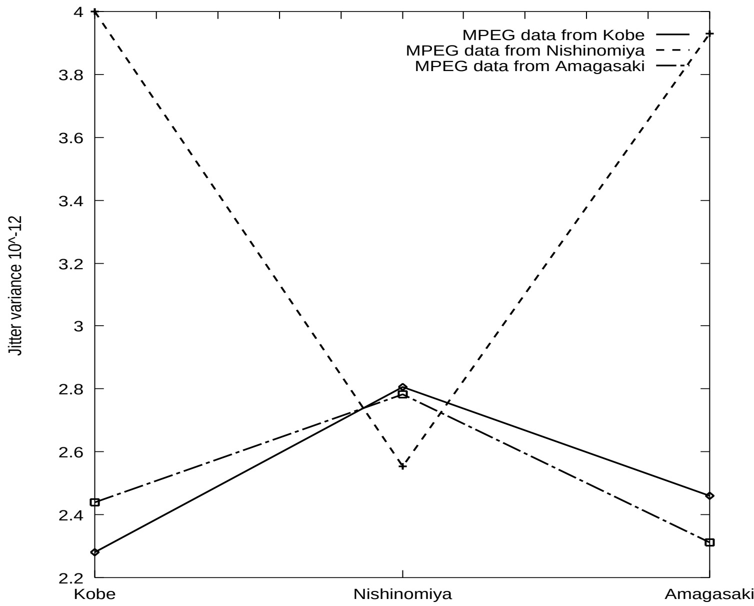


Figure 2.12. Jitter Values: MPEG = 13Mbps, IP = 0bps

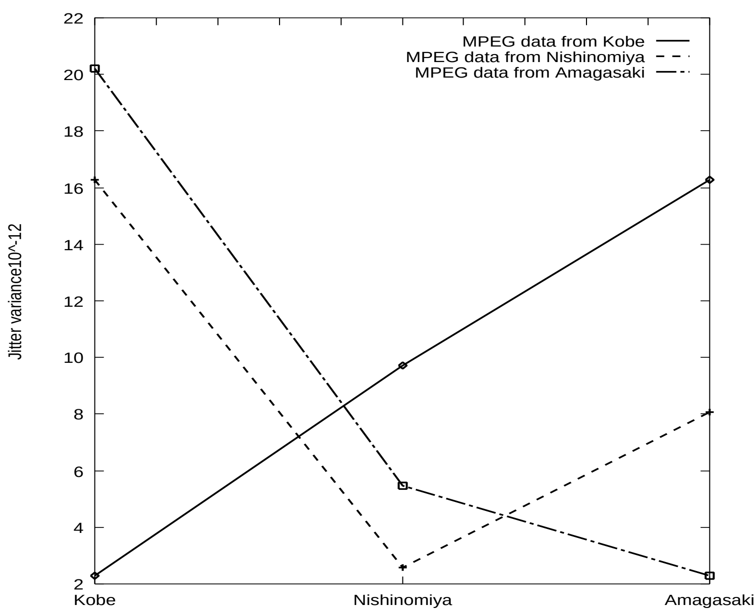


Figure 2.13. Jitter Values: MPEG = 13Mbps, IP = 13Mbps

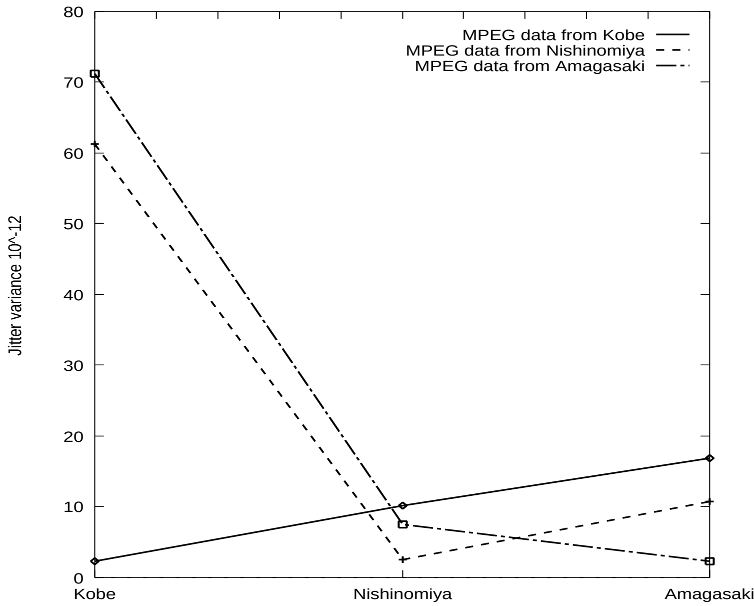


Figure 2.14. Jitter Values: MPEG = 13Mbps, IP = 26Mbps

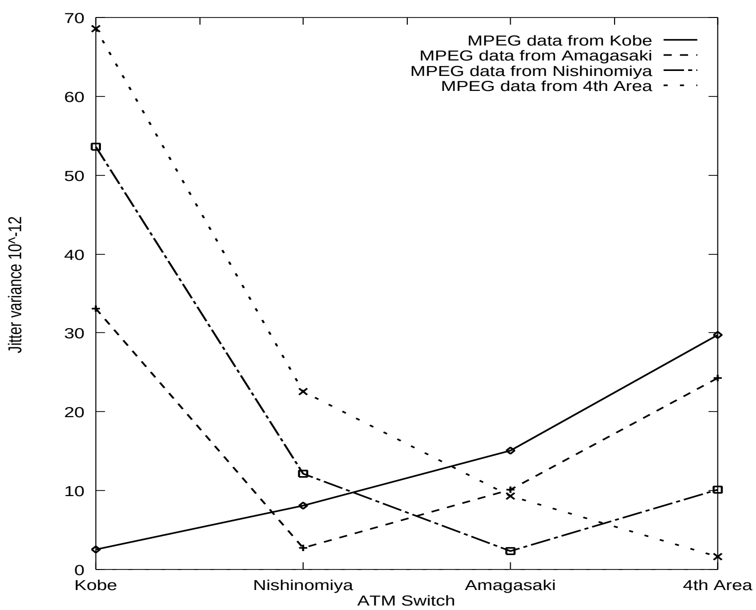


Figure 2.15. Jitter Values: MPEG = 8Mbps, IP = 13Mbps, 4 Areas

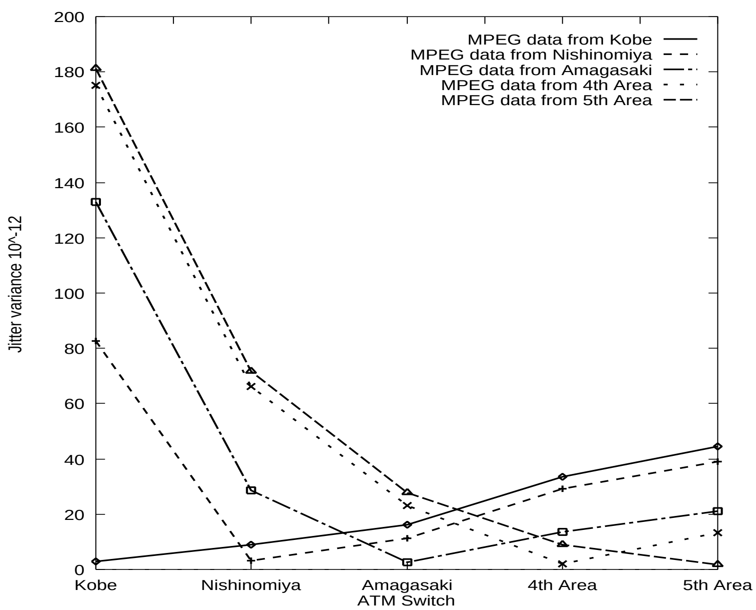


Figure 2.16. Jitter Values: MPEG = 8Mbps, IP = 13Mbps, 5 Areas

Chapter 3

Jitter and Self-similarity with LRD

3.1. Introduction

The statistical characteristics of self-similarity have been shown to apply to ATM network, differing from usual telephone network and packet exchange network [16]. In [16], the authors investigate the characteristics of ATM cell level aggregated traffic from the point of self-similarity. They also analyze the separated data in the aggregated traffic. Hence our testbed network is also expected to have self-similarity on its traffic. In general, the traffic with LRD degrades the network performance and hence it is important to capture the traffic characteristics in terms of LRD for investigating the performance measures such as cell loss probability, transmission delay, and jitter process. So, in this chapter, in order to construct the simulation model of this network for investigating the correlation between Internet and MPEG cells, we investigate the characteristics of cell

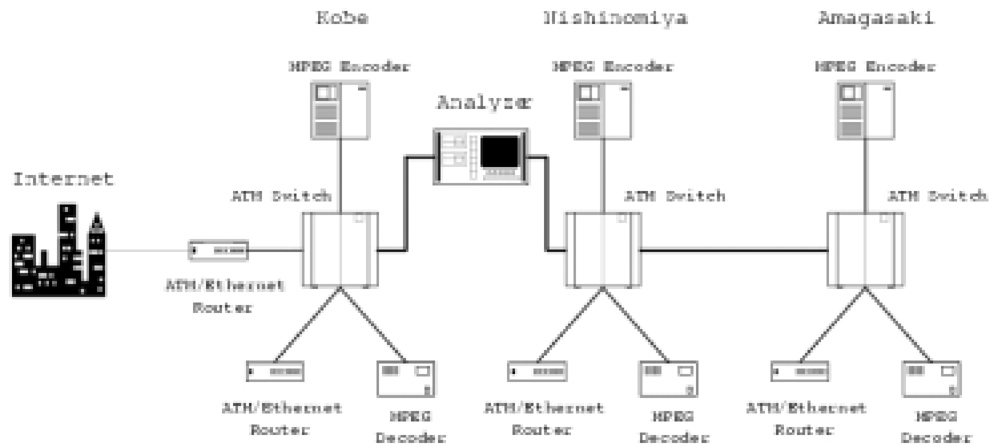


Figure 3.1. Measuring Environment.

stream transmitted on the backbone network and estimate its Hurst parameter. In addition we modify the simulation model of testbed network which we used in Chapter 2 focusing on generation of self-similar traffic, and investigate the relation between the self-similar traffic and jitter of MPEG cells.

This chapter is organized as follows. In Section 3.2, we outline the measuring environment of our testbed network and explain the measured data sets. In Section 3.3, we summarize our analysis methods of measured data for self-similarity. In Section 3.4, we estimate the Hurst parameter from the data collected by KOMIC. In Section 3.5, we present the modified simulation model and show some numerical examples.

3.2. Traffic Measurements

In this section, we analyze several kinds of data set collected by KOMIC on January 8, 1999 and February 3, 1999. These data sets are collected from the

Table 3.1. Qualitative Feature of Traffic Measurements for High Bitrate Data.

Measurement Period: January 8, 1999				
Data Set	Measurement Span (s)	Direction	# of MPEG Sources	Bitrate (Mbps)
R-1_1	6.0910829	Right	1	9.124
R-1_2	5.5301356	Right	1	10.049
R-1_3	6.3653521	Right	1	8.731
L-1_1	6.3312494	Left	1	8.778
L-1_2	6.4701180	Left	1	8.589
L-1_3	6.7403174	Left	1	8.245
L-2_1	3.6361174	Left	2	15.284
L-2_2	3.5231654	Left	2	15.774
L-2_3	3.4062480	Left	2	16.315

Table 3.2. Detailed Description of Internet Cells for High Bitrate Data.

Data Set	Total of Internet Cells	Ratio of Internet Cells (%)	Mean Bitrate of Internet Cells (Mbps)
R-1_1	33804	25.8	2.353
R-1_2	42785	32.6	3.280
R-1_3	29455	22.5	1.962
L-1_1	16312	12.4	1.092
L-1_2	13562	10.3	0.888
L-1_3	8856	6.7	0.557
L-2_1	9770	7.5	1.139
L-2_2	9318	7.1	1.121
L-2_1	13716	10.5	1.707

multiplexed traffic on the fiber trunk lines of the ATM backbone (see Figure 3.1).

In this measuring environment, we have two directions. We call cell stream from Kobe to Nishinomiya "Right" and that from Nishinomiya to Kobe "Left". We have the data of particular case in which Nishinomiya's Encoder didn't work. We call the data one MPEG source data in contrast with the other data under which two MPEG sources (Nishinomiya and Amagasaki) work. These data sets include the time-stamps accurate to within $10^{-1}\mu s$.

An ATM analyzer (HP E4210B) is used for measuring the cell stream and equipped in Kobe CATV office. The analyzer records the time-stamps of cells coming in and going out. The analyzer's capacity being able to measure continuously is 7.34MB. Hence, a measurement span of ATM traffic is restricted within a few seconds. The total number of cells are 131,072 cells and total number of bytes are 6,946,816 bytes for each data series.

In Tables 3.1 to 3.4, we summarize the qualitative feature of measured data. Table 3.1 shows the description of data collected on January 8, 1999. We present the feature of the Internet cells contained in the measured traffic in Table 3.2. We also show the corresponding descriptions on February 3, 1999 in Tables 3.3 and 3.4. It is observed that we have two types of data in terms of Internet cells. We call data of Table 3.1 "high bitrate data" and that of Table 3.3 "low bitrate data".

The naming rule of measured data set is as follows. The name of measured data set consists of the three characters: one capital letter and two decimals (e.g. L-1_1). An alphabet means the direction of MPEG stream, first decimal presents the number of active MPEG source in the measured data, and second decimal is

Table 3.3. Qualitative Feature of Traffic Measurements for Low Bitrate Data.

Measurement Period: February 3, 1999				
Data Set	Measurement Span(s)	Direction	# of MPEG Sources	Bitrate (Mbps)
R-1_4	8.1896399	Right	1	6.784
R-1_5	8.1560761	Right	1	6.816
R-1_6	8.1951826	Right	1	6.776
R-1_7	8.1204604	Right	1	6.840
R-1_8	8.0994105	Right	1	6.864
L-1_4	8.2097140	Left	1	6.768
L-1_5	8.2052066	Left	1	6.776
L-1_6	8.2072090	Left	1	6.768
L-1_7	8.2011678	Left	1	6.776
L-1_8	8.2092061	Left	1	6.768
L-2_4	4.1038306	Left	2	13.544
L-2_5	4.1058530	Left	2	13.528
L-2_6	4.1042450	Left	2	13.536
L-2_7	4.1043757	Left	2	13.536
L-2_8	4.1023647	Left	2	13.544

Table 3.4. Detail Description of Internet Cells for Low Bitrate Data.

Data Set	Total of Internet Cells	Ratio of Internet Cells (10^{-1}%)	Mean Bitrate of Internet Cells (kbps)
R-1_4	65	0.496	0.421
R-1_5	109	0.832	0.708
R-1_6	53	0.404	0.342
R-1_7	363	2.769	2.368
R-1_8	902	6.882	5.902
L-1_4	219	1.670	1.413
L-1_5	56	0.427	0.361
L-1_6	115	0.877	0.742
L-1_7	48	0.366	0.310
L-1_8	110	0.839	0.709
L-2_4	95	0.725	1.227
L-2_5	42	0.320	0.541
L-2_6	29	0.221	0.374
L-2_7	86	0.656	1.110
L-2_8	68	0.519	0.878

the data set number.

3.3. Estimating Methods for Self-Similarity

We summarize our estimating methods. For details, readers are referred to [4, 19, 33, 34].

3.3.1 Aggregated Variance Method

Given a stationary time series $X = \{X_t : t = 1, 2, \cdots, N\}$, we define the aggregated series $\{X_k^{(m)} : k \geq 1, m = 1, 2, \ldots\}$ by summing the original series X over non-overlapping blocks of size m. That is,

$$X_k^{(m)} = \frac{1}{m}\sum_{i=1}^{m} X_{m(k-1)+i},\ \ k \geq 1. \tag{3.1}$$

for successive values of m. The index k, is a label of kth block. If the time series has the self-similarity, $\mathrm{Var}X^{(m)} \sim \sigma^2 m^\beta$ as $m \to \infty$ where $\beta = 2H - 2 < 0$.

For a given m, dividing the data, $\{X_t : t = 1, 2, \cdots, N\}$, into N/m blocks of size m, its sample variance is given by,

$$\begin{aligned}\widehat{\mathrm{Var}}X^{(m)} &= \frac{1}{N/m}\sum_{k=1}^{N/m}(X_k^{(m)})^2 \\ &\quad -(\frac{1}{N/m}\sum_{k=1}^{N/m}(X_k^{(m)}))^2.\end{aligned} \tag{3.2}$$

We compute this quantity for different values of m repeatedly and plot the sample variance versus m on a log-log plot. Since $\widehat{\mathrm{Var}}X^{(m)}$ is an estimate of $\mathrm{Var}X^{(m)}$, the resulting points will form a straight line with slope $\beta = 2H - 2$. If X has no long-range dependence, the slope should equal -1.

3.3.2 Absolute Value Method

We consider the aggregated series (3.1). We take the first absolute moment of this series,

$$AM^{(m)} = \frac{1}{N/m}\sum_{k=1}^{N/m}|X_k^{(m)} - EX^{(m)}|, \tag{3.3}$$

where $X_k^{(m)}$ is defined in (3.1) and $EX^{(m)}$ is the overall series mean. $AM^{(m)}$ behaves like m^{H-1} for large m. For successive values of m, the sample absolute first moment of the aggregated series is plotted versus m on a log-log plot. The result should be a straight line with a slope of $H-1$. If the series has no long-range dependence and finite variance, then $H = 0.5$ and the slope of the fitted line should be $-1/2$.

3.3.3 R/S Method

For a time series $X = \{X_i : i \geq 1\}$, $Y_j = \sum_{i=1}^{j} X_i$ is a partial sum. We define $R(k, m)$ as follows.

$$\begin{aligned} R(k,m) &= \max_{0<i<m}[Y_{k+i} - Y_k - \frac{i}{m}(Y_{k+m} - Y_k)] \\ &- \min_{0<i<m}[Y_{k+i} - Y_k - \frac{i}{m}(Y_{k+m} - Y_k)]. \end{aligned} \tag{3.4}$$

$R(k, m)$ is called the adjusted range. In order to study the properties that are independent of the scale, $R(k, m)$ is standardized by

$$S(k,m) = \sqrt{\frac{1}{m}\sum_{i=k+1}^{k+m}(X_i - \bar{X}_{k,m})^2}\ , \tag{3.5}$$

where $\bar{X}_{k,m} = m^{-1}\sum_{i=k+1}^{k+m} X_i$. The ratio is called the rescaled adjusted range or R/S-statistic.

$$\begin{aligned} \frac{R(k,m)}{S(k,m)} &= \frac{1}{S(k,m)}\Big(\max_{0<i<m}[Y_{k+i} - Y_k \\ &\quad -\frac{i}{m}(Y_{k+m} - Y_k)] \end{aligned}$$

$$- \min_{0<i<m} [Y_{k+i} - Y_k$$
$$- \frac{i}{m}(Y_{k+m} - Y_k)]\Big). \quad (3.6)$$

For a large values of m, we have the properties as follows from empirical finding

$$\log E[R/S] \quad \approx \quad a + \log m^H, \quad (3.7)$$

where a is constant. R/S-statistic is plotted versus m on a log-log plot. The result will be a straight line with a slope of H.

For a time series of length N, subdivide the series into K, each size of N/K. Then, for each lag m, compute $R(k_i, m)/S(k_i, m)$, starting at points $k_i = iN/K + 1, i = 1, 2, \cdots$, such that $k_i + m \leq N$.

3.3.4 Periodogram Method

The periodogram is defined as follows

$$I(\lambda) = \frac{1}{2\pi N} |\sum_{j=1}^{N} X_j e^{ij\lambda}|^2, \quad (3.8)$$

where λ, $\{\lambda : 0 \leq \lambda \leq \pi\}$, is the frequency and N is the length of the series. $I(\lambda)$ is an estimator of the spectral density of X_t, $\{X_t : t = 1, \ldots, N\}$, and a series with long-range dependence will have a spectral density proportional to $|\lambda|^{1-2H}$ close to the origin. That is

$$I(\lambda) \quad \approx \quad c_f |\lambda|^{1-2H}, \quad (3.9)$$

where c_f is constant. A log-log plot of periodogram versus the frequency have a straight line with a slope of $1 - 2H$ as $\lambda \to 0$. In practice, only the lowest 10% of the frequencies is usually used for the calculation.

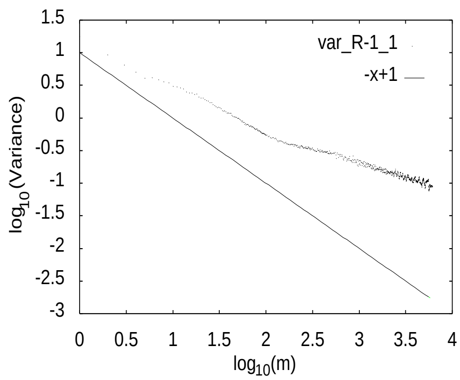
3.4. Estimating Results of Self-Similarity on Testbed Network

We show the results of the analysis for the traffic of the testbed network and estimate the Hurst parameter.

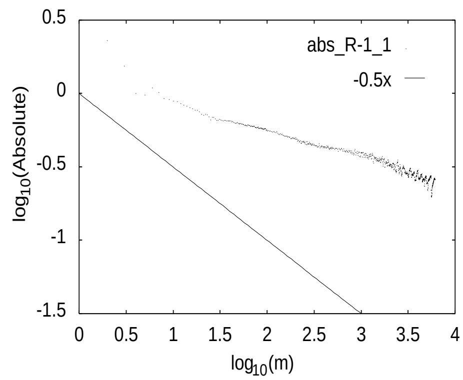
3.4.1 Results of Data Analysis for High Bitrate Data

First, we apply four graphical methods described in the previous section to the high bitrate data (Table 1). In order to evaluate the self-similarity of ATM cell stream, we estimate the Hurst parameter of the measured data. We need to decide the cut-off points for each method. For decision of cut-off points, we assume that measured series X_t is distributed according to Fractional Gaussian Noise (FGN). We generated samples of FGN by Fast Fourier Transform (FFT) with several Hurst parameters (see [28] for details). The maximum time scale of FGN data is equal to measured data. We applied four graphical methods to these data sets, and chose the cut-off points in which slope of the least square line indicates the given Hurst parameter. Readers are referred to [40] for details.

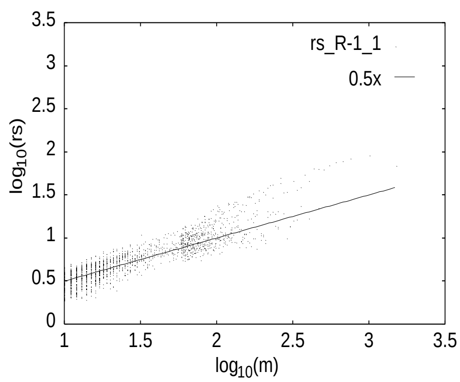
Remark: Most of previous researches have not indicate the criteria for decision of cut-off points. This is still an open problem and hence we make the FGN assumption. The further research is needed for the validity of this assumption.



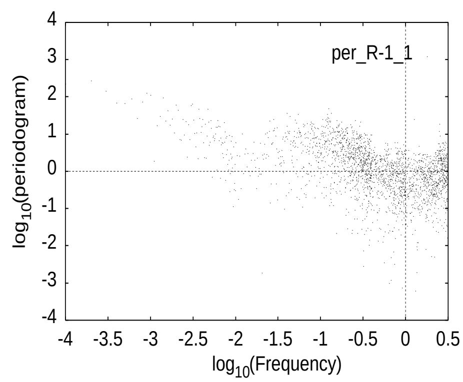
(a) Aggregated Variance



(b) Absolute Value



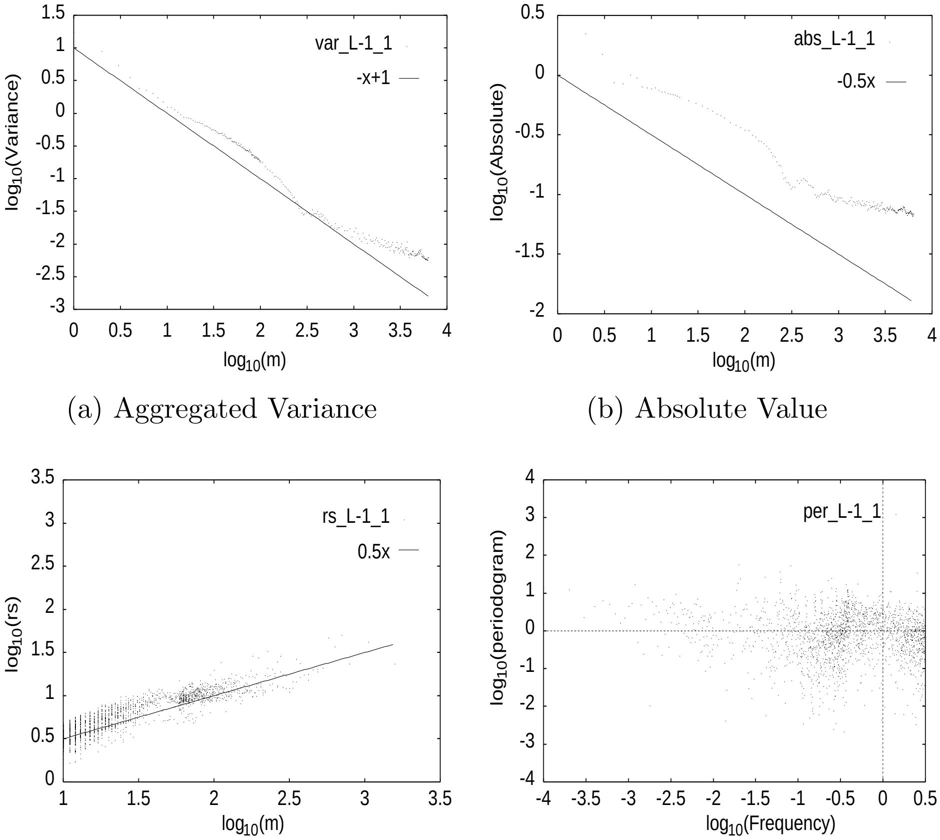
(c) R/S



(d) Periodogram

Figure 3.2. Graphical Results for R-1_1.

We chose $10^{0.3}$ and $10^{2.0}$ as the cut-off points for Aggregated Variance and Absolute Value methods. For the R/S methods, we chose that the cut-offs are $10^{2.5}$ and $10^{3.5}$. Note that the R/S method is very sensitive for deciding the cut-off points in case of the data having the weak long-range dependency [40]. For periodogram method, according to [19], [33], the only lowest 10% of the frequencies are used for calculation of estimating the Hurst parameter. However, our results for periodogram showed that 10% of the frequencies are not always



(a) Aggregated Variance (b) Absolute Value

(c) R/S (d) Periodogram

Figure 3.3. Graphical Results for L-1_1.

sufficient for estimating H, so we estimate Hurst parameter using all plots [40].

We show examples of the results of Aggregated Variance, Absolute Value, R/S and Periodogram Methods in Figure 3.2 for Right traffic (R-1_1), and in Figure 3.3 for Left traffic (L-1_1). For the results of other data sets, readers are referred to [40]. In Figures 3.2 and 3.3, the solid lines present $H = 0.5$.

Figure 3.4 is the result of estimating Hurst parameters applying four methods. From this figure, we can see that each Right data has larger value than each Left

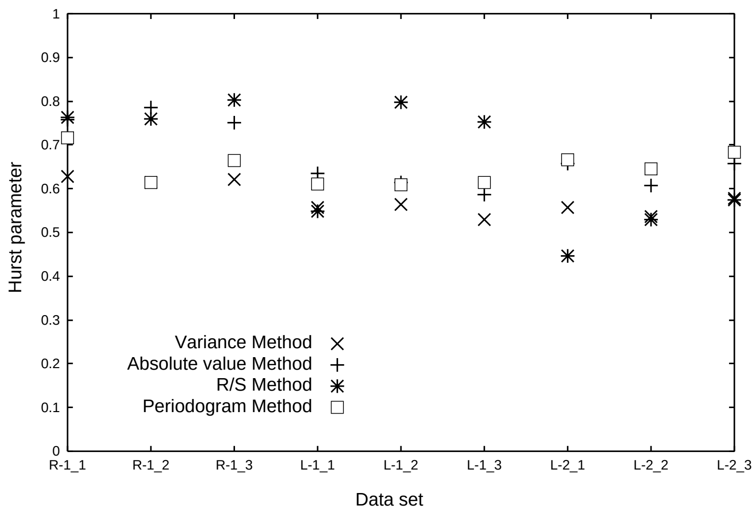


Figure 3.4. The Result of Estimating Hurst Parameter.

data except R/S method. Seeing the Table 3.2, the mean bitrate of Internet cell of each Right data is larger than that of Left data. This implies that when the Internet traffic on the ATM network increases, the long-range dependency on the ATM cell streams become strong.

As for R/S method we can't observe the tendency that the Internet traffic increase the long-range dependency of whole stream. One of the reasons is that R/S Method is very sensitive to the presence of short range dependency [39] in the whole stream. As a result, it is difficult to decide the cut-off points in case of the data contained weak long-range dependent traffic.

It is difficult to determine the value of Hurst parameter of this ATM cell streams because the time scale of measuring is not long enough. However, estimated Hurst parameters in Figure 3.4 are indicating that the ATM cell streams in this project exhibits self-similar nature in a certain level. If we assume that our testbed network has self-similarity, then its Hurst parameter would be around

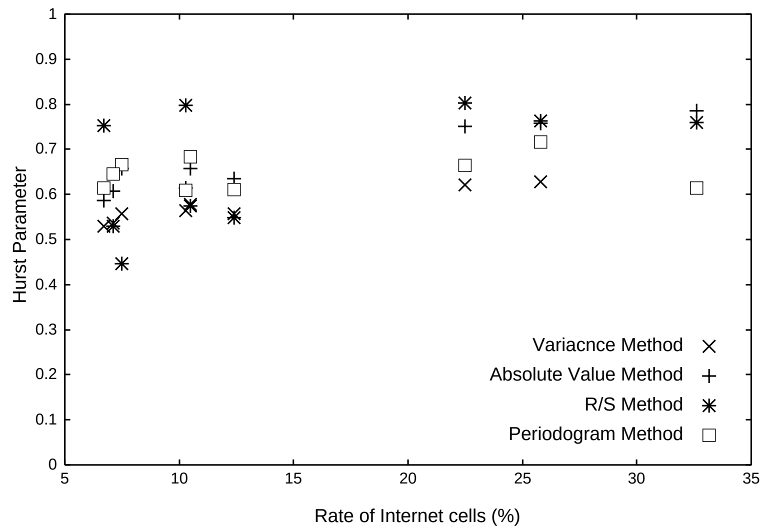
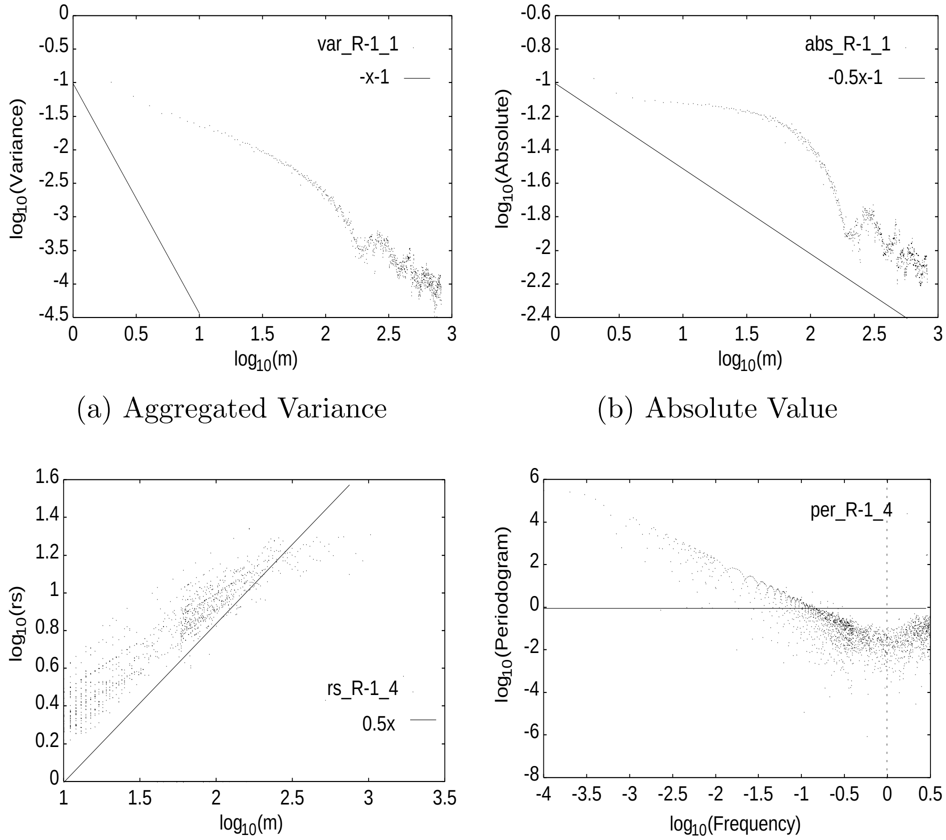


Figure 3.5. Estimation Results for High Bitrate Data.

0.6.

Figure 3.5 depicts the relation of the rate of Internet cell and Hurst parameter without taking the direction into account. From Figure 3.5, it is observed that as the rate of Internet cells becomes large, the Hurst parameter tends to increase except R/S method. From these results, we conclude that the Hurst parameter is about 0.6 when Internet cells account for nearly 10% of all cells transmitted on the backbone line, and that the Hurst parameter becomes larger than 0.6 as the Internet cells increase. Therefore we can assume that when the rate of Internet cell is 20 $\sim$ 30%, from Figure 3.5, the Hurst parameter is more than 0.6, on the other hand, when the rate of Internet cell is nearly 10%, the Hurst parameter is about 0.6.



(a) Aggregated Variance (b) Absolute Value

(c) R/S (d) Periodogram

Figure 3.6. Graphical Results for Low Bitrate Data.

3.4.2 Results of Data Analysis for Low Bitrate Data

In this subsection, we analyze the low bitrate data (Table 3.3) using four graphical methods. We illustrate the examples of analysis results of four graphical methods in Figure 3.6. As for the cut-off points, we use the same values in the high bitrate data case.

In Figure 3.7, we show the relation between average bitrate of Internet cells and Hurst parameter without taking the direction into account. From this figure

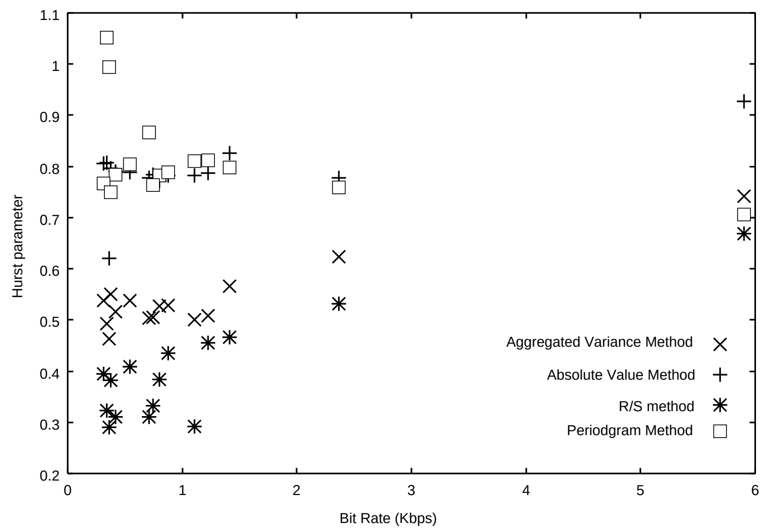


Figure 3.7. Estimation Results for Low Bitrate Data.

we observe that the the self-similarity with $H > 0.7$ appears in the low bitrate data with Absolute Value and Periodogram methods. In our experimental network, MPEG cells are transmitted with PVC-CBR. In [16], it was reported that the Hurst parameter of CBR stream is quite large. This means that MPEG stream is a highly correlated time series. Thus the Hurst parameter might be close to 1 [16]. So the estimating results in Figure 3.7 is reasonable.

On the other hand, as for the Variance and R/S methods we find that the estimated value for low bit data is very small. One of the reasons of this phenomena is that the choice of the cut-off points causes the small Hurst parameter. In order to decide the cut-off points for estimating the Hurst parameter, we assume that the measured data is distributed according to FGN. We chose the cut-off points $10^{0.3}$ and $10^{2.0}$ for Variance method and $10^{2.5}$ and $10^{3.5}$ for R/S method from FGN sample sequence. However the estimation with these cut-off points failed to capture the high correlation. When we select $10^{0.3}$ and $10^{1.5}$ for Variance method and $10^{0.5}$ and $10^{3.5}$ for R/S method as the cut-off points (these are chosen heuristically), it is possible to capture the high correlation with Hurst parameter about equal to 0.8 for MPEG stream. From this result, it seems that FGN assumption is not suitable to estimate the Hurst parameter for our measured data of low bitrate case with Variance and R/S methods. It should be careful to choose the cut-off points for Variance and R/S methods.

Another reason under which the Variance method does not work well is the periodicity of the measured data due to the CBR stream. If the time series is periodic, as is the case of the CBR stream, the Variance of its averaged process is quite small. Therefore it seems that the Variance method is not suitable for the estimation of the Hurst parameter if we don't have enough data.

It is difficult to estimate the Hurst parameter since the results of Absolute Value and Periodogram methods are different from those of Variance and R/S methods. However, as we stated there are some difficulties for the estimation of the Hurst parameter with Variance and R/S methods. If we adopt the results of Absolute Value and Periodogram method, the Hurst parameter would be around $0.7 \sim 0.8$.

3.5. LRD and QoS for Multimedia

From estimation results in the previous section, we find that the Hurst parameter of Right stream is around 0.6 to 0.8 while that of Left stream is about 0.6. From these results we observe that the traffic in our testbed network possesses self-similarity. In this section, we investigate the relation between the self-similar traffic and jitter of MPEG cells by simulation.

3.5.1 Module of Internet Server

In [40], we made the module of Internet Server which generates Internet cells according to the Pareto distribution. The Pareto distribution is reported to be the best model of the document size distribution for WWW service [24]. We assumed that the Internet server generates cells according to the Pareto distribution when the request cell arrives at the server. However, we observed very high self-similarity ($H > 0.9$) with this module.

In this study, we modify this module using Fractional Gaussian Noise (FGN). For making FGN data we use the fast Fourier transform method [28]. We set the parameters for generating the FGN data so as to match the average and variance of measured data. We consider the discrete-time stochastic process whose unit length of time is equal to 10^{-5}sec. Let X_n denote FGN random variables at time n. Our improved module of Internet Server generates X_n cells at time n. The weak point of using FGN is that the generated sequence contains negative values. We selected the parameter sets for generating FGN carefully such that the number of negative values are less than 10% of that of the generated sequence. Then we regard the negative value as zero.

3.5.2 Module of ATM Switch

In chapter 2.1, we constructed the ATM switch is implemented the multicast function. In this chapter, we modify the output buffer in the ATM switch and investigate the jitter process under the two cases. The case 1 is that the ATM switch is equipped with two output buffers. MPEG and Internet cells are put into these output buffers separately. In this case the service discipline of two buffers is round-robin. The case 2 is that the ATM switch is equipped with one output buffer. In this case, MPEG cells contend with Internet cells in same buffer section.

3.5.3 Simulation and Result

Using the modified module described in the previous subsection, we finally construct the simulation model of the experimental network same as that in chapter 2 (see Figure 2.5).

In Figure 2.5, jitters 1 to 3 are modules of MPEG2 encoders. Jitter 1 to 3 are corresponding to Kobe, Nishinomiya and Amagasaki, respectively. Decodes 1 to 3 are the modules of MPEG decoder. Sw 1 to 3 are the modules of ATM switch. In the testbed network, the switching capacity is assumed to be at least 5 Gbps. Hence we chose 6Gbps as the value of the switching capacity and set the cell switching time in ATM switch equal to $53 \times 8/(6 \times 10^9) \simeq 7.0 \times 10^{-8}$ (sec). The capacity of all links is equal to 156Mbps.

In Figure 2.5, ip_server is the module of the Internet server and generates self-similar traffic. Ip_clients 1 to 3 are the modules of Internet client. At first, in order to confirm that our modified modules generate the self-similar traffic, we run the simulation with only Internet cell stream from Internet server to client at Amagasaki, and investigate the Hurst parameter H of the stream from Kobe to Nishinomiya. We set the parameters of the stream from Internet server based on the result of [40] as shown in Table3.5.

Figure 3.8 shows the result for Variance method of Internet stream from Internet server. From this figure, we get the Hurst parameter as $H = 0.67$. The

Table 3.5. Parameter Sets of FGN for Module of Internet Server

Parameter	value
Hurst Parameter	0.7
Mean Cells within a Slot(10^{-5}sec)	0.05514
Variance of Cells within a Slot	0.16082
Number of Sample Points	1048576

reason why we get the smaller value of Hurst parameter is that we have regarded the negative value as 0. Although resulting Hurst parameter becomes small, the self-similar traffic is realized by our modified module. In the following, referred values of Hurst parameter are those under which FGN data are generated.

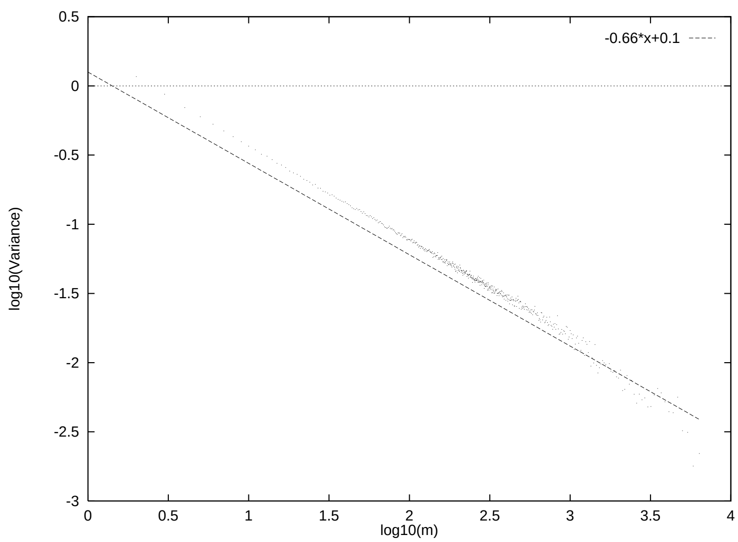


Figure 3.8. Result of Variance Method

Next, we investigate the jitter process at the three decoder points for MPEG cells from Kobe. As for the cell-generating rate of the MPEG2 encoder, we set 6.771 Mbps, that is, we set the interdeparture time to 6×10^{-5}sec. The service type of MPEG2 cells is CBR. In order to study the relation between the self-similarity of Internet and MPEG cells, we run the simulation with parameter sets of Hurst parameter and MPEG bitrate shown in Table 3.6. We make the 23.53Mbps Internet stream by adding ten 2.353Mbps MPEG streams with different seeds of random numbers.

Table 3.6. Parameter Sets

Figures	Hurst Parameter	Internet Bitrate (Mbps)
3.9	0.5, 0.6, 0.7, 0.8	2.353
3.10	0.5, 0.6, 0.7, 0.8	23.53
3.11	0.5, 0.6, 0.7, 0.8	2.353
3.12	0.5, 0.6, 0.7, 0.8	23.53

In Figures 3.9 to 3.12, the horizontal axis represents the location of the CATV company and the vertical axis means the jitter value. Figure 3.9 shows the simulation results in the case that Hurst parameter is equal to 0.5, 0.6, 0.7 and 0.8, respectively, and Internet bitrate equal to 2.353Mbps. We set two buffers for MPEG and Internet cells in ATM switch module. From this figure, we can observe that the jitter value becomes small at Nishinomiya and Amagasaki as the value of Hurst parameter increases. In general, the traffic with large Hurst parameter causes the growth of the buffer contents. When the Internet bitrate is small, the Internet cells with large Hurst parameter tends to make the buffer length for Internet cells large. Since the service discipline of output buffers is round-robin, large buffer length of Internet cells tends to smooth the output process of MPEG cells. We also observe that jitter value is largest at the decoder point of Nishinomiya. This is because there are no bursty inputs of Nishinomiya ATM switch whose destination is Amagasaki. The MPEG2 stream from Kobe is regulated by the output buffers of Kobe and Nishinomiya and hence resulting jitter value of Amagasaki is smaller than that of Nishinomiya.

In Figure 3.10, we show the simulation result when Hurst parameter is equal to 0.5, 0.6, 0.7 and 0.8 under Internet bitrate equal to 23.53Mbps. We set two buffers for each QoS in ATM switch module. From this figure, we can see the same tendency for the fluctuation of jitter values in the previous case. However the jitter value becomes large when the Hurst parameter is getting large. Note that the range of jitter values is 1.09 to 1.19, which is smaller than Figure 3.9. That is, the jitter values are almost same under various values of Hurst parameter. From this result, we can see that the jitter value is not affected by Hurst parameter when the bitrate of Internet is large.

Next we investigate one buffer case within the ATM switch module. In one buffer case, MPEG cells contend with Internet ones for the occupation of the buffer. On the other hand, a buffer is allocated for each type of cells in the case of two buffers. We show the results under Internet bitrate equal to 2.353Mbps in Figure 3.11 and those of 23.53Mbps in Figure 3.12. In both figures, we can see the same tendencies of Figure 3.9 and 3.10. However the jitter values of 3.11 (resp. 3.12) are larger than those of 3.9 (resp. 3.10). This is due to one output buffer within ATM switch.

3.6. Conclusion

In this chapter, we investigated the characteristics of cell streams in the ATM network integrating CATV and Internet. Using the four graphical methods, we estimated the Hurst parameter of our testbed network. We observed that the Hurst parameter is around 0.6 provided that the traffic has self-similarity. In

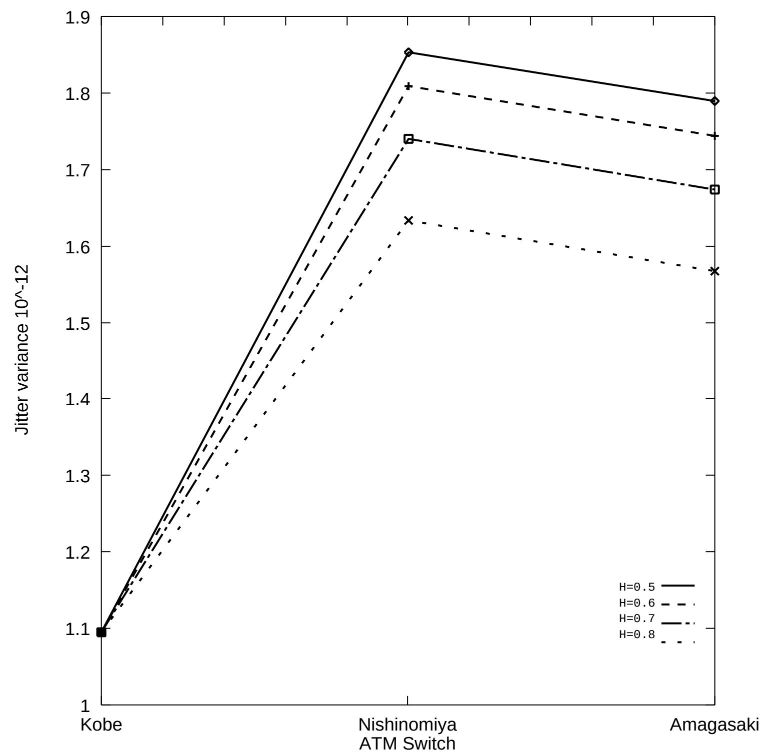


Figure 3.9. Jitter Values: Internet Stream Bitrate= 2.353Mbps, Two Buffer Case

addition, we introduced the FGN assumption to decide the cut-off points for each method.

We also investigated the effect of the amount of the Internet cells on the Hurst parameter of whole stream. We conclude that the Hurst parameter is about 0.6 when Internet cells account for nearly 10% of all cells transmitted on the backbone line, and that the Hurst parameter becomes larger as the Internet cells increase.

We also investigated the Hurst parameter of the streams which include a little amount of the Internet traffic. We observed that the deciding the cut-off points chosen from FGN assumption for Variance and R/S methods are not suitable to capture the long-range dependency of low bitrate data. Though MPEG stream

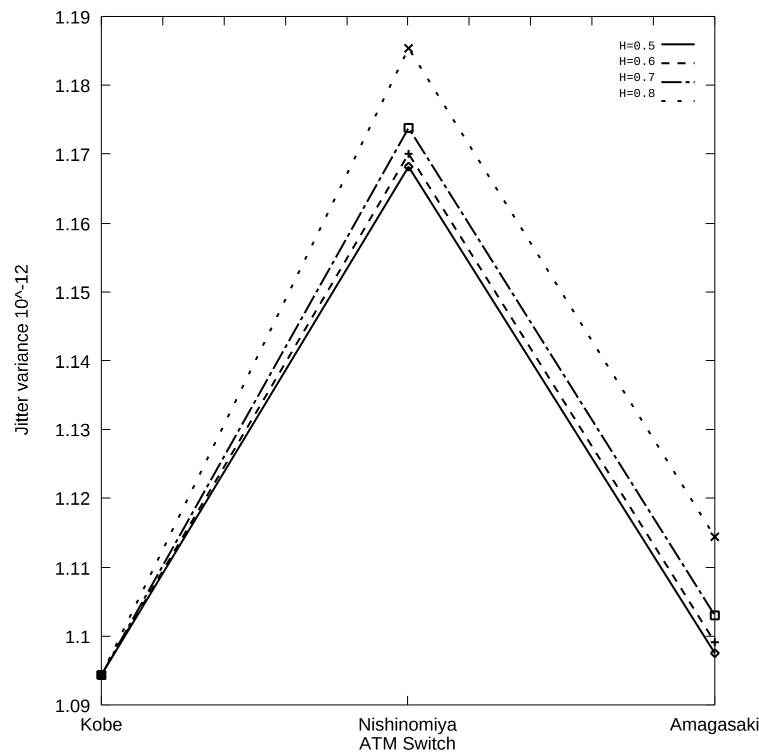


Figure 3.10. Jitter Values: Internet Stream Bitrate= 23.53Mbps, Two Buffer Case

is highly correlated, the periodicity of MPEG stream does not matter in its QoS control. However Internet stream will cause the degradation of QoS for MPEG2 since Internet stream exhibits LRD when Internet cells increase.

In addition, we reconstructed the Internet cell generator in our simulation model and investigated the jitter behavior of MPEG2 at the decoding points of the testbed network. From the simulation results, we observed that the jitter process is affected by the Hurst parameter when the Internet bitrate is small, while not affected when the Internet bitrate is large.

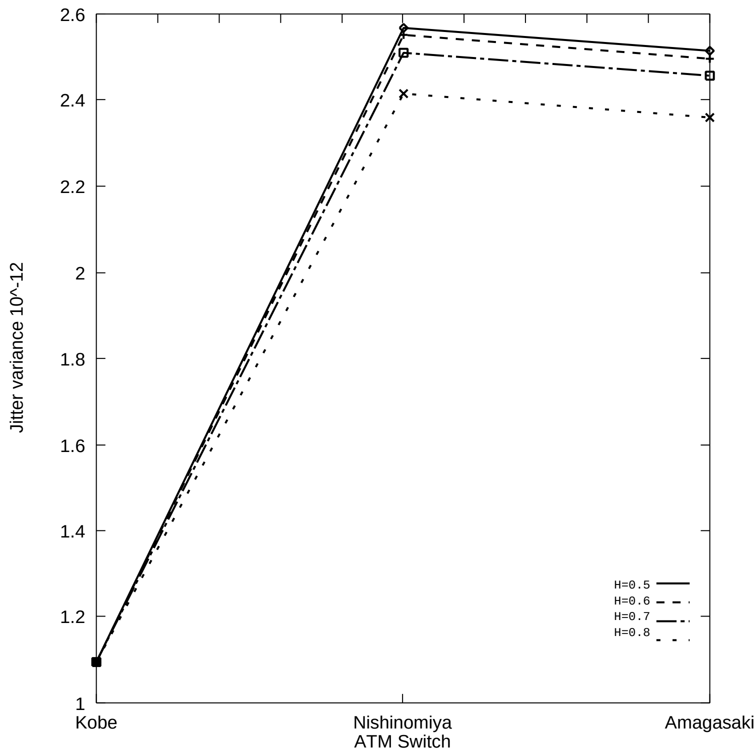


Figure 3.11. Jitter Values: Internet Stream Bitrate= 2.353Mbps, One Buffer Case

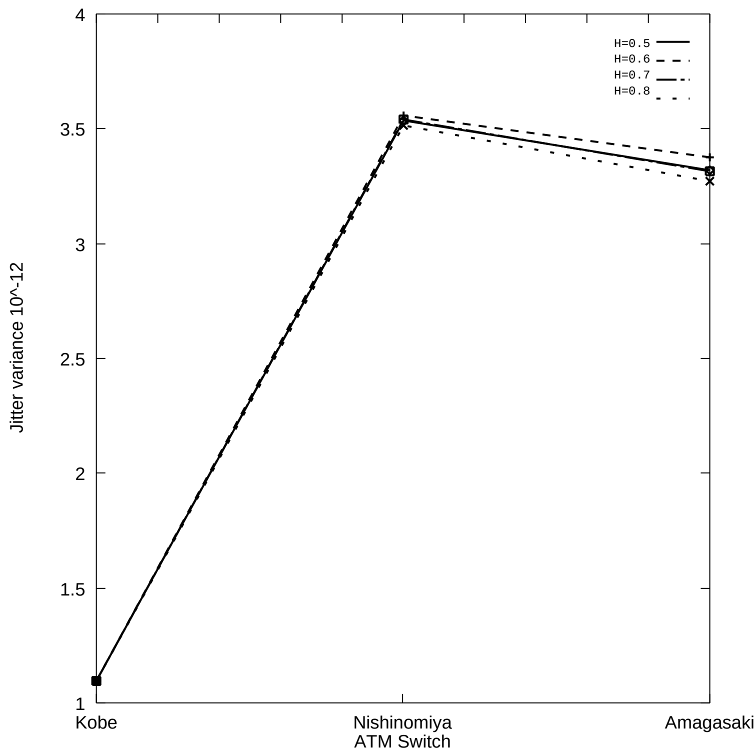


Figure 3.12. Jitter Values: Internet Stream Bitrate= 23.53Mbps, One Buffer Case

Chapter 4

Jitter Reduction Scheme

4.1. Introduction

In this chapter, we consider how the jitter affects the QoS of multimedia application and propose the jitter reduction scheme for multimedia communication.

In general, the jitter is classified into cell-level jitter and application-level one. However, the cell-level jitter is not important for the QoS at application level. The application data packet is segmented into the ATM cells at the ATM adaptation layer (AAL) of the source node and those cells are sent to the network link. The cells arriving at the destination node are assembled into the application data at the AAL layer. The QoS at the application level is largely affected by the arriving points of cells corresponding to the end of the data packet. That is, the time to finish assembling the data packet from ATM cells is crucial for the jitter at the application level. This finishing time is determined by the arrival time of the last segmented cell of the data packet. Throughout the chapter, the cells corresponding to the end part of the original data packet are called critical cells, and the application-level jitter is defined as the variance of the inter-arrival time of critical cells at destination node. For the QoS guarantee at the application level, it is crucial to keep the application-level jitter as small as possible. Here we define the jitter at the application level as the variance of inter-arrival time of critical cells at destination node.

The Available Bit Rate (ABR) service class in ATM is designed mainly for data transmission[18]. The ABR guarantees only the cell loss ratio (CLR) and does not provide any other QoS guarantees such as cell transmission delay (CTD) and cell delay variation (CDV). Therefore the ABR service class is not sufficient for the multimedia communication such as the real-time video.

Recently, [37, 38] proposed the design method with the queue control function which is used for calculating the bandwidth allocated to the source node. Using the queue control function, it is possible to control the transmission delay and to achieve the low CLR by adjusting the allowed cell rate (ACR) according to the queue length of the bottleneck switches along the path. The queue control function algorithm is quite attractive since the ABR service category can support the multimedia communication with small delay. However the CDV, or equivalently, the jitter is not taken into consideration in their algorithm. The jitter is also important for the real-time video transmission where the jitter affects the quality of the decoded video at the destination node.

So we propose the scheduling scheme at the source node to reduce the jitter at application level under the ATM-ABR service class. In our proposed scheme, we focus on the departure points of critical cells. The critical cell is intentionally delayed until the next data packet generation and transmitted at the beginning of the next cycle of packet generation. According to this scheme, the departure points of critical cells at the source node are like the CBR traffic and therefore the reduction of the jitter at application level is expected. Since the points of sending the critical cells are intentionally delayed, we call our proposed scheme intentionally delayed transmission (IDT). The strong point of IDT scheme is that we need not change the existing ATM facilities except the source node.

This chapter is organized as follows. In Section 4.2, we describe our proposed scheme in detail. In Section 4.3, we show the analytical model proposed in [20, 21, 22] and apply it to our proposed scheme. In Section 4.4, we describe the simulation models for our proposed scheme and present the numerical results of the analytical model and simulation.

4.2. Jitter Reduction Scheme on ABR

In this section, we describe the IDT scheme in detail. First we suppose that the application layer generates the data packet and that the interarrival time of consecutive packets is constant equal to T (Figure 4.1). The period T is regarded as a cycle of packet generation. In addition, we assume that the application program generates at least one cell during each period T and that the number of cells generated within the period T is bounded according to the ACR. Though it seems that this bound is not suitable for the model of the multimedia traffic, it is possible to introduce this bound depending on the period T by using the dynamic encoding rate adjusting scheme proposed in [8, 17, 13]. These assumptions are valid for the multimedia applications such as PCM sampling audio and MPEG video.

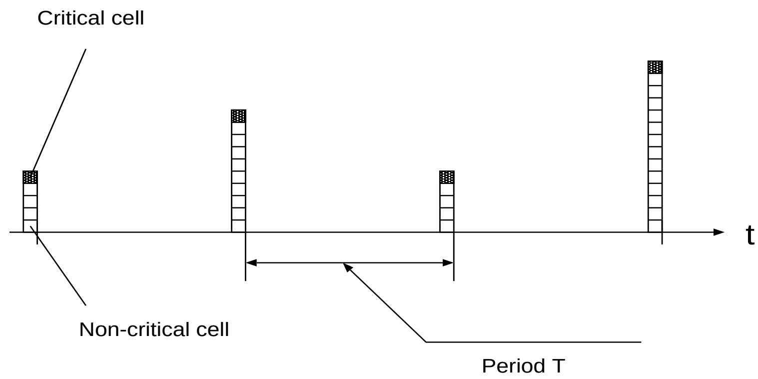


Figure 4.1. Departure Process at Source Node

In the case without IDT algorithm, the source node sends the cells as fast as possible according to the ACR. Since the packet size at the application level is variable, interdeparture time of critical cells varies depending on the packet size.

The strategy of jitter reduction is as follows. The time to complete reassembling the segmented cells into the original packet is determined by the arrival of the critical cell. Therefore we focus our attention on the departure points of critical cells at the source node. In our proposed method, the critical cell is delayed until the next data packet generation and transmitted at the beginning of the next cycle of packet generation. Therefore the interdeparture time of critical cells is constant with period T and it is expected that the resulting interarrival time of critical cells at destination node varies less than that of ordinary ABR service (Figure 4.2). The strong point of IDT scheme is that we need not change the existing ATM facilities except the source node.

The procedure of IDT scheme is as follows.

1. The application data is segmented into ATM cells at AAL.

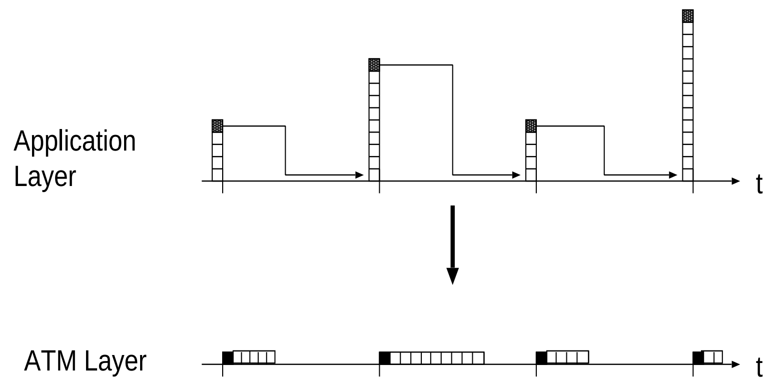


Figure 4.2. IDT Scheme

2. AAL tags the critical cells.

3. All ATM cells are sent to the ATM Switch Layer through the ATM Layer.

4. In the ATM switch layer;

   (a) Non-critical cells are transmitted to the network according to the ACR.

   (b) The critical cell is not sent until next data packet generation. This critical cell is sent into the network when the cells generated at the next packet generation enter the ATM switch layer.

As for the implementation of IDT scheme on ATM protocol stack, we have to modify the two layers, AAL and ATM switch layer. The AAL tags the critical cells and notifies to the ATM switch layer which cell is critical. This implementation needs the layer violation between AAL and ATM switch layer. Though this layer violation is a weak point of IDT scheme, this modification is required only at the source node. It seems to be inevitable for supporting the QoS of real-time video over ATM-ABR service class.

4.3. Jitter Process of Critical Cells

As for the previous researches of the jitter behavior in ATM networks, [20, 21, 22] considered the cell-level jitter. In [20, 22], the authors considered the two types of traffic, tagged stream and the background one. They analyzed the jitter process of the tagged renewal stream in the case of single node. [21] analyzed the jitter process in the multiple node case using the results of [20, 22]. We apply the jitter model in [20, 21, 22] to our case, and analyze the jitter process of critical cells. We also verify the effectiveness of our proposed method by simulation.

Here, we summarize the analysis of jitter process studied in [20, 21, 22], and apply their results to our proposed scheme.

We consider a discrete-time single-server queueing system with infinite buffer (Figure 4.3). The time axis is segmented into a sequence of slots and one cell transmission time is equal to a slot. The cells are served according to the FIFO discipline. Here we are interested in the jitter of the critical cells at the destination node. At the source node, data packets are generated periodically with cycle T. Data packets are segmented into ATM cells which consist of a critical cell and non-critical cells. Let U denote the number of non-critical cells generated within a cycle. We assume that the maximum number of cells generated from one packet is equal to T. Hence $0 \leq U \leq T-1$. Let $u(k)$ $(u \leq k \leq T-1)$ denote the probability distribution function (pdf) of U and $U(z)$ the probability generating function (pgf) of $u(k)$.

Now consider nth transmission cycle of the data packet. As described in the previous section, there are T slots between the consecutive packet generation points. The first slot of n-th transmission cycle is used for the critical cell which

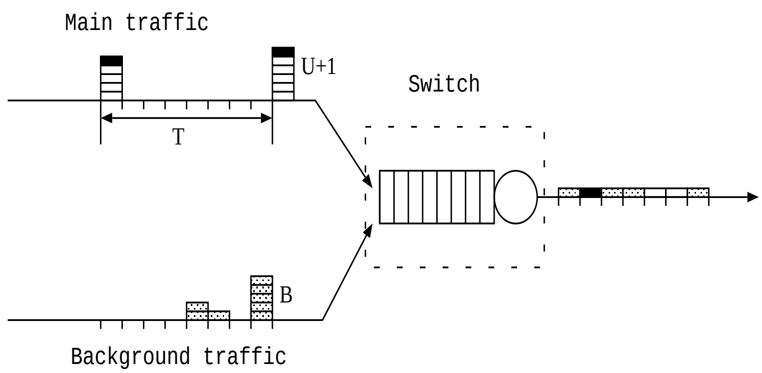


Figure 4.3. Queueing Model

is the last cell of $n-1$st data packet. Then $T-1$ slots are used for non-critical cells where consecutive departure points are conformed with the allocated ACR. However we assume for the analytical simplicity that a non-critical cell is transmitted within a slot with probability p. Let $V(z)$ denote the pgf of the number of non-critical cells transmitted within a slot. Then we obtain

$$V(z) = 1 - p + pz. \tag{4.1}$$

Therefore $U(z)$ becomes

$$U(z) = V(z)^{T-1} = (1 - p + pz)^{T-1}. \tag{4.2}$$

In this chapter, we consider the single node case where there is a switch node between source and destination. At the ATM switch, the tagged packet stream is multiplexed with a background traffic. Let B denote the batch size of the background traffic within a slot. We assume that B is independent and identically distributed (i.i.d.) and that its pdf and pgf are $b(k)$ $(k \geq 0)$ and $B(z)$, respectively.

[20, 21] analyzed the jitter process in the case that the queue accepts two classes of cells, the GI class and the B class. GI represents the tagged cell stream of interest while B stands for the background traffic. The GI class cells arrive at the queue with an interarrival time I which is distributed according to a renewal process. It is assumed that the interarrival time I has a finite upper bound G_{max}. We denote pgf of the integer-valued random variable I by $G(z)$.

Let Q and $Q(z)$ denote the queue length at the arrival points of GI class cells and the pgf of Q, respectively. From [20], we obtain

$$Q(z) = G^{'}(1)(1-\rho_t)\frac{(1-z^{-1})(B(z)/z)^K}{1-zG(B(z)/z)} \times \frac{\Pi_{k=1}^{K}[(z/B(z)) - (r_k/B(r_k))]}{\Pi_{k=1}^{K}[1-(r_k/B(r_k))]}, \quad (4.3)$$

where $K = G_{max} - 2$ and

$$\rho_t = B^{'}(1) + \frac{1}{G^{'}(1)} \stackrel{\text{def}}{=} \rho + \rho_{GI}, \quad (4.4)$$

is the total offered load. It is also shown that if $\rho_t < 1$ the equation

$$1 - zG\left(\frac{B(z)}{z}\right) = 0, \quad (4.5)$$

has K roots inside the unit circle excluding 1. We denote these roots by $r_1, r_2, \cdots, r_K$. We define J as the interdeparture time of two successive GI-cells. We have

$$J \stackrel{\text{dst}}{=} Q_2 - Q_1 + I, \quad (4.6)$$

where Q_1 and Q_2 are the queue sizes seen by two consecutive GI class cells. From

[20],

$$J(z) = \sum_{i=G_{min}}^{G_{max}} g_i J_i(z), \tag{4.7}$$

where

$$J_i(z) = z(B(z))^i + (B(z))^{i-1}(z-1) \times \sum_{k=1}^{i-1} (z^{-1}B(z))^{-k} \Phi(z^{-1}; k), \tag{4.8}$$

and

$$\Phi(z; k) = \sum_{l=0}^{k-1} z^l \pi_k^B(0; l) Pr(Q = l), \ 1 \leq k \leq T-1. \tag{4.9}$$

The $\pi_k^B(0; l)$'s are obtained by the recursive algorithm described in [20] and the $Pr(Q = l)$'s are obtained by inverting (4.3).

In the case of our IDT scheme, the interarrival time of GI class cells which are corresponding to critical cells is constant and equal to T. Hence

$$g_i = \begin{cases} 1, & i = T, \\ 0, & i \neq T. \end{cases} \tag{4.10}$$

The B class cells consists of non-critical cells and background traffic. Therefore the pgf of the number of B class cells within a slot is given by $V(z)B(z)$. Replacing $B(z)$ in (4.8) with $V(z)B(z)$ and using (4.10), the pgf of the interdeparture time of critical cells is given by

$$J(z) = z(V(z)B(z))^T + (V(z)B(z))^{T-1}(z-1) \times \sum_{k=1}^{T-1} (z^{-1}V(z)B(z))^{-k} \Phi(z^{-1}; k). \tag{4.11}$$

In the heavy traffic case, the total utilization ρ_t close to 1. From [20] we obtain that

$$J(z) \to z(V(z)B(z))^T,\ when\ \rho_t \to 1, \tag{4.12}$$

since $\Phi(.;.) \to 0$ as $\rho_t \to 1$.

On the other hand, from [22], as $T \to \infty$ the system behaves like a $Geo^{[X]}/D/1$ queue, that is

$$Q(z) \to (1-\rho_t)\frac{z-1}{z-V(z)B(z)},\ as\ T \to \infty. \tag{4.13}$$

This approximation yields

$$J(z) = E(z^{Q_2-Q_1+I}) = z^T Q(z)Q(z^{-1}). \tag{4.14}$$

We will show the numerical examples calculated from (4.12) and (4.14) and compare the results with simulation.

4.4. Performance Evaluation of IDT Scheme

In this section, we investigate the performance of our proposed scheme by both analytical results and simulation. First we present the simulation models and then show the numerical results in both single and multiple nodes cases.

4.4.1 Simulation Model

In the simulation experiment, we use OPNET version6.0[26]. Figure 4.4 shows the block diagram of the source node in OPNET. We modified the AAL and ATM_switch blocks of the source node for implementing the IDT scheme. When the AAL block receives the data packet from the traf_src block, the AAL block

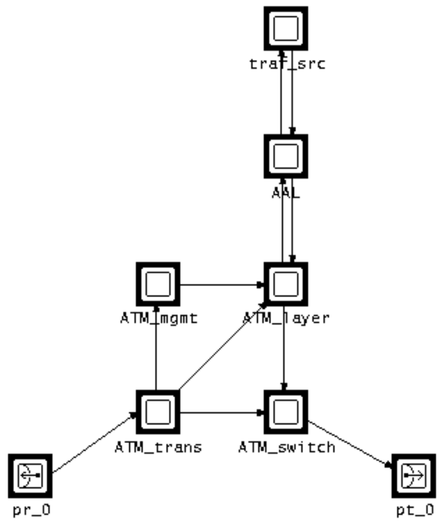


Figure 4.4. Block Diagram of Source Node in OPNET

segments the packet into cells and marks the cell corresponding to the end of the packet. The ATM_switch block receives the ATM cells through the ATM_layer block and adjusts the departure points of critical cells.

In this simulation model, the capacity of all links is equal to 155 Mbps and all connections belong to ABR service category. Since the multimedia application of interest is the real-time video, we assume that the time between the consecutive points of packet generation is 1/30 sec. The number of slots corresponding to this interval is

$$T = \frac{155,000,000(bps)}{53(byte) \times 8(bit)} \times \frac{1}{30}(sec) \simeq 12186(slot). \qquad (4.15)$$

We also assume that the bitrate of application data is 7.2Mbps, which is a typical value of the MPEG2 encoder[40]. The mean size of the application packet

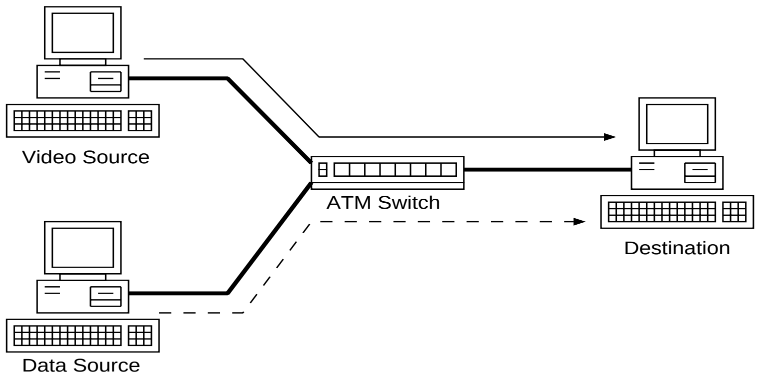


Figure 4.5. Simulation Model for Single Node Case

is given by

$$U^{'}(z)|_{z=1} + 1 = (T-1)p + 1 = \frac{7,200,000}{53 \times 8 \times 30}(cells). \tag{4.16}$$

Therefore p is set to $565/(12186-1) \simeq 0.04637$.

The number of cells for the background traffic generated within a slot is distributed according to the geometric distribution where its mean is set to 0.35 cell/slot (55Mbps), 0.50 (77), 0.57 (88), 0.65 (100) and 0.71 (110), respectively.

4.4.2 Single Node Case

We consider the network topology shown in Figure 4.5 to investigate the jitter process in the single node case. The critical and non-critical cells are generated at the Video Source as shown in Figure 4.1 and are transmitted to the Destination. The background traffic cells are generated at Data Source and transmitted to the same destination. The IDT scheme is implemented at Video Source. In this case, these two streams are multiplexed at the output buffer of the ATM Switch

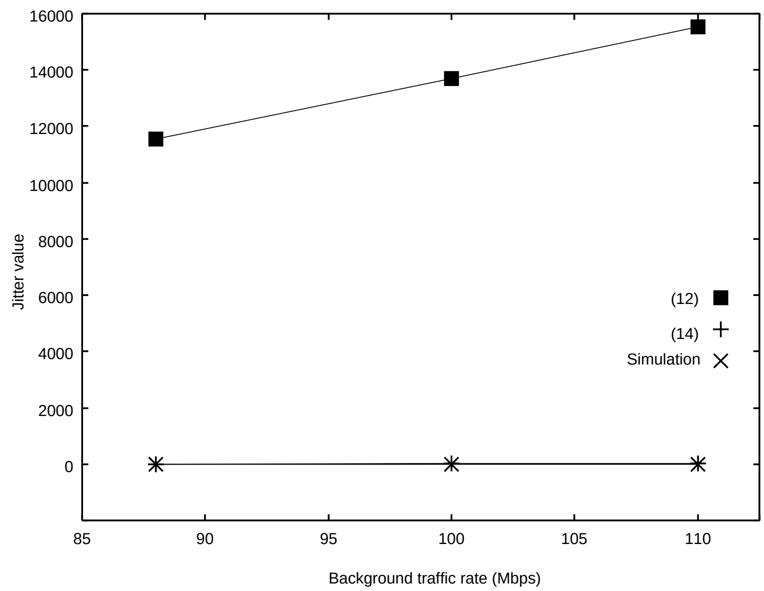


Figure 4.6. Numerical Results of Single Node Case

and share the link between ATM Switch and Destination. In the simulation, we record the interarrival times of critical cells from video Source at Destination and calculate the jitter, the variance of the interarrival time. We also calculate the jitter values using approximations (4.12) and (4.14).

Figure 4.6 shows the jitter values calculated from (4.12), (4.14) and simulation. In Figure 4.6, the horizontal axis represents the bitrate of the background traffic and the vertical axis means the jitter value ($slot^2$). We observe that the jitter values of (4.14) and simulation are almost same while the result calculated from (4.12) is quite different. This implies that T is too large to calculate the jitter from (4.12). That is, T is large enough to consider the system as $Geo^{[X]}/D/1$ and hence (4.14) is more suitable than (4.12). In Table4.1, we show the jitter values of (4.14) and simulation. Though the simulation results are not strictly equal to those of (4.14), the tendency to increase is same in both cases.

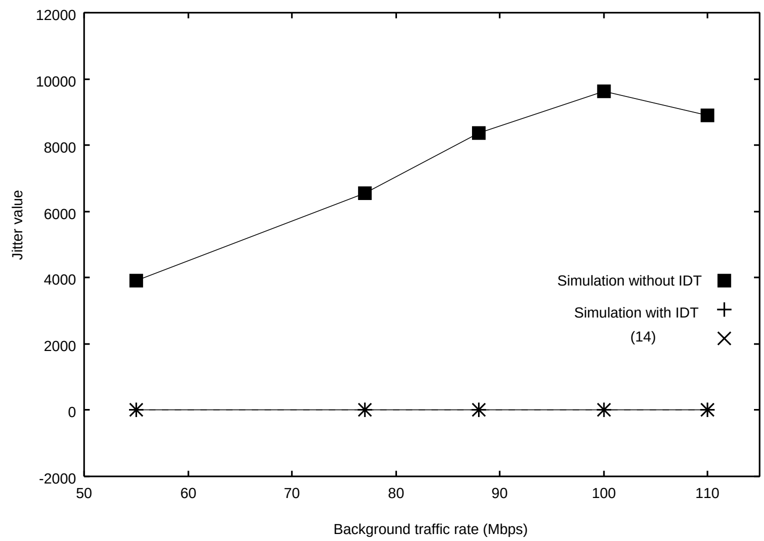


Figure 4.7. Jitter in Single Node Case

Table 4.1. Jitter Values of Simulation and Approximation (4.14)

	Background Traffic (Mbps)		
	88	100	110
Simulation	0.366	0.53	0.546
Approximation (4.14)	5.6	11.3	20.3

Figure 4.6 also shows that the jitter is small even when the rate of background traffic is large. To investigate the efficiency of the IDT scheme, we plot the jitter values without IDT in Figure 4.7. From this figure, we observe that the jitter value with IDT algorithm is smaller than that without IDT irrespective of the rate of background traffic. Intuitively, the number of cells between consecutive critical cells becomes large as the background traffic increases. In the case without IDT, the interarrival time of critical cells at ATM switch varies according to the packet size at application level of the source node and this causes the large jitter at destination. However, in the case with IDT, the interarrival time of critical

cells at ATM switch is constant and this makes the amount of background traffic less variable. Note that the mean interarrival time of critical cells at destination becomes large in both cases when the background traffic increases.

One more important characteristic observed from Figure 4.7 is that the jitter with IDT scheme is quite small and insensitive to the background traffic. This implies that the output process of the ATM switch is also constant regardless of the background traffic. Therefore we can expect that the jitter with IDT is small even in the multiple nodes case. We investigate this in the next subsection.

4.4.3 Multiple Nodes Case

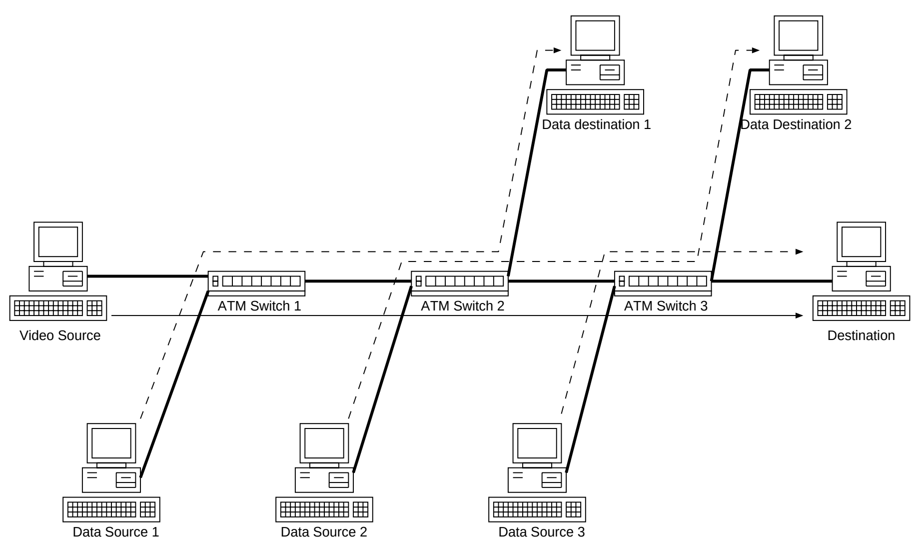


Figure 4.8. Simulation Model for Multiple Nodes Case

In order to investigate the jitter behavior in the multiple nodes case, we consider the network topology shown in Figure 4.8. In this case, there are three switch nodes in the network. The critical and non-critical cells are generated at

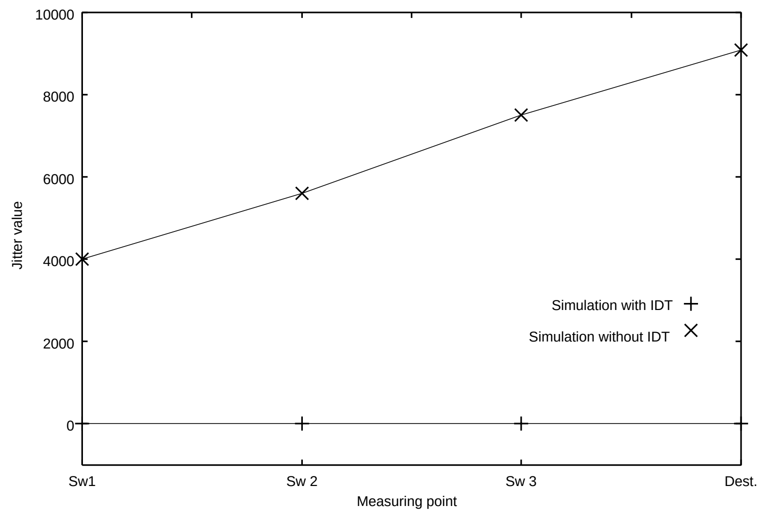


Figure 4.9. Jitter in Multiple Nodes Case with 50Mbps Background Traffic

Table 4.2. Destination of Each Source

source	Video Source	Data Source 1	Data Source 2	Data Source 3
destination	Destination	Data Destination 1	Data Destination 2	Destination

the Video Source and are transmitted to the Destination. Data Source 1, Data Source 2 and Data Source 3 generate the background traffic and Data Destination 1, Data Destination 2 and Destination are the corresponding destinations, respectively. The Destination is the destination for Video Source and Data Source 3. In this case, the traffic from Video Source is multiplexed with the background traffic from Data Source 1 at the output buffer of ATM Switch 1. Then the aggregated traffic is transmitted to ATM Switch 2 and the background traffic from Data Source 1 is switched to Data Destination 1 while the traffic from Video Source is multiplexed with another background traffic from Data Source 2. The same situation as the ATM Switch 2 occurs at ATM Switch 3 and finally the

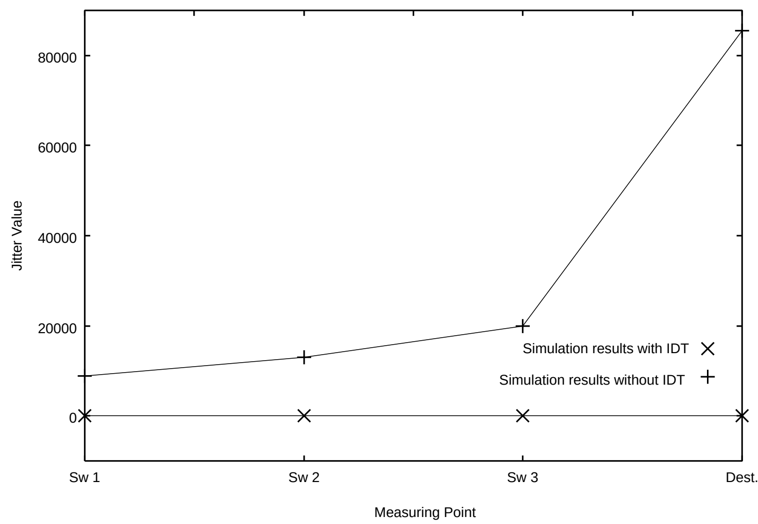


Figure 4.10. Jitter in Multiple Nodes Case with 100Mbps Background Traffic

aggregated traffic from Video Source and Data Source 3 is transmitted to Destination. We summarize the source-destination pairs in Table 4.2. As for the parameter of Video Source, we use the same values as the single node case and investigate the jitter value of the Video Source traffic when the cell-generation rate of background traffic is set to 50 and 100 (Mbps), respectively.

As we stated in the previous subsection, we cannot use (4.12) due to large T. Though [21] provides the jitter analysis in the multiple nodes case, the main results are derived with (4.12) and hence we cannot use the results in [21]. Therefore we investigate the jitter behavior in multiple nodes by simulation.

Figure 4.9 shows the simulation results with and without IDT scheme under the 50Mbps background traffic. In Figure 4.9, the horizontal axis represents the measuring point of the jitter for critical cells, and vertical axis means the jitter value. From Figure 4.9, we observe that the jitter value without IDT becomes large as the number of ATM switches increases and that it is always larger than that with IDT. We also observe that the jitter value with IDT algorithm is almost same even when the number of intermediate ATM switches increases. This is just what we expected in the previous subsection.

Figure 4.10 shows the simulation results with the 100Mbps background traffic. From this figure, we observe that the jitter value has the same tendency as the case with 100Mbps background traffic. Note that the jitter values are quite larger than those under 100Mbps case. In the case without IDT, the jitter rapidly increases at ATM switch 3. This implies that the variation of the interdeparture time between critical cells causes the large variation of the interdeparture time at next ATM switch. Therefore keeping the interdeparture time of critical cells constant is effective for reducing the jitter at destination. From this reason, the IDT scheme is efficient for assuring the QoS at application level.

4.4.4 Robustness of IDT Scheme

In order to investigate the robustness of IDT scheme against the background traffic, we focus on the dynamics of the interarrival time of critical cells at destination. Figures 4.11 to 4.14 show the simulation results with and without IDT scheme in the following case: The video traffic is transmitted to the destination during the simuation time from 10 to 15, and the data source nodes start to transmission of 100 Mbps background traffic at 11 and end at 13. In these figures, the horizontal axis represents the simulation time and the vertical axis means the interarrival time of critical cells at destination. Figures 4.11 and 4.12 are the single node case while Figures 4.13 and 4.14 are the multiple nodes case where the number of ATM switches is three.

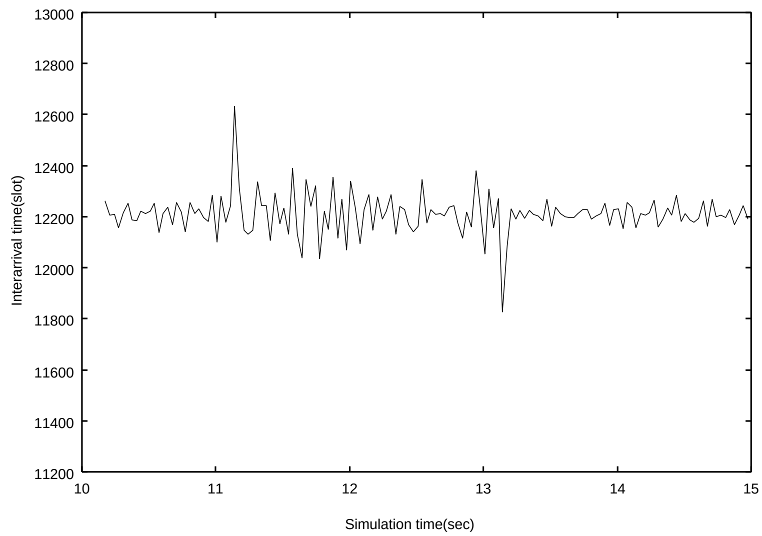


Figure 4.11. Interarrival Time without IDT in Single Node Case

From Figure 4.11, we observe that the interarrival times vary largely when the background traffic is multiplexed. We also observe that the interarrival times still vary even when there is no background traffic. This is because the packet size at application level is variable. On the other hand, from Figure 4.12, the interarrival time is almost constant even when the background traffic multiplexed.

Figure 4.13 shows the simulation result of the multiple nodes case without IDT scheme, and we find the same tendency as Figure 4.11. Note that the degree of variation of interarrival time is larger than that of Figure 4.11. Figure 4.14 is the case with IDT. We observe that the interarrival time is almost constant insensitive to the background traffic. In addition, the number of intermediate nodes does not affect the variation of interarrival time so much.

From these results, we conclude that the IDT scheme is robust against the impact of the background traffic.

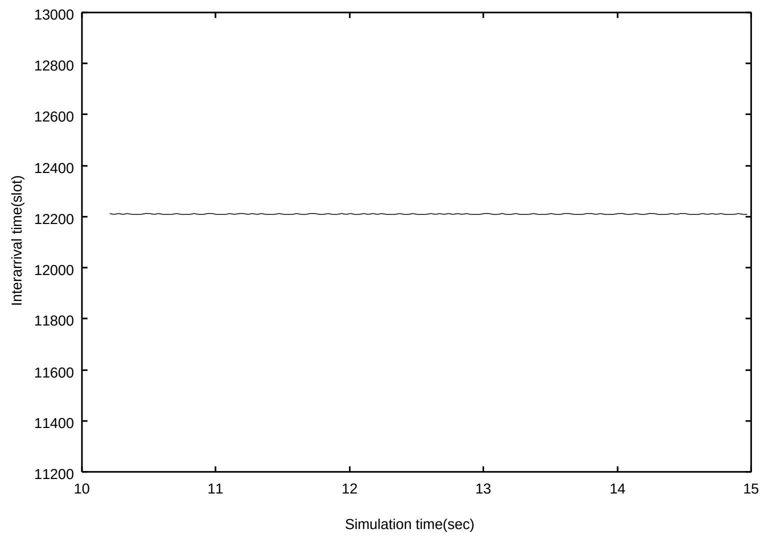


Figure 4.12. Interarrival Time with IDT in Single Node Case

4.5. Conclusion

In this chapter, we focused our attention on the departure point of the last cell for the packet and proposed IDT scheme to reduce the application level jitter. We investigated the jitter process with IDT scheme by analysis and simulation. We also compared the IDT scheme with the original ABR system. Finally we investigated the robustness of IDT scheme against the interruption of the background traffic.

As we see in the numberical examples, the variation of the interdeparture time of the tagged node causes the further variation of the interdeparture time of the next node. Therefore it is important for the source node to make the departure process of critical cells less variable. From this point, the IDT scheme is quite efficient.

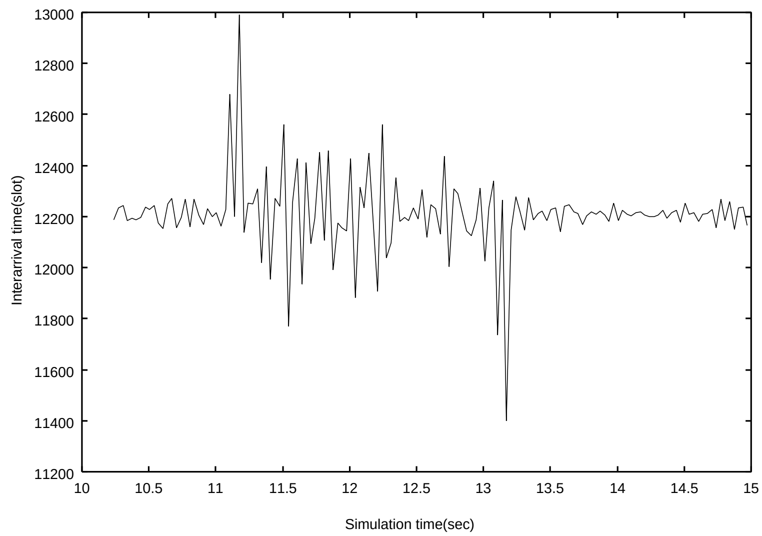


Figure 4.13. Interarrival Time without IDT in Multiple Nodes Case

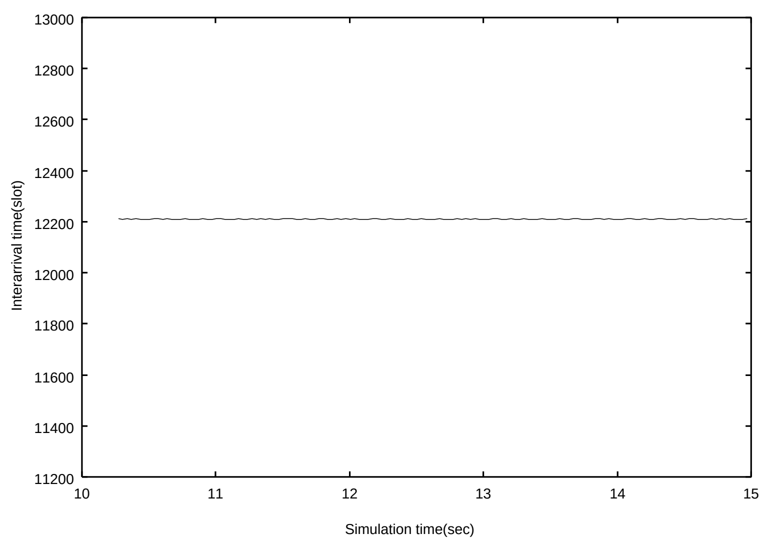


Figure 4.14. Interarrival Time with IDT in Multiple Nodes Case

Chapter 5

Conclusion

In this study, we investigated the jitter processes and proposed the scheduling scheme at the source node to reduce the jitter at application level on the ATM-ABR service class.

In Chapter 2 we presented the wide area project in Hyogo Prefecture and showed the outline of the testbed network constructed in Kobe, Nishinomiya and Amagasaki. We constructed a simulation model of the experimental network, investigated the jitter processes under some scenarios and showed how the jitter process is affected by the Internet traffic and the other cell streams of MPEG2 by the simulation. We also investigated the jitter processes with the simulation and presented the effects of the bit rates of Internet and MPEG cells and the number of ATM switches. From simulation examples, we found that when the background trafffic and the number of ATM switches increase, the jitter becomes large and it results in the degradation of QoS for MPEG2.

In Chapter 3, we investigated the characteristics of cell streams in the ATM network integrating CATV and Internet. Using the four graphical methods, we estimated the Hurst parameter of our testbed network. We observed that the Hurst parameter is around 0.6 provided that the traffic has self-similarity. In addition, we constructed the simulation model where the self-similar nature was taken into consideration and investigate the relation between the self-similar traffic and the jitter of MPEG2 cells. From simulation results, we showed that the jitter process is affected by the Hurst parameter when the Internet bitrate is small, while not affected when the Internet bitrate is large.

In Chapter 4, we focused our attention on the departure point of the last cell for the packet and proposed IDT scheme to reduce the application level jitter. We investigated the jitter process with IDT scheme by analysis and simulation. We also compared the IDT scheme with the original ABR system and we investigated the robustness of IDT scheme against the interruption of the background traffic. From simulation and numerical results, we showed that the jitter value with IDT scheme was much smaller than that without IDT irrespective of the rate of background traffic, and that our IDT scheme is quite efficient for jitter reduction when the number of ATM switches increase. In addtion, IDT scheme indicated the robustness of dynamics against the interruption of background traffic.

In practice other service categories such as CBR and VBR which have higher priority than ABR are used for communication at the same time. In such case, ACR for ABR service category is dynamically varied according to the congestion state of the network. Even in this situation, it seems that it is possible to reduce the jitter value if we use the queue control function proposed in [37, 38] and the dynamic encoding rate adjusting scheme proposed in [8, 17, 13]. Further research is needed for the mutual effects between the transmission algorithm of the source node and the rate adjusting scheme within the ATM switch.

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