# Inter-Channel Synchronization at the ATM Adaptation Layer

*S. Koutroubinas, T. Antonakopoulos and V. Makios*

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# Inter-Channel Synchronization at the ATM Adaptation Layer: The Stereoscopic Imaging Transmission Case

S. Koutroubinas, T. Antonakopoulos and V. Makios
Laboratory of Electromagnetics, Electrical Engineering Department
University of Patras, 26500 Patras, Greece

*Abstract:* In this paper an inter-channel synchronization method for multimedia services supported by the B-ISDN network is presented. The method utilizes the timing relationships existing between the various service components at the User-ATM Adaptation Layer interface and modifies the existing synchronization techniques, not only to eliminate the intra-channel delay jitter, but also to regenerate the initial inter-channel timing relations. As an example, the stereoscopic imaging service is examined, the synchronization problems arising when it is transmitted through an ATM network are discussed and the results of the application of the proposed method in constant bit rate services are presented.

## I. INTRODUCTION

One of the most important characteristics of Broadband ISDN is that the used switching technique, the Asynchronous Transfer Mode-ATM, can handle effectively the requirements of different services, usually with different traffic characteristics. Such services are: bulk data, voice, standard quality video, multimedia services etc. [1]. It is expected that in the near future, the development and expansion of the multimedia usage will speed up the B-ISDN evolution. Multimedia services are composed of various service components like video, audio, data etc. and their components require different Quality of Service (QOS) parameters when they are transmitted through the communication subsytem and generate traffic with different bit rates. Most of these components are of synchronous type since they require intra-component synchronization at the receiver side in order to eliminate the delay jitter imposed during transmission. For these reasons, various synchronization techniques have been developed so far, like measuring the filling level of a buffer at the receiver, and by using time stamps [1], [2], [3]. Another type of synchronization, the inter-channel or inter-media synchronization, is required between the various components of a multimedia service to coordinate their ordering in the time domain and various techniques addressing this problem have been proposed up to now [4]. In this work, a type of multimedia service, the stereoscopic imaging service, is examined and its synchronization problems during transmission in an ATM network are concerned.

The image produced by a stereoscopic camera is processed by a stereoscopic codec and two separate video data streams are produced: the primary and the alternate stream. The primary stream is coded as stand alone and contains the information of the left camera channel, while the alternate data stream contains information of the right channel coded with motion and disparity compensation [5]. These two data streams are transmitted over an ATM network and the stereoscopic decoder must be supplied with the respective data streams having their initial time relation. The information transmission is carried out using different virtual communication channels for each data stream and a synchronization method must be used for inter-channel timing recovery.

In this paper, an inter-channel synchronization method is presented. This method can be used for the recovery of the initial timing relations of the two synchronous traffic streams at the User-AAL interface. The implementability of the proposed method is examined from the communication point of view and some experimental results on a Constant Bit Rate (CBR) multimedia service are presented. The transmission of stereoscopic imaging can be used as a special case of a multimedia service since the primary data stream is compatible to and can be used with monoscopic codecs for 2D image presentation while the alternate data stream is useful only with the presentation, of the primary stream. The same holds with the timing information used by the ATM Adaptation entities of the proposed method.

In Section II, the initial problem is described using the information generated by a stereoscopic imaging service and the problem is generalized. The proposed method is described in Section III, where the method's adaptability to various synchronization techniques is discussed. The experimental results of the application of the proposed method to a multimedia service with CBR streams are presented in Section IV.

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## II. The Synchronization Problem

The problem of synchronization is adressed in various levels in digital communications networks and the success of solving these timing problems influence the network performance and the quality of services offered by the network [6]. Synchronization starts from the bit level where the clock of an incoming bit stream is extracted, up to the user level where application timing must be provided. The simplest synchronization problem is to adjust a FIFO output clock to the frequency variations of the input clock (Fig. 1a) for avoiding the appearance of the slip mechanism [6]. When the synchronous service data are transmitted through a point-to-point connection, the transmission clock variations must also be taken into account (Fig. 1b). At a higher level on the network structure, the synchronization is related with the absorption of jitter introduced by the network characteristics (buffering, statistical multiplexing etc). For a single synchronous service the synchronization problem becomes more complicated when this network delay jitter has to be removed close to the network interface. At the receiving end, the initial inter-channel timing relations have been lost and a FIFO buffer with a timing recovery mechanism is usually used (Fig. 1c).

In the ATM environment, a crucial problem is how to resynchronize the decoder to the encoder clock for eliminating the network delay jitter due to the statistical multiplexing [7]. The introduction of multimedia services to the network environment altered also the synchronization concept since a new type of synchronization must be achieved. This is to recover the initial timing relationship between different data streams of a multimedia service. The stereoscopic imaging service is considered to be a multimedia service since it combines synchronous and asynchronous data streams with strict inter-stream timing relationships.

A stereoscopic encoder generates two seperate video data streams: the primary channel which contains the left channel and the alternate channel which contains the right channel coded with motion and disparity compensation [5]. The right channel reconstruction can be performed by processing both data streams, while the stand-alone coded primary channel is the only requirement for the left channel reproduction.

There are three options for bit rate allocation to the primary and the alternate channels:

i) Use of constant bit rate for both channels,
ii) The total bit rate must remain constant, while variable bit rate is generated in each of the two channels and
iii) The total and the specific bit rates are variable.

When a multimedia service is transmitted through an ATM network, there are two options concerning the virtual channel allocation. First, to multiplex the two data streams at the user level and use only one virtual channel and second, to use different virtual channels for better QOS support, having the drawback of increasing the complexity of the ATM Adaptation Layer (AAL). From the communication point of view, the Convergence Sublayer (CS) of AAL is of main concern. The AAL is service specific and enhances the communication services provided by the ATM layer to the requirements of the supported data stream. It is mainly used to map the user data stream into the information field of the ATM cells and provides services like error handling, timing control etc. Among other functions, the CS handles the delay jitter introduced by the network using synchronization techniques to transfer timing information between source and destination.

When the various components of a multimedia service are supported separately by the communication subsystem, they are processed independently, following the QOS parameters determined during connection set-up. The CS-entities of each connection generate the respective timing control and the clock is reconstructed at each CS-entity. Following such a method, the internal synchronization at each multimedia component is performed at the decoder side.

It is obvious that the independence of the various channels

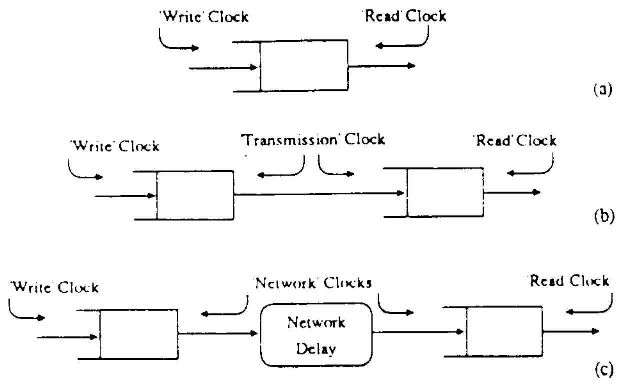


Fig. 1. The synchronization problem approaches

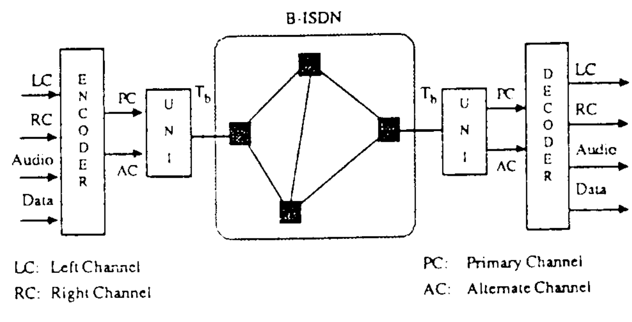


Fig. 2. The stereoscopic imaging service over B-ISDN

at the communication system makes the system more flexible and simpler but one of the basic characteristics of a multimedia service is not exploited. That is the initial inter-channel synchronization, which is lost after the data reception in the ATM Adaptation Layer at the User Network Interface (UNI). The information flow of such a service over the B-ISDN architecture is shown in Figure 2. In services like the stereoscopic image, where the primary data stream can be used as synchronization reference, a modification of the CS structure must be done in order to include the inter-channel timing information in the used time control method of the various virtual channels which correspond to the same multimedia service.

## III. The Inter-channel Synchronization Method

The basic method for synchronizing constant bit rate services in ATM networks has been described in [2] while for variable bit rate services, a solution to the synchronization problem has been proposed in [8]. For CBR services, the PLL control using the receiver's buffer filling level is adequate, but the use of time stamps, which are not influenced by the bit rate fluctuations, is a must concerning VBR services.

The interconnetion at the User-AAL interface for the same data stream is plesiochronous since both parts (the transmitter and the receiver) user the same nominal bit rate but their actual values differ in certain limits. In Fig. 3a, the $f_T$ and $f_R$ are plesiochronous but the relation of $f_{NR}$ and $f_R$ must be considered asynchronous. The $f_{NR}$ denotes the rate of the incoming traffic at the receiving buffer and is the result of the statistical multiplexing of $f_T$ with the other network services. The influence of the statistical multiplexing is eliminated using a PLL for modifing the output frequency ($f_r$), in order to filter the delay jitter deviation. The proper selection of the buffer size and the PLL parameters according to the traffic characteristics can absorb the delay fluctuation. The example in Fig. 3a shows two initially time correlated data streams which use two different virtual circuits in the network. At the receiver side a PLL is used in each stream to recover the respective's transmitter clock but the initial phase relation of the two data streams is not reestablished. For this reason, a new synchronization approach is proposed to be used in the receiver side, as it is shown in Fig. 3b. One of the two data streams is considered to be the primary channel while the other is called alternate and includes complementary information to the primary one. This is the case in most multiple-stream services, like the stereoscopic imaging, as it was explained previously. The proposed method is called inter-channel synchronization since it can regenerate the initial inter-channel timing relations at the receiver side. The PLL at the primary channel operates as a stand-alone unit for timing recovery at this channel and generates an output signal, the phase correction signal, indicating the variation of the output frequency, from its initial value. At the alternate channel, the PLL uses three parameters to determine the actual output frequency: the receiver's initial frequency for this channel, the value of the estimated transmiter's frequency and the phase correction signal from the primary channel. In this case the PLL is used to preserve the same frequency variance at both streams from their nominal values.

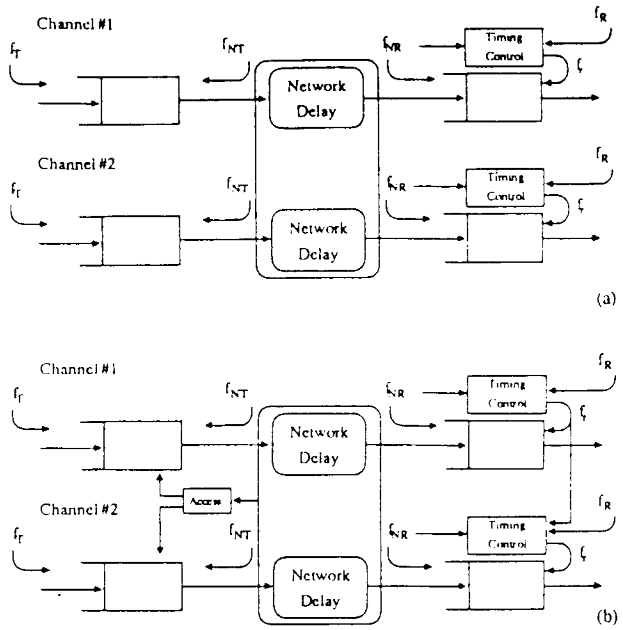


Fig. 3. The timing model of two multimedia data streams: (a) when no inter-channel synchronization is used and (b) when the inter-channel synchronization method in different virtual circuits is used.

When the buffer filling level method is used, the primary PLL estimates its length variation from the target filling level and sends this information to the alternate PLL. The alternate PLL uses this information in the frequency shift procedure in order to achieve the same filling level variation instead of approaching its initial target filling level. The PLLs' structure at the receiving node for two time-correlated virtual channels is shown in Figure 4. The buffer filling level counter accepts 'write - W' or 'cell loss detection - CL' pulses for incrementing its value, while a 'read - R' pulse is used for each byte accepted by the User level. The Length Comparator of the primary channel estimates also the filling level deviation and controls the 'Limit Adjustment' unit of the alternate channel. The function of the generated frequency depends on the target buffer filling level of the respective buffer according to the relations of Figure 5. The frequency of the primary PLL is given by the following equation:

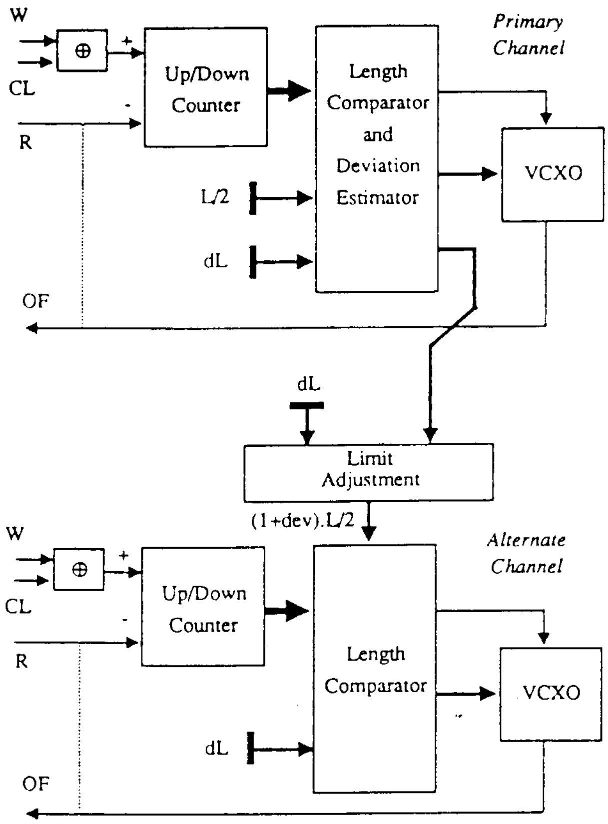


Fig. 4. The structure of PLLs at the receiving node using the inter-channel synchronization method.

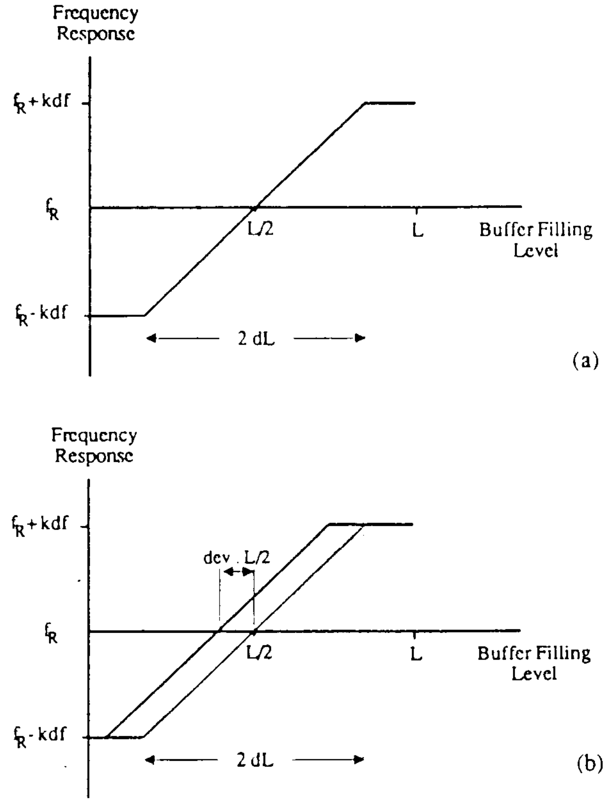


Fig. 5 The frequency response of the two PLLs at (a) the primary channel and (b) the alternate channel.

$$f = \begin{cases} f_R - k.df & 0 \le \lambda < \frac{L}{2} - dL \\ f_R + \frac{k.df}{dL}\cdot\left(\lambda - \frac{L}{2}\right) & \frac{L}{2} - dL \le \lambda \le \frac{L}{2} + dL \\ f_R + k.df & \frac{L}{2} + dL < \lambda \le L \end{cases}$$

while the frequency of the alternate PLL is equal to:

$$f = \begin{cases} f_R - k.df & 0 \le \lambda < \frac{L}{2}.(1+dev) - dL \\ f_R + \frac{k.df}{dL}\cdot\left(\lambda - \frac{L}{2}.(1+dev)\right) & \frac{L}{2}.(1+dev) - dL \le \lambda \le \frac{L}{2}.(1+dev) + dL \\ f_R + k.df & \frac{L}{2}.(1+dev) + dL < \lambda \le L \end{cases}$$

given that:

L: the total buffer length,

$\lambda$: the buffer filling level,

$f_R$: the nominal output bit rate,

k.df: the maximum frequency deviation and

dev: $1 - 2.\lambda/L$.

These values are determined independently in each PLL using the nominal bit rate of the supported service.

The proposed method can be easily implemented in the Convergence Sublayer (CS) of AAL, where the User-AAL interface buffer is placed. The method requires a new CS structure which must allow information exchange between CS-entities in the same system, probably using the management plane. During connection set-up, one CS-entity becomes the master entity of a cluster of entities which supports the same multimedia service. The other CS-entities of this cluster behave like slave-entities in the following notion: The master CS-entity generates the local clock using the timing information of its virtual channel, but also provides timing information to the slave CS-entities. Using this method, the regenerated data streams at the AAL interface have their initial timing relation and can be more easily used by the decoder for generation of the initial multimedia service. When the buffer filling level method is used, the inherent cell error rate of the network must be taken into account, otherwise a permanent error may be introduced. A cell loss introduces an error in the frequency estimation procedure since it acts like noise in the filling level/frequency function. This type of error can be characterized as additive because it adds a permanent erroneous offset at the frequency estimation. In the case of high cell error rate at the primary channel, the inter-channel synchronization method will fail when no cell

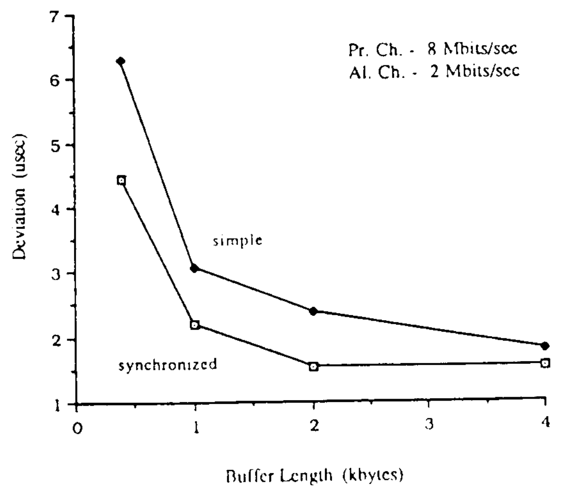


Fig. 6. The deviation of the inter-channel synchronization versus the buffer length at the receiving side.

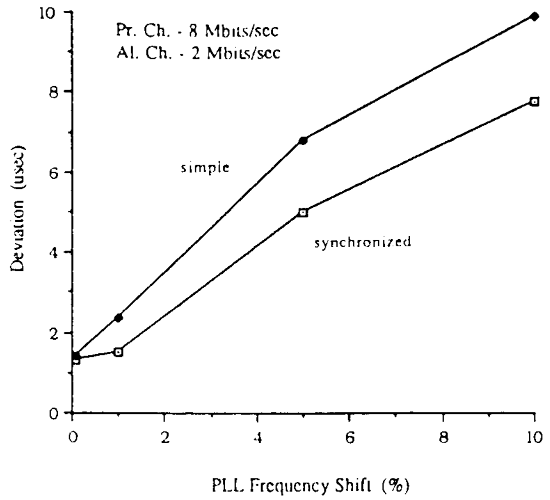


Fig. 7. The deviation of the inter-channel synchronization versus the PLL frequency shift from its nominal value.

loss compensation is used. A solution to this problem is to use the cell sequence numbering at the Segmentation and Reassembly Sublayer (SAR) of AAL, so that, when a cell loss is detected, the filling level indicating counter at CS is incremented (the buffer's actual write pointer is not affected) and the cell loss has no influence at the inter-channel synchronization procedure.

In the next section, some experimental results for constant bit rate services are presented and the performance of the proposed method is analysed.

## IV. Experimental Results on CBR Services

For analyzing the performance of the proposed method, constant bit rate has been allocated to both channels. The bit rate of the primary channel has been set to 8 Mbits/sec while the alternate channel has 2 Mbits/sec data rate. The data are transmitted using 47 bytes of the ATM cell payload. In the receiving side, data delivery begins when half of the respective buffer is full. For the network delay jitter characteristics, the assumptions of [7] have been used. The cell inter-arrival jitter has a Gaussian distribution
and for 80% network load, its maximum range is ±250 μsec. The method has been tested under various traffic conditions and for various system configurations. The results presented in the following discussion determine how the performance of the method is affected by:

- The total buffer length on the receiving side and
- The PLL frequency response.

Figure 6 shows the deviation of the inter-channel synchronization versus the buffer length at the receiving node. When the buffer length increases, the influence of the proposed

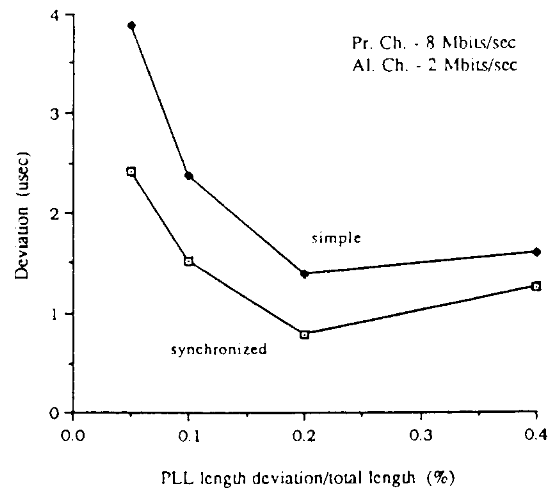


Fig. 8. The deviation of the inter-channel synchronization versus the ratio of the PLL length deviation to the total buffer length.

method decreases since the buffer absorbs the traffic fluctuations more easily. The inter-channel synchronization method can be used in order to decrease the receiver's buffer length when a specific deviation of the inter-channel jitter must be achieved. The influence of the PLL frequency shift to the jitter deviation is shown in Figure 7. The PLL frequency is determined according to the function depicted in Figure 5. The slope of this function is equal to k.df/dL. When a greater deviation is allowed to the PLL frequency and the length limits remain constant, the jitter deviation increases. When the length deviation

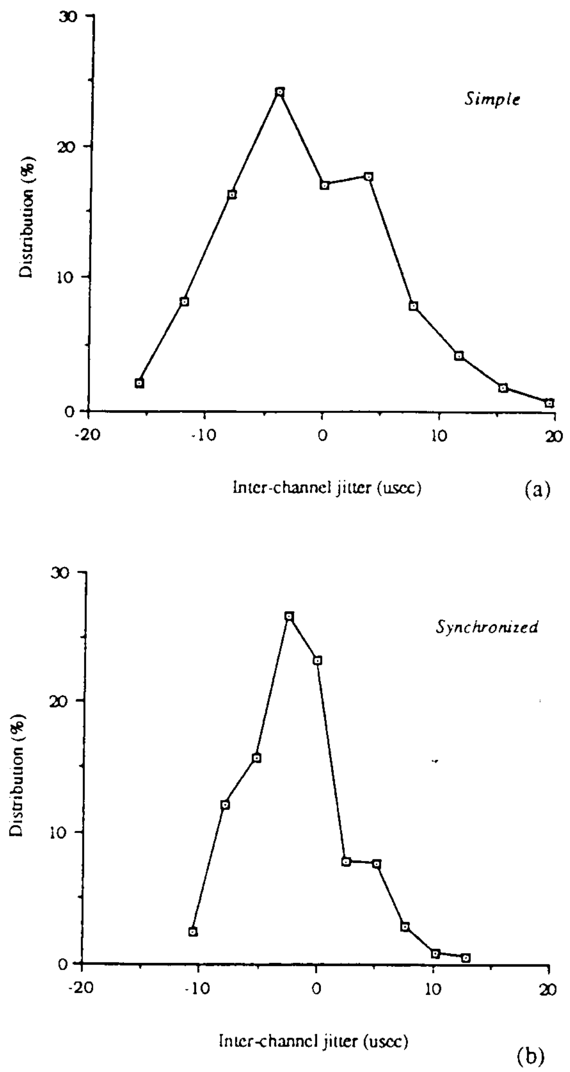


Fig. 8. The distribution of the inter-channel synchronization jitter for (a) the simple case and (b) the proposed method.

increases and the frequency area remains constant, the jitter deviation decreases. In both cases, when the PLL is very sensitive to the buffer length changes, the performance of the proposed method decreases. Finally, the distribution of the inter-channel synchronization jitter is shown in Figure 9. Using inter-channel synchronization, the jitter is concentrated near zero, while in the simple case, it is spread in a wider area.

## V. Conclusions

An inter-channel synchronization method at the User-ATM Adaptation Layer interface of B-ISDN network has been presented in this paper. The proposed method exploits the timing relations of the various service components at this interface and modifies existing synchronization techniques for regeneration of the initial inter-channel timing.

The proposed method has been applied on the 'buffer filling level' PLL control method for supporting constant bit rate services. For experimental purposes, the constant bit rate allocation to both channels of a stereoscopic imaging service was considered and as the results show the use of the proposed method improve the inter-channel synchronization. The receiver's buffer length and the PLL timing characteristics are the critical parameters and their proper selection will determine the method's implementability.

## References.

[1] Martin de Prycker: "Asynchronous Transfer Mode: Solution for Broadband ISDN", Ellis Horwood Ltd, 1991.

[2] J. Cochennes, P. Adam and T. Hondoin: "Asynchronous Time-Division Networks: Terminal Synchronization for Video and Sound Signals" in *Proc. IEEE GLOBECOM*, 1985, pp. 25.6.1-25.6.4.

[3] W. A. Montgomery: "Techniques for Packet Voice Synchronization", *IEEE Journal on Selected Areas in Communication*, Vol. SAC-1, No. 6, December 1983, pp. 1022-1028.

[4] T. D. C. Little and A. Ghafoor : "Multimedia Synchronization Protocols for Broadband Intergrated Services", *IEEE Journal on Selected Areas in Communications*, Vol. 9, December 1991, pp.1368-1382.

[5] RACE-II Project R2045 DISTIMA: "Annual Project Review Report", R2045/TUD/IT/DS/B/001/b1, September 1992.

[6] P. Kartaschoff: "Synchronization in Digital Communications Networks", *Proceeding of the IEEE*, Vol. 79, No. 7, July 1991, pp. 1019-1028.

[7] J. P. Leduc and S.D. Agostino: "Universal VBR Videocodec for ATM networks in the Beglian Broadband Experiment". *Signal Processing: Image Communication 3* (1991) pp. 157-165.

[8] W. Verbiert and L. Pinnoo: " A Variable Bit Rate Video Codec for Asynchronous Transfer Mode Netowrks", *IEEE Journal on Selected Areas in Communication*, Vol. 7, No 5, June 1989.