Performance analysis of a multimedia synchronization mechanism based on buffer compensation in a mobile environment

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In a mobile communication system network performance varies considerably when handovers occur. This occurrence strongly impacts the design of the buffer compensation based techniques usually used in the fixed communication environments for minimizing probability of asynchronism between the different media composing a multimedia session. This paper provides an analytical paradigm for dimensioning synchronization buffers at the interface node between the wired and the wireless networks when network delay varies during a multimedia session due to the user mobility. For this purpose, the factors related to terminal mobility which have to be taken into account in the design of a synchronization mechanism are briefly introduced and appropriate user-perceived Quality of Service parameters referring to the synchronization of multimedia services are also introduced

I. Introduction

In the last few years, multimedia systems and mobile communication systems have been leading factors in the telecommunications market. Multimedia computing, communications and services in the wired network sector have reached a certain degree of maturity thanks to the considerable effort devoted to developing techniques ensuring an adequate user-Perceived Quality of Service (P_QoS) [1, 2]. In particular, new performance requirements specific to multimedia services - i.e. the maintenance of both intramedia and intermedia time relationships - have been defined and taken into account. At the same time, mobile communication systems have spread in the field of telephone and low-speed data transmission services for mobile users [3].

The mobile communication market will certainly develop in a more consistent manner if users can be provided with services and performance levels comparable to those provided in a fixed communication environment by the wired networks. For this reason, considerable attention is being paid to wireless ATM networks, whose final goal is the seamless wireless extension of ATM to mobile devices [4, 5].

In a fixed communication environment where Quality of Service (QoS) can be controlled (as in the case of networks based on ATM technology), users of multimedia services are guaranteed that synchronization requirements are met thanks to various control and network resource management mechanisms inserted in the networks. In a mobile communication environment, on the other hand, the problem is understanding whether the same mechanisms can still be used and what new parameters linked to user mobility need be taken into account. In fact, some of the techniques currently used in wired networks may be inefficient or impossible to implement in a mobile environment because of the routing changes occurring in the wired network to guarantee connectivity to the roaming users and unreliable and time-varying radio transmission channel.

In this paper we deal with some aspects of the problem of synchronizing multimedia streams in a mobile communication environment where wireless networks are interconnected by a wired broadband network. First, the problem of synchronizing multimedia streams is outlined and some indications about the impact of terminal mobility on the design of a mechanism for intra/intermedia synchronization are pointed out. In particular some of the main techniques currently used in wired networks are briefly reviewed in order to discuss their capacity to operate in a mobile communication environment and an architecture for multimedia synchronization is presented. In order to quantify the effectiveness of the synchronization mechanisms we also introduce P_QoS parameters which take into account the effect of user mobility in the synchronization of multimedia streams. Next, we propose a scheme for guaranteeing multimedia synchronization requirements, based on the use of compensation buffers (CBs) to equalize wired network delay, placed at the interface between the wired and wireless networks. Then, an analysis of the synchronization technique introduced is provided, by deriving analytical relations that allow statistical characterization of the delay jitter and skew not compensated for by the mechanism and the loss probability due to handover. Finally, some considerations are made about how the synchronization mechanism parameters have to be sized.

The paper is organized as follows. The next section introduces the problem of multimedia synchronization in a mobile environment. Section 3 describes the communication system model taken as a reference in the paper and introduces P_QoS parameters which take into account the effect of user mobility in the synchronization of multimedia streams. Section 4 gives some guidelines for the definition of an architecture for a synchronization system in a mixed wired/wireless network environment. Section 5 focuses on the synchronization of multimedia streams by using compensation buffers (CB) in the radio access interface, and shows how CB sizing affects P_QoS parameters. Section 6 gives some indications about how to apply the proposed synchronization mechanism. Section 7 contains final considerations on the topic dealt with in the paper.

II. Problem definition

A characteristic of multimedia services is the synchronization [1]. A multimedia stream is, in fact, characterized by multiple monomedia streams related to each other by means of time relationships which must be preserved. Due to the various delays the monomedia streams may undergo during transmission in a communication network, appropriate mechanisms must be introduced so as to meet both the requirements of each monomedia stream (*intramedia synchronization*) and those related to how the monomedia streams are integrated to form the multimedia stream (*intermedia synchronization*) [6, 7]. These requirements, in a wired communication environment, are basically linked to the end-to-end delay jitter of the presentation units and their skew, that is, the difference between the instantaneous delays of presentation units belonging to two different monomedia streams. As measurements of human perception have shown that monomedia streams may appear to be "in synch" if the delay jitter and skew are limited to appropriate values [1], the P_QoS parameters are usually expressed as restrictions on the statistics of both the jitter and the skew [7].

In literature several mechanisms dealing with the problem of limiting delay jitter and/or skew in multimedia services on a wired network have been presented [6, 7, 8, 9, 10, 11, 12, 13, 14]. The majority of these mechanisms can be classified into two categories: the first one comprises mechanisms based on the use of buffers at the destination site or at the intermediate nodes, the second on modification of retrieval times at the source site. The application of buffering techniques at the destination site requires the bounds of the delay jitter introduced by the network to be known a priori or estimated [15, 16, 13]. These techniques are extremely simple to implement at the application level, and for this reason are widely used in multimedia services today [11]. They are also applied in the case of fixed hosts connected to a wired network by means of a wireless link [17]. Unfortunately, the buffer needed to guarantee the synchronization requirements may sometimes be so large that it causes unacceptable delays. Moreover, when the task of containing network delay within suitable limit values is distributed over the whole network, buffering has to be applied at the intermediate nodes and a great end-to-end average delay may occur. The second category of synchronization mechanisms can be implemented in various ways according to how the retrieval time modification is computed; some follow a deterministic approach [18, 9], others a statistical approach [7, 19]. In any case, the modification of retrieval times permits a reshaping of the probability density function (pdf) of the delay jitter at the destination site, but requires either a priori knowledge of the characteristics of the whole multimedia stream, or a transmission overhead due to the presence of feedback traffic.

All the above synchronization techniques were devised to operate in a fixed communication environment. They, in fact, often assume the availability of: 1 – wired virtual connections whose statistical characteristics are predictable in certain time intervals; from the statistical point of view the wired network is often seen as a wide-sense stationary system; 2 – adequate resources at the intermediate nodes and at the source and destination sites, in terms of both buffers and the computing power needed to implement control techniques with varying degrees of sophistication; 3 – reliable, high-quality transmission media, i.e. high transmission capacity, low error rates and limited variations in the end-to-end latency time.

In a mobile communication environment, on the other hand, the statistical characteristics of the connections vary in time and disconnections may occur depending on both the type of user mobility and how re-routing is performed in the event of handover [20, 21, 22, 23]. Mobile users moreover, have at their disposal relatively unreliable, poor-quality wireless transmission channel and limited hardware and software resources due to the size of portable terminals and the limited amount of power supplied by their batteries. These are all elements that have to be taken into account, as they have a great impact on the design of multimedia synchronization techniques and can render the techniques and protocols used in a fixed environment inefficient or impossible to implement in a mobile environment. For example, synchronization mechanisms which involve an initial estimate of either the delay bounds [13] or the resynchronization interval in which reference times are updated [19], or those requiring the transmission of feedback units to detect any asynchronism and keep it within certain limits [18], may fail if the channel characteristics undergo sharp variations in a short time or if there is a disconnection. In a mobile environment the latency time bounds cannot be predicted in the long term, because channel characteristics vary whenever a handover occurs.

Moreover, the handover phenomenon imposes certain constraints on synchronization mechanisms: the increase in the wired network delay has to be limited, in order to facilitate the retrieval of information already routed towards the old radio access port and to reduce loss of information; the synchronization algorithm has to be highly adaptive, to reach the steady state in a very short time and to be unaffected by discontinuities introduced by the handover. At the same time, the hardware and software resources needed by the synchronization mechanisms and the exchange of control information between a mobile terminal and a radio access port have to be limited, in order to reduce power consumption in the mobile terminal.

III. Reference scenario and quality of service parameters

In this paper we take as our reference scenario a communication system made up of a wired broadband network interconnecting static hosts and a wireless access network with mobile terminals (MTs). Some of the static hosts (the Base Stations - BSs) act as a radio access interface, thus allowing MTs located in a cell to access the wired network. Cells may partially overlap, but we assume that even if an MT can exchange control information with more than one BS, it only exchanges user information with one BS at a time.

In this scenario the MT is the destination of a multimedia stream. This stream is received through the BS and it is the combination of two or more monomedia streams, related to each other by logical, spatial and/or temporal relationships which must be preserved during the presentation of the multimedia stream. As it is our intention to focus on use of buffers in the BS to equalize wired network delay, we will only consider a multimedia stream in which compound monomedia streams are temporally synchronized. The sources of monomedia streams can be located in a single static host or distributed over the wired network: in any case we assume that the streams are transported autonomously through the wired network.

Each monomedia stream is seen as being made up of an ordered sequence of Information Units (IUs) [1, 24], the size of which affects the synchronization granularity. Details concerning the architecture or protocol stack used in wired and wireless networks are not dealt with here, as they lie beyond the scope of the paper.

The end-to-end delay between the sender and receiver of the monomedia streams is the summation of several terms, such as the time needed at the source site to collect and prepare IUs for transmission, the network delay and the time needed at the destination site to process and prepare IUs for playback [10]. Here we will only consider network delay. However, the same line of reasoning can be used to include the other two kinds of delay in solving the synchronization problem.

In the reference scenario outlined above the network delay the n-th IU of the i-th media undergoes when the MT is located in the radio-electric coverage area of the L-th BS, $d_i^L(n)$, comprises the delay between the source and the L-th BS, any delay introduced in the BS, and the access and propagation delays in the wireless channel. Obviously $d_i^L(n)$ depends on the mobility of the MT because the path through the wired network changes as the user moves towards a new BS. As $d_i^L(n)$ is a random process, the intramedia and intermedia time relationships between IUs are not maintained at the destination site. Synchronization mechanisms therefore have to limit variations in the network delay, that is, they have to equalize $d_i^L(n)$.

As far as intramedia synchronization is concerned, obviously, the i-th media can be said to be perfectly synchronized if $d_i^L(n)$ is constant for any n. In order to quantify the effect of synchronization mechanisms on network delay equalization, let us define the *residual jitter*, $j_i^L(n)$, of the n-th IU of the i-th media which is transmitted to the MT through the L-th BS as the difference between the actual delivery time at the MT and the ideal delivery time by which the intramedia time relationship should be recovered. As far as intermedia synchronization is concerned, the time relationships between IUs belonging to different media are altered on account of the different starting times for the sources and the different delays the IUs undergo. Bearing in mind, however, that the starting times can be controlled when the connection is set up, we will henceforward assume, without loss of generality, that they are the same for all the monomedia streams. So, the displacement in time of corresponding IUs of different media is only due to different network delays. Therefore, by considering the above definition of *residual jitter* we define the *residual skew*, $s_{k,i}^{L}(n)$, between the n-th IUs of the media "k" and "i" received by the MT through the L-th BS as the displacement in time of the corresponding IUs after the application of intramedia synchronization mechanisms.

A multimedia stream appears to be "in synch" even if the residual jitter and skew are not identically null, but are limited to appropriate values [1]. At the same time, their effect on synchronization quality depends on the kinds of monomedia streams which make up the multimedia stream. Bearing in mind that synchronization mechanisms based on compensation buffers act by introducing an additional delay on the IUs, such as to equalize their end-to-end delay, and therefore that there has to be a monomedia stream which can be used as a time reference to syncronize the various monomedia streams, we assume that monomedia streams are classified as *master* and *slave* streams [8, 25].

From the multimedia synchronization point of view, in wired networks the P_QoS the multimedia system has to provide the user with can be defined in terms of the probability that the residual jitter suffered by the master stream and the residual skew suffered by each slave stream with respect to the master stream lie beyond a certain range of admissible values [7]. This is also valid in the case of communications in a mobile and wireless environment within a single cell, as the end-to-end delay an IU undergoes during a connection can be assumed as a wide-sense stationary process during the time interval the MT resides in the same cell. On the other hand, statistic bounds on residual jitter and skew are not sufficient to specify P_QoS, as they do not take into account any effect of user mobility. Let us note, in fact, that when an MT passes from one cell to another, and thus a handover takes place, two phenomena may occur:

- a sharp variation in the statistical properties of the end-to-end network delay;
- a loss of the IUs that were only routed towards the BS to which the MT is no longer connected.

The first phenomenon causes a time displacement in the IU delivery times at the MT even if no IUs are lost in the BS. In fact, if the MT passes from the L-th cell to the M-th cell, the time displacement in IU delivery times, $D_i^{M,L}$, is the difference between the end-to-end delay of the first IU received by the MT through the M-th BS and that of the last IU received by the MT through the L-th BS. If $D_i^{M,L}$ is greater than a certain threshold value, a pause must be introduced in the playback of the IUs; if, on the other hand, it is less than another certain threshold value, an IU must be skipped in the playback. Let us note that if $D_i^{M,L} \neq D_k^{M,L}$ the above phenomenon also affects the intermedia synchronization, as a skew occurs between media "i" and "k". The user therefore has to specify the P_QoS by requiring statistical bounds on the value assumed by $D_i^{M,L}$ and $D_i^{M,L} - D_k^{M,L}$, for any 'i' and 'k', every time a handover occurs.

The second phenomenon related to user mobility, i.e. the loss of IUs when a handover occurs, depends on how the handover is managed. In any case the definition of the P_QoS has to include a bound on the IU loss probability due to handover or on the number of IUs lost every time an MT moves towards a new BS.

On the basis of the above considerations the P_QoS in a mobile communication environment can therefore be specified as:

$$p\{j_{master}^{L}(n) \notin [J\_min_{master}, J\_max_{master}] < \varepsilon_{master}\} \text{ for the master} \quad (1)$$

$$p\{s_{slave,master}^{L}(n) \notin [S\_min_{slave,master}, S\_max_{slave,master}] < \varepsilon_{slave,master}\} \text{ for the slaves} \quad (2$$

within a cell, and

$$p\{D_{master}^{M,L} \notin [D\_min_{master}, D\_max_{master}] < \eta_{master}\} \text{ for the master} \quad (3)$$

$$p\{D_{slave}^{M,L} \notin [D\_min_{slave}, D\_max_{slave}] < \eta_{slave}\} \text{ for the slaves} \quad (4)$$

$$p\{n_{master} > N_{master}\} < \zeta_{master}\} \text{ for the master} \quad (5)$$

$$p\{n_{slave} > N_{slave}\} < \zeta_{slave}\} \text{ for the slaves} \quad (6)$$

for each handover between cells, where

- the symbol $p\{.\}$ is used to indicate probability
- J_min_{master} and J_max_{master} are the minimum and maximum admissible residual jitter that the master stream can undergo;

- $S\_min_{slave,master}$ and $S\_max_{slave,master}$ are the minimum and maximum admissible residual skew;

- D_min_{master} and D_max_{master} are the minimum and maximum admissible values of the time displacement in IU delivery times for the master stream when a MT passes from one cell to another;

- D_min_{slave} and D_max_{slave} are the minimum and maximum admissible values of the time displacement in IU delivery times for the slave stream when a MT passes from one cell to another;

- n_{master} and n_{slave} are the number of master and slave stream IUs lost when a handover takes place;

- ε_{master}, $\varepsilon_{slave,master}$, η_{master}, η_{slave}, ζ_{master} and ζ_{slave} are the maximum admissible values for the relevant probabilities.

Therefore, the P_QoS parameter, that is, the set of values which specifies the P_QoS requested by the user, is the set of the above bounds, which depends on the characteristics of the multimedia stream [1]. Therefore, to guarantee P_QoS parameters the synchronization mechanisms have to reshape the probability density function of the network delay so as to satisfy relations (1)–(6).

The above relations constitute a subset of the QoS parameters that specify a multimedia service in a wired/wireless communication environments. In fact, according to ITU-T Rec. E.800 the QoS *"is the collective effect of service performances which determine the degree of satisfaction of a user of the service"* [29]. QoS has therefore to take into account available throughput, end-to-end delay, jitter delay and error rate in wired and wireless channel, multicasting and broadcasting capabilities, loss profile, probability of seamless communication, etc.. In the contest of this paper we only consider the QoS parameters which have been included in relations (1)–(6), as we intend to focus on the degradation of multimedia synchronization due to delay variation in the wired section of the network: an analysis of how QoS has to be specified for multimedia and mobile services, and how the QoS is translated into application requirements, is controlled and managed, is an open issue and is out the scope of the paper.

IV. Architecture of a synchronization system for a wireless network

In a mixed wired/wireless network, synchronization mechanisms have to be designed in such a way as to minimize the impact of mobile users on the wired network; by so doing, in fact, multimedia applications destined to static hosts can use the mechanisms oriented to the wired network without being affected by synchronization mechanisms for MTs. A fundamental role in solving the problem of synchronization is therefore played by the interface node between the wired and wireless network, i.e. the BS. The location of synchronization mechanisms on the BS, in fact, allows the system designer to make the application on the wired network unaware of problems deriving from the wireless network and user mobility by confining them to the interface with the wireless network [26].

Fig. 1 shows a possible solution as to which synchronization mechanisms can be inserted, and where, in the communication reference scenario. In this figure the delay control mechanisms are located in four different places: the source site, the intermediate nodes, the BS and the MT. At the source site, for example in the case of multimedia retrieval services, it is possible to locate a mechanism which acts on the IU retrieval times according to the feedback information it has received from either the destination site if this is an static host [7, 18, 9], or the BS if the destination site is an MT. At the intermediate nodes of the wired network we can locate mechanisms which have the task of imposing maximum and minimum delay values in each node involved in the connection [15, 11]. In order to achieve the required independence between the wired and wireless parts of the network, the mechanism located in the BS "faces" the wired network in order to equalize the delay suffered by IUs on the wired network, for example by means of a compensation buffer (CB) [27], and, at the same time, to provide the feedback information needed by the mechanism located at the source site, if necessary; on the other hand, the mechanism in the BS "faces" the wireless network in order to compensate for its delay jitter, by acting, for example, on the access technique of the wireless channel [28]. Finally, a proper mechanism should be located in the MT which compensates for the jitter by using CBs, and detects and notifies the BS of asynchronism in the multimedia stream.

To reach the overall synchronization target, some or all of the mechanisms mentioned above can be used, according to the kind of service (for example, retrieval or real-time service), link (wired or wireless) and, if

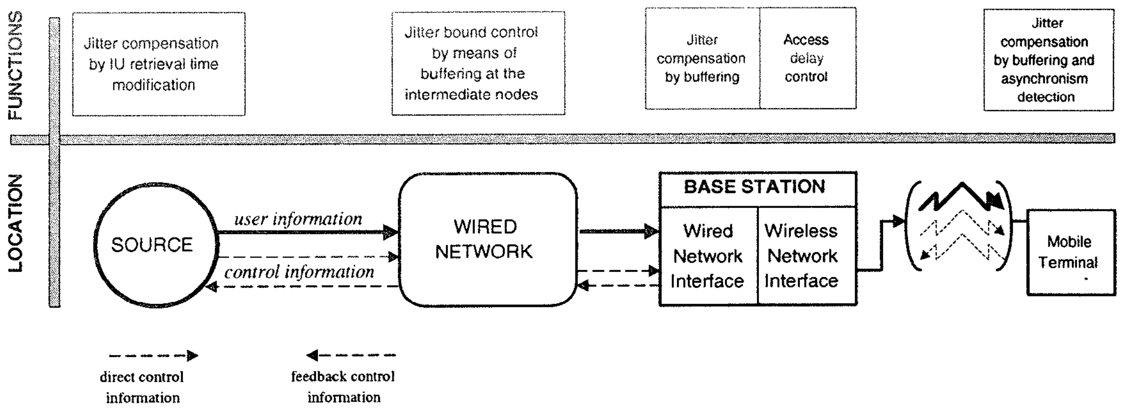

Figure 1: Possible locations and functions of multimedia synchronization mechanisms.

the user is a mobile one, its mobility. For example, prediction-based mechanisms or synchronization mechanisms based on CBs located at the destination site can be activated if a multimedia stream is sent to users connected to the wired network or to fixed users using a wireless link, according to the statistical characteristics of the wired network delay or the P_QoS parameters [7]; on the contrary, only sychronization mechanisms which are not affected by user mobility can be activated when the multimedia stream is sent to an MT. However, bearing in mind that a synchronization mechanism always has to be provided at the boundary between the wired and wireless networks, the use of a CB in the BS seems to be the most effective synchronization mechanism and, at the same time, the easiest to implement. For this reason in the next section we will investigate intra/intermedia synchronization achieved by means of a CB in the BS, and will show how the synchronization mechanism parameters (i.e. the buffer size and the delay to be introduced on the first IU inserted into the buffer) affect the P_QoS requirements expressed by relations (1)–(6). As we intend to focus on the CB in the BS, we will assume that this is the only synchronization recovery mechanism in the network.

V. Analysis of a synchronization technique based on compensation buffers in the Base Station

In order to explain the symbols used in the rest of the section, let us consider the segment of the reference communication system from the i-th monomedia source to the MT when this is within the generic L-th cell, as shown in Fig. 2, based on the reference scenario described in Section III.

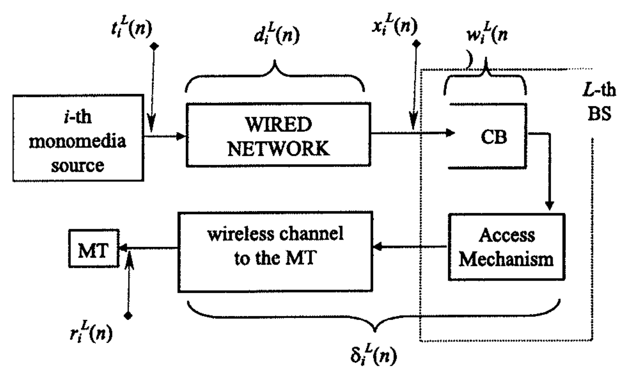

Figure 2: Segment of the reference communication system from the i-th monomedia source to the MT

Let us indicate:

- $t_i^L(n)$ as the time instants at which the n-th IU from the i-th media is sent by the source to the MT, currently placed in the cell of the L-th BS;
- $d_i^L(n)$ as the delay in transmission of the n-th IU from the i-th source to the L-th BS;
- $x_i^L(n)$ as the time instants at which the n-th IU is received at the CB in the L-th BS;
- $w_i^L(n)$ as the waiting time of the n-th IU in the CB in the L-th BS;
- $\delta_i^L(n)$ as the delay of the n-th IU due to the wireless network, from the output of the CB to the input of the MT;
- $r_i^L(n)$ as the time instants at which the n-th IU is delivered to the MT to be processed and displayed to the user.

Let us note that the superscript "L" in the above terms has been used to indicate that the MT is in the L-th cell, whereas the index "n" refers to the sequence number of the IUs received at the MT through the L-th cell; so the index $n = 1$ indicates the first IU received at the L-th BS.

In a multimedia scenario, IUs obviously have to be time-stamped. Without loss of generality, we assume that the time stamp is the time instant at which the IU is transmitted by the source, measured according to the clock reference at the source site. If the n-th IU comprises several ATM cells, the time-stamp can be inserted in the payload of the first of these cells, so all the ATM cells which make up the n-th IU undergo the same delay $w_i^L(n)$ in the CB.

Computation of the waiting time in the CB

In order to compensate for the delay jitter in the wired network, we have to distinguish between two cases: a globally synchronized clock and a locally synchronized clock. If the clock is globally synchronized, i.e. if the absolute time is the same for both the source and the BS, by means of the time stamps in the IUs it is possible for the BS to calculate the delay each IU has undergone in the wired network as the difference between the time instant at which the IU is received by the BS at the CB input and the time stamp contained in the IU. If, on the other hand, the clock is locally synchronized [16][13], i.e. the clock ticks at the source site and at the BS have the same advancement, it is not possible for the BS to measure the delay of each IU. However, from the difference between the time stamps of two consecutive IUs it is possible for the BS to measure the IU interarrival time at the source site and the difference between the network delay of two IUs. Then, by taking the first IU received as a reference, the BS can calculate the wired network delay variation at the reception of the n-th IU with respect to the actual delay of the first IU received, according the following relation:

$$d_i^L(n) - d_i^L(1) = x_i^L(n) - x_i^L(1) - \left(t_i^L(n) - t_i^L(1)\right) \quad (7)$$

Here we will assume the most general case, i.e. the clock is only locally synchronized. Extension to the case of a globally synchronized clock is straightforward.

By considering the definition of residual jitter given in Section III and relation (7), it can be derived that the residual jitter perceived at the MT is given by $j_i^L(n) = d_i^L(n) + w_i^L(n) + \delta_i^L(n) - d_i^L(1) - w_i^L(1) - \delta_i^L(1)$, where we assume that the ideal delivery times by which the intramedia time relationships are recovered are $t_i^L(n) + d_i^L(1) + w_i^L(1) + \delta_i^L(1)$, as intramedia synchronization is maintained if all the IUs undergo the same end-to-end delay of the first IU.

As in this paper we are focusing our attention on CB sizing with the aim of equalizing the wired network delay, and considering that in many wireless systems contention on access to the wireless medium is during the call set-up phase only, we will henceforward assume that delay variation on the wireless channel is negligible when an MT remains in the same cell. The residual jitter is therefore given by:

$$j_i^L(n) = d_i^L(n) + w_i^L(n) - d_i^L(1) - w_i^L(1) \quad (8)$$

In order to calculate the waiting time of an IU in the CB, we have to decide a strategy for the scheduling of IUs in the CB. A possible strategy is to consider the CB as a shift register which is scheduled in such a way as to minimize the absolute residual jitter value of each IU of each media, once the delay introduced for the first IU, $w_i^L(1)$, has been fixed and considering, obviously, only the wired network delay suffered by each IU. In this case the waiting time of each IU in the CB is given by the relation:

$$w_i^L(n) = \begin{cases} 0 \\ \quad \text{if} \quad d_i^L(n) > w_i^L(1) + d_i^L(1) \\ \\ w_i^L(1) + d_i^L(1) - d_i^L(n) \\ \quad \text{if} \quad 0 \le w_i^L(1) + d_i^L(1) - d_i^L(n) \le \Delta_i^L \\ \\ \Delta_i^L \\ \quad \text{if} \quad d_i^L(n) < w_i^L(1) + d_i^L(1) - \Delta_i^L \end{cases} \quad (9)$$

where Δ_i^L is the size of the CB for the i-th media in the L-th BS, expressed as the maximum delay it can introduce in an IU. Fig. 3 shows a pseudo-code which implements the algorithm to calculate the value of $w_i^L(n)$ for each IU received at the CB input.

Computation of the residual jitter and skew pdf's

As the P_QoS requirements in relations (1)–(2) are specified in terms of cumulative probability functions of the residual jitter and skew perceived at the MT, and the residual skew is, in turn, a function of the residual jitter, it is necessary to calculate its pdf. From relations (8) and (9), by indicating as $z_i^L(n)$ the wired network delay measured at the CB with respect to the first IU, that is, $z_i^L(n) = d_i^L(n) - d_i^L(1)$, we obtain:

$$j_i^L(n) = \begin{cases} z_i^L(n) - w_i^L(1) \\ \quad \text{if} \quad z_i^L(n) > w_i^L(1) \\ \\ 0 \\ \quad \text{if} \quad w_i^L(1) - \Delta_i^L \le z_i^L(n) \le w_i^L(1) \\ \\ z_i^L(n) - w_i^L(1) + \Delta_i^L \\ \quad \text{if} \quad z_i^L(n) < w_i^L(1) - \Delta_i^L \end{cases} \quad (10)$$

If we assume that the wired network delay is a random process consisting of a sequence of independent, identically distributed random variables and that

```
wait for IU_i(n)
t_in = actual time
read time stamp t_i^L(n)
if n=1
  set output time t_out(1) = t_in + w_i^L(1)
  set t_ideal(1) = t_out(1)
else
  compute t_ideal(n) = t_ideal(n-1) + t_i^L(n) - t_i^L(n-1)
  if t_in ≥ t_ideal(n)
     output of IU immediately
  else
     set IU output time t_out(n) = min [t_ideal(n), t_in + Δ_i^L]
  endif
endif
```

Figure 3: Pseudo-code of the synchronization algorithm in the compensation buffer. $t_{ideal}(n)$ is the IU output time at which intramedia time relationships should be recovered.

the delay suffered by one monomedia stream is statistically independent of that of another monomedia stream, the pdf's of the residual jitter and skew, $p_{j_i^L}(\xi)$ and $p_{s_{k,i}^L}(\xi)$ respectively, are:

$$p_{j_i^L}(\xi) = \begin{cases} p_{d_i^L}(w_i^L(1)+\xi) * p_{d_i^L}(-w_i^L(1)-\xi) & \text{if} \quad \xi > 0 \\ \int_{w_i^L(1)-\Delta_i^L}^{w_i^L(1)} p_{j_i^L}(\xi) * p_{j_i^L}(-\xi) \cdot d\xi & \text{if} \quad \xi = 0 \\ p_{d_i^L}(w_i^L(1)+\xi-\Delta_i^L) * p_{d_i^L}(\Delta_i^L - w_i^L(1) - \xi) & \text{if} \quad \xi < 0 \end{cases} \quad (11)$$

$$p_{s_{k,i}^L}(\xi) = p_{j_k^L}(\xi - \overline{d_k^L} + \overline{d_i^L}) * p_{j_i^L}(-\xi + \overline{d_k^L} - \overline{d_i^L}) \quad (12)$$

where '$*$' indicates convolution.

For the sake of illustration Figs. 4 show some numerical examples of pdf's of $p_{j_i^L}(\xi)$ and $p_{s_{k,i}^L}(\xi)$, and the complementary cumulative probability functions by means of which P_QoS requirements are specified. By observing relations (11) and (12) and the figures, the effect of the CB, applied to both the master and slave streams, is evident. Moreover, by comparing Figs. 4e) and 4f), it can be noted that by choosing an appropriate value for the delay introduced on the first IU, $w_i^L(1)$, the average value of the skew perceived at the MT can be reduced to zero.

By using the above IU scheduling algorithm, the range of delay values which are compensated for by the CB as a whole is an interval that depends on both $w_i^L(1)$ and Δ_i^L. So, for the whole wired network delay to be compensated for, we must have $\Delta_i^L \geq 2 \cdot (max[d_i^L] - min[d_i^L]$. As far as the residual skew is concerned, let us note that, as it is not possible to estimate the delay for the first IUs, even though the residual jitter can be eliminated, the residual skew cannot, as it is not possible to estimate $d_k^L(1) - d_i^L(1)$.

Computation of the pdf of the time displacement

Once the pdfs of the residual jitter and skew are calculated, we calculate the pdf of $D_i^{M,L}$, the time displacement in the IU delivery time when an MT passes from one cell to another. Let us assume that no IUs are lost when an MT moves from the *L*-th BS to the *M*-th BS, and let us consider two subsequent IUs: the first (which, from the point of view of the *L*-th BS, is the *n*-th IU), being delivered to the MT through the old BS and the second (which, from the point of view of the *M*-th BS, is the first received IU), through the new BS. According to the definition given in Section 3 and relation (8) we have:

$$\begin{aligned} D_i^{M,L} &= d_i^M(1) + w_i^M(1) + \delta_i^M(1) - \\ &\quad j_i^L(n) - d_i^L(1) - w_i^L(1) - \delta_i^L(n) \end{aligned} \quad (13)$$

If we again assume all the delays to be random processes consisting of a sequence of independent, identically distributed random variables and the delay in the wired and wireless networks to be independent of each other, the pdf of $D_i^{M,L}$ is:

$$\begin{aligned} p_{D_i^{M,L}}(\xi) &= p_{d_i^M}(\xi_1) * p_{\delta_i^M}(\xi_1) * p_{j_i^L}(-\xi_1) * \\ &\quad p_{d_i^L}(-\xi_1) * p_{\delta_i^L}(-\xi_1) \end{aligned} \quad (14)$$

where $\xi_1 = \xi - w_i^M(1) + w_i^L(1)$.

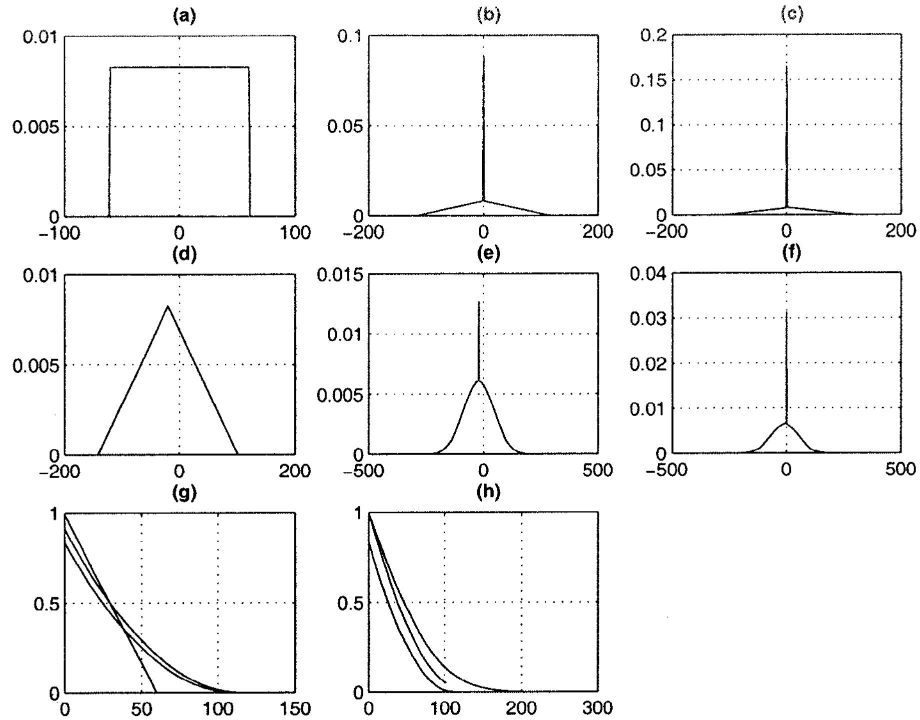

Figure 4: An example of jitter and skew pdf's at the output of BS – (a) $p_{j_i^L}(\xi)$ when no CB mechanism is applied – (b) $p_{j_i^L}(\xi)$ when $\Delta_i^L = 10$ and $w_i^L(1) = 0.5 \cdot \Delta_i^L$ – (c) $p_{j_i^L}(\xi)$ when $\Delta_i^L = 20$ and $w_i^L(1) = 5$ – (d) $p_{s_{k,i}^L}(\xi)$ when no CB mechanism is applied – (e) $p_{s_{k,i}^L}(\xi)$ when $\Delta_i^L = 10$ and $w_i^L(1) = 0.5 \cdot \Delta_i^L$ – (f) $p_{s_{k,i}^L}(\xi)$ when $\Delta_i^L = 20$ and $w_i^L(1) = 5$ – (g) complementary cumulative probability function of absolute value of the jitter – (h) complementary cumulative probability function of absolute value of the skew. – Network parameters: $d_i^L(n)$ and $d_k^L(n)$ uniform in the range [140, 260] and [120, 240].

For the sake of illustration Fig. 5 shows an example of $D_i^{M,L}$ pdf. By comparing the curves it can be noted that the CB reduces the variance of $D_i^{M,L}$ but, at the same time, makes a shift in its average value. Moreover, the absolute value of this average value can be reduced by choosing suitable values for the delays introduced on the first IU of each media.

Computation of the number of IUs lost

Finally, let us calculate the number of IUs lost when a handover occurs. Obviously, it depends on how the handover is managed. Here we will refer to two possible scenarios:

1. *hard handover*. The handover is instantaneous, and there is no recovery of the IUs in the CB of the old BS or those already routed towards the old BS;

2. *soft handover with double IU routing*. The handover takes place in a finite time, indicated as t_h, during which the IUs are routed from the source to both the old BS and the new one. During the handover there is no updating of the state between the CBs of the two BSs. The cells have a partially overlapping radio cover and the MT may receive duplicated IUs, which will require appropriate management.

Let us refer to the scheme of the communication system shown in Fig. 2). As the aim of this section is to highlight the IU loss due to the CB-based synchronization mechanism, we consider as lost only IUs emitted by the source and not yet delivered to the mechanism providing access to the wireless channel when an MT passes into a new cell. We will therefore not calculate the loss of IUs queued for access to the wireless channel.

It is clear that IU loss is related to the time at which the handover occurs. To evaluate the phenomenon in the worst case, we will assume that the handover takes place at a time just after the instant at which an IU is emitted by the source. This assumption is a pessimistic one for the IU loss probability, as at least one IU is between the source and the BS when a handover occurs, and therefore at least one IU is lost in the case of a hard handover.

Let us assume that the MT is in the L-th cell and calculate the probability, $P_i^L(n, a, t_h)$, that the

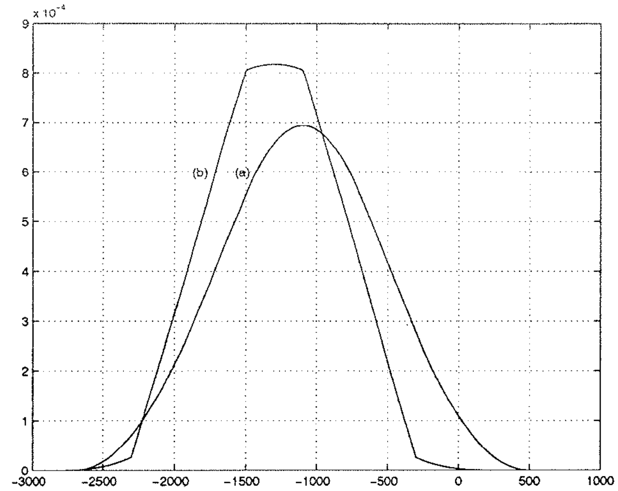

Figure 5: pdf of $D_i^{M,L}$ when: (a) no CB mechanism is applied – (b) CB mechanism is applied and residual jitter is null. – Network parameters: $d_i^L(n)$ and $d_i^M(n)$ uniform in the range $[1400, 2600]$ and $[500, 1300]$ respectively; $\Delta_i^L = 1200$; $\Delta_i^R = 800$; $w_i^L(1) = 600$; $w_i^R(1) = 400$

$\mathrm{IU}_i(n-a)$, with $a \in [0, n-1]$, is not delivered to the access mechanism of the wireless channel at a time t_h after the time instant at which the n-th IU is emitted by the source, $t_i^L(n)$. This is the probability that the time instant at which the $\mathrm{IU}_i(n-a)$ is placed at the output of the CB is greater than or equal to $t_i^L(n)+t_h$, that is:

$$P_i^L(n,a,t_h) = prob[d_i^L(1) + w_i^L(1) + j_i^L(n-a) + t_i^L(n-a) \geq t_i^L(n) + t_h] \quad (15)$$

If both the source interarrival times and jitter are independent, identically distributed random variables, the above relation can be rewritten as:

$$P_i^L(a,t_h) = prob[d_i^L(1) + w_i^L(1) + j_i^L(n) \geq t_i^L(a) + t_h] \quad (16)$$

As $P_i^L(a,t_h)$ is also the probability that the number of IUs emitted by the i-th source and not delivered to the access mechanism of the wireless channel after a time t_h is equal to or greater than "$(a+1)$", the probability $prob\{n_i > N_i\}$ that the number of IUs of the i-th media lost is greater than N_i when a handover occurs just after the source emission time of an IU and its duration is t_h, is:

$$\begin{aligned} prob\{n_i > N_i\} &= P_i^L(N_i,t_h) \\ &= prob[d_i^L(1) + w_i^L(1) + j_i^L(n) \\ &\quad \geq t_i^L(N_i) + t_h] \end{aligned} \quad (17)$$

The above relation is valid for both hard and soft handover. The first case, in fact, corresponds to $t_h = 0$.

By assuming that the wired network delay and source interarrival times pdf's are known, the probabilities in relation (17) can be derived once the compensation buffer size and waiting delay of the first IU have been fixed. For the sake of illustration Fig. 6 shows an example of $prob\{n_i > N_i\}$ by varying the value of t_h.

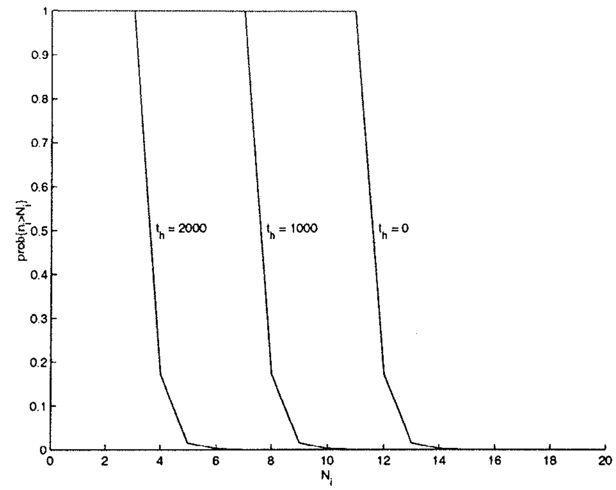

Figure 6: An example of $prob\{n_i > N_i\}$. – Network parameters: $d_i^L(n)$ uniform in the range $[1400, 2600]$; $\Delta_i^L = 1200$; $w_i^L(1) = 600$;

In the case of soft handover, to avoid IU loss, the L-th cell has to be visited at least up to the time instant at which the n-th IU is placed at the output of the CB. For this to hold for every "n" the minimum value of the handover duration is:

$$t_{h,zeroloss} = max[d_i^L(1) + w_i^L(1) + j_i^L(n)] \quad (18)$$

By considering relation (10) we have:

$$t_{h,zeroloss} = 2 \cdot max[d_i^L(n)] - min[d_i^L(n)] \quad (19)$$

If $t_h < t_{h,zeroloss}$ IU loss may occur.

VI. Application of the synchronization technique

As shown in Section III, and, in particular in relations (1)–(6), the P_QoS can be specified in terms of a set of statistical bounds on probability functions. If a CB buffer is used to equalize network delay and the synchronization algorithm in Fig. 3 is used, these probability functions, given by relations (11), (12), (14) and (17), depend on the wired and wireless network delays, the residual jitter and the emission process of the monomedia source. From the CB point of view, they depend on the size of the CB and the delay introduced in the first IU, which determine the pdf of the residual jitter. Sizing the parameters of the synchronization technique therefore means determining, for each cell visited by the MT:

- which monomedia stream is to take on the role of the master stream;
- the size of the CB, Δ_i^L, assigned to each monomedia stream;
- the delay, $w_i^L(1)$, introduced on the first IU addressed to the BS.

As far as the first point is concerned, the media to be chosen as the master stream, in the scenario assumed here, depend on the network delay the various media are affected by. If, in fact, it is considered that the main principle of the CB is to introduce a variable delay on the information streams in order to equalize the wired network delay, and that this delay may undergo sharp variations when the MT makes a handover, to make the synchronization mechanism efficient the master has to be the media that on average undergoes the least delay. The media taking on the role of master may therefore change from one cell to another.

At this point sizing the synchronization mechanism means determining, in each cell visited by the MT, for the master and each slave, the CB size and the delay affecting the first IU addressed to the BS in such a way as to satisfy relations (1)–(6) for the set of P_QoS parameters specified by the user. The BS obviously needs to know the statistical properties of the delay on both the wired and wireless network and those of the residual jitter in the MT's original cell. To achieve this we can envisage the presence of mobile agents throughout the wired and wireless network, whose task is to perform statistical measurements and send reports to the various BSs whenever a handover towards a new BS takes place.

The basic problem is that relations (1)–(6) are not direct functions of Δ_i^L and $w_i^L(1)$, but of the pdf of the residual jitter, which can only be determined once Δ_i^L and $w_i^L(1)$ have been set. It is therefore no simple task to size the CB on the basis of all the possible values which allow the P_QoS parameters to be met, as it requires considerable calculation power. Fig. 7 is a pseudocode algorithm for a complete search for all the possible buffer sizes and first IU delays that meet the P_QoS parameters, with reference to hard handover (it can easily be extended to cover soft handover). It is a complete algorithm in the sense that, with the exception of the step in which all the buffer sizes and initial delays in the various cycles are scanned, all the possible CB size configurations and first IU delays are evaluated.

To prevent the calculation from becoming prohibitive in terms of the amount of time required, the step in which the "for" cycles are evaluated obviously has to be sufficiently large. When it is not possible to find a solution in the algorithm in Fig. 7, the P_QoS parameters are re-negotiated. If, on the other hand, various solutions are ultimately possible, the configuration chosen is the one which costs less (e.g. the one requiring the smallest buffer size in bytes, thus also taking the size of the IUs into account). If the range of wired network delays for the master and slaves is notable different, the complete search may be so complex as to make the proposed mechanism practically inapplicable. It will therefore be necessary to introduce a number of simplifications that will reduce the calculation complexity of the search for the buffer size and delay value that will meet the P_QoS parameters. One solution is to set the first IU delay before calculating the residual jitter. This can be done by reducing to a minimum the absolute value of the average time displacement following a handover, as determined in the previous section. Another solution is to choose a single buffer size value for the master, so as to limit the search to all the possible slave configurations.

VII. Conclusions

In the past few years the problem of multimedia stream synchronization has been widely studied with reference to wired networks, but only recently has the problem been dealt with in the mobile wireless communication environment. In this paper we have conducted an initial study of the impact of terminal mobility on some of the synchronization mechanisms used in wired networks. On the basis of this study we have provided some guidelines for the implementation of an architecture for multimedia synchronization in a mobile wireless environment, and we have extended the definition of user-perceived Quality of Service so as to take into account the degradation in intra/intermedia synchronization due to terminal mobility. We have then focused on the use of compensation buffers at the interface nodes between the wired and wireless access networks, to recover the intra/intermedia time relationships. Finally, we have derived analytical relations which make it possible to relate the synchronization mechanism parameters to the P_QoS requirements. To perform this analysis we hypothesised a mechanism to schedule the IUs in the compensation buffer and two possible handover mechanisms. Obviously the choices made are not the only ones possible, but the line of reasoning followed in the report can also be applied to other cases as long as it is possible to calculate the pdf of the residual jit-

```
for Δ^L_master = 0 : 2 · (max[d^L_master] − min[d^L_master]
  for w^L_master(1) = 0 : Δ^L_master
    compute pdfs from relations (11) and (14) and loss probability from relation (17)
    verify if relations (1), (3) and (5) are satisfied
  endfor
endfor
let V_master be the set of pair [Δ^L_master, w^L_master(1)] which satisfies P_QoS parameters on master stream
if V_master = {∅}
  renegotiate P_QoS parameters on master stream
endif
for Δ^L_slave = 0 : 2 · (max[d^L_slave] − min[d^L_slave]
  for w^L_slave(1) = 0 : Δ^L_slave
    for all pairs in V_master
      compute pdfs from relations (11), (12) and (14) and loss probability from relation (17)
      verify if relations (2), (4) and (6) are satisfied
    endfor
  endfor
endfor
let V the set of elements [Δ^L_master, w^L_master(1), Δ^L_slave, w^L_slave(1)] which satisfies P_QoS parameters
if V = {∅}
  renegotiate P_QoS parameters on slave stream
endif
if size(V) > 4
  select the minimum cost solution
endif
```

Figure 7: Algorithm, in pseudo-code, to search for all the CB size and first IU delay values that will meet the P_QoS parameters expressed in relations (1)–(6)

ter.

The work reported on is only an initial approach to the problem of synchronization by means of compensation buffers. Moreover our study is based on a reference communication scenario where all the monomedia streams are transmitted to the MT through the same BS. The paper therefore does not take into account the possibility of macro-diversity where different monomedia components of a multimedia service can be transmitted from different BS each supporting different cell-sizes and different delays in the wireless channel. In any case, the use of compensation buffers in each BS helps to solve the synchronization problem, as it reduce the end-to-end delay variation suffered by each monomedia stream. In this case the part of the proposed synchronization algorithm which refers to the master stream can be applied to each monomedia stream to guarantee intramedia synchronization on each stream. The authors are currently investigating various CB scheduling algorithms, how to schedule CB when macro-diversity is taken in account and how to define P_QoS parameters which will take into account the effect on user-perceived quality and at the same time allow for fast, simple calculation.

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List of acronyms

BS	Base Station
CB	Compensation Buffer
IU	Information Unit
MT	Mobile Terminal
pdf	probability density function
QoS, P_QoS	Quality of Service , user-Perceived Quality of Service

List of symbols

$d_i^L(n)$ the network delay the n-th IU of the i-th media undergoes, from the source to the MT through the L-th BS

$\overline{d_i^L}$ the mean value of $d_i^L(n)$

$D_i^{M,L}$ the time displacement in the IU delivery times at the MT when it hands over from the L-th cell to the M-th cell. It is the difference between the end-to-end delay of the first IU received by the MT through the M-th BS and that of the last IU received by the MT through the L-th BS

$\delta_i^L(n)$ the delay the n-th IU of the i-th media undergoes in the wireless network, from the output of the CB placed in the L-th BS to the MT

$j_i^L(n)$ the *residual jitter* of the n-th IU of the i-th media which is transmitted to the MT through the L-th BS. It is the difference between the actual delivery time at the MT and the ideal delivery time by which the intramedia time relationship is recovered.

$J_{master,min}$, $J_{master,max}$, ε_{master}, $S_{slave,min}$, $S_{slave,max}$, ε_{slave}, $D_{master,min}$, $D_{master,max}$, $\varepsilon_{master,D}$, $D_{slave,min}$, $D_{slave,max}$, $\varepsilon_{slave,D}$, N_{master}, N_{slave}, $\varepsilon_{master,h}$ and $\varepsilon_{slave,h}$ are the P_QoS parameters, that is, the set of values which specify the perceived quality of service requested by the user by means of relations (1)–(6)

$p_{D_i^{M,L}}(\xi)$ the pdf of $D_i^{M,L}$

$p_{j_i^L}(\xi)$ the pdf of $j_i^L(n)$

$p_{s_{k,i}^L}(\xi)$ the pdf of $s_{k,i}^L(n)$

$r_i^L(n)$ the time instants at which the IUs from the i-th media are delivered to the MT

$s_{k,i}^L(n)$ the *residual skew* between the n-th IUs of the media "k" and "i" received by the MT through the L-th BS. It is the displacement in time of corresponding IUs after the application of intramedia synchronization mechanisms, that is, $s_{k,i}^L(n) = j_k^L(n) - j_i^L(n) + \overline{d_k^L} - \overline{d_i^L}$

t_h the soft handover duration

$t_i^L(n)$ the time instants at which the IUs from the i-th media are sent by the source to the MT through the L-th BS.

$x_i^L(n)$ the time instants at which the IUs from the i-th media are received at the input of the CB in the L-th BS

$w_i^L(n)$ the waiting time of the n-th IU of the i-th media in the CB placed in the L-th BS

$z_i^L(n)$ the wired network delay of the n-th IU of the i-th media which is measured by the BS when the clock is locally synchronized, that is, $z_i^L(n) = d_i^L(n) - d_i^L(1)$