Data Link Level Support for Handoff in Wireless ATM Network

Sueng-Yong Park, Vaduvur Bharghavan, Sung-Mo Kang
University of Illinois at Urbana-Champaign
Department of Electrical and Computer Engineering
405 N. Mathews Ave., Urbana, IL 61801
(Tel) (217) 244 - 4492
(Fax) (217) 244 - 8371

Abstract

Handoffs enable seamless migration of virtual connections between cells as users move in an indoor wireless ATM environment. Most current handoff schemes in such environments operate at the network layer, without any explicit data link layer support for reducing cell loss during handoffs. In this paper, we present a new data link layer mechanism that will minimize the delay and delay variation due to handoffs. Our handoff scheme inserts specially marked cells in the data cell flow. Analysis and simulation of our approach show a reduction in delay by 50%, and reduction in delay variance by an order of magnitude.

1. Introduction

In recent years, there have been a number of research proposals for integrating indoor wireless networking technologies with ATM backbone networks for the purpose of enabling indoor user mobility [1][3]. As users move between cells, one key issue in mobility support over ATM networks is the dynamic re-establishment of virtual connections with minimal cell loss [2]. In the Internet domain, loss of a few datagrams during handoff can be tolerated at the network layer because the onus of reliable and sequenced delivery of packets is at the transport or higher layers. However, in ATM networks, loss of cells during handoffs is a more serious issue that needs to be resolved at the virtual connection level. Most of the approaches that currently support seamless handoff in wireless ATM networks propose network-layer solutions for dealing with transient cell loss during handoffs. While handling recovery from cell loss at the link-layer would reduce the protocol overhead and consequently the handoff delay, the absence of sequence numbers in ATM cells makes loss recovery a non-trivial problem at the link-layer. In this paper, we propose a simple link-layer mechanism to handle the cell loss recovery at the link layer. Our approach does not assume any particular network-layer handoff approach, and can co-exist with several published handoff mechanisms such as VC re-establishment [3], VC partial re-establishment [4], and VC multicasting [5]. Our approach is to insert special *marking cells* in ATM cell flows. Marking cells contain sequence numbers, and thus provide well defined checkpoints at which cell loss recovery can be handled. As shown from our performance analysis, using the marking cell approach for link-layer cell loss recovery can achieve a significant improvement in the worst-case, average-case and variance in the handoff latency over a purely network-layer solution. The rest of the paper is organized as follows. Section 2 describes the link-layer marking cell algorithm. Section 3 presents a simulation-based performance analysis of the algorithm, and Section 4 concludes the paper.

2. Marking Cell Algorithm

In this section, we first describe the steps in a generic network layer handoff algorithm and motivate the need for link-layer support in Section 2.1, and then describe the marking cell algorithm in Sections 2.2 and 2.3.

2.1 Handoff

In order to provide reliable data delivery at the network layer in connection-oriented networks, handoffs must prevent or recover from packet loss. Typically, this is handled as follows: when a mobile host (MH) enters a new cell, it greets the new base station (BS) with the sequence number (S) for the (network-layer) packet it should receive next. Depending on the handoff algorithm, BS will receive the packets starting with sequence number S by re-establishing (fully or partially) the network layer connection with the source, or by requesting the old base station to forward packets in transit, or joining a multicast group which enables all potential next-cells of MH to receive packets destined for MH in advance of the actual handoff. The point to note is that irrespective of the actual handoff algorithm used, reliable deliv-

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ery is performed at the packet level rather than the cell level because cells do not have sequence numbers. In particular, when MH moves into the domain of BS, it can identify the next packet it expects to receive by the sequence number S, but not the next cell it expects to receive. Thus, BS must reassemble the received ATM cells in the AAL5 layer and forward it to the network layer for sequence number identification. If any cell is missing in the packet, the whole packet is dropped. In the next section, we show that by adding a very simple mechanism, we can covert the granularity of loss recovery from the packet level to the cell level.

2.2 Marking Cell Approach

Our solution to the problems described in section 2.1 is to place a marking cell when the data packet is segmented. The marking cell is categorized as an Operation & Management (O&M) cell, bearing a sequence number which is unique during the higher layer protocol assigned timeout interval of the connection. In the traffic source side, there are a sequence counter, S, and an interval counter, IC. S increments by one whenever the traffic source transmits a marking cell. The marking cell stores S value in the payload. IC counts the number of ATM cells between the transmission of two marking cells. If IC reaches a certain limit, the data link layer protocol inserts a marking cell in the cell stream. After the insertion, IC is reset to 0 and counts again. The receiving node keeps track of S value of the marking cell. When a new marking cell arrives, the receiver records the buffer address of the marking cell and the S value. Marking cell approach can be added to both partial VC re-establishment and VC multicasting approaches without changing the original algorithms because of its independence from the higher layer protocol layers.

2.3 Protocol description

In this section, we describes our data link layer handoff approach.

2.3.1 Sender

The sender has one register and two counters: {Interval Counter Register (ICR), Interval Counter (IC), Sequence Counter (S)}. ICR stores the maximum number of ATM cells transmitted between two marking cells. IC counts the number of data ATM cells transmitted after the insertion of marking cell. S is a 16-bit counter, incrementing by one when a marking cell is inserted in the traffic. The inserted marking ATM cell carries the value of S.

2.3.2 Base station (BS)

BS assigns a certain portion of addressable FIFO buffer to each MH. A linked list keeps management information of the FIFO queue. Every cell in the linked list stores the FIFO address of the mark cell and its S value. For the simulation, there are 128 ATM data cells between successive marking cells. (This number is controlled by IC in sending node.) The marking cell is not transmitted to MH, instead, when BS finds a marking cell in FIFO queue, it just announces a S value to MH, then transmits the next available ordinary ATM cell.

2.3.3 Receiver

The receiver stores the FIFO buffer addresses of the marking cells and their S value in the linked list. The Figure 1 presents pseudo code for the VC multicast based handoff mechanism with the marking cell approach in the data link layer.

```
MH greets a new BS with MH_GREET(MH_S, MH_IC, System Specific Data)
BS reads the greet
BS:  if MH_S >= BS_LIST(1)_S and MH_S < BS_LIST(N)_S{
         S_index = MH_S - BS_LIST(1)_S + 1;
         if MH_S!= BS_LIST(S_index)_S
             return (BAD_DATA);
         cell_index = BS_LIST(S_index)_BufAddr$_{markcell}$ + MH_IC;
         Flush FIFO queue between 1 ~ cell_index - 1
         Transmit Fifo(Cell_index) at Next Transmission Time
     }
     else if MH_S <  BS_LIST(1)_S {
         Cells are lost. Activate a higher layer protocol
         Notify MH
     }
     else if MH_S > BS_LIST(N)_S{
         Cells are not arrived yet.
         Wait
     }
}
```

FIGURE 1. Algorithms for handoff

3. Analysis and Simulation

This section defines and derives the parameters used in the simulation. The simulations are done for the simulated FTP trace [6] and an MPEG [7]. MH is assigned a fixed air bandwidth. BS transmits data by the request of MH.

3.1 Definition of parameters

We assume that BS receives traffic with Quality of Service (QoS) guarantees for delay and jitter. The raw wireless bandwidth is at maximum 650 kbps, which is enough to transfer a bursty MPEG traffic within a certain delays. A wireline connection is provided by DS 3 link at the speed of 44.767 Mbps. For FTP traffic, we assume that the traffic reserved the average 650 kbps bandwidth. We also assume that the interarrivial time of Maximum Transfer Unit (MTU) packets is normally distributed. Table 1. shows the other parameters used in the simulation. FIGURE 2 is the our model of handoff procedure. For MPEG traffic, the long time average of traffic is 450 kbps and the traffic arrives according to their real time traces. The arrival is very bursty for some duration, as shown in FIGURE 5.

TABLE 1. Definition of Parameters

Parameter	Definition	Value
BW_{wl}	Reserved wireless bandwidth	650 kbps
MH_GREET	Average time for a MH to greet a new BS	5 ms
MH_ACC_{wl}	Minimum time required for MH to access a base station	1 ms
$Jitter_w$	Traffic jitter in wireline network	5 ms
POH_{BS}	Minimum protocol overhead in the base station	1 ms
POH_{MH}	Minimum Protocol overhead in the mobile host	1 ms
$Cellsz_{ATM}$	Size of ATM cell	53 bytes
$Packet_{control}$	Size of ATM cell	64 bytes
$Packet_{data}$	Size of wireless packet	64 bytes

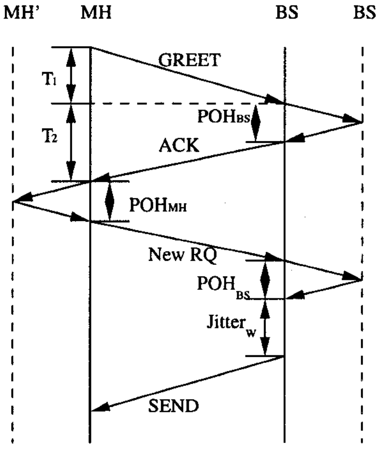

FIGURE 2. Model for a Handoff

3.2 Derivation of Metrics

We derive the metrics for the performance based on the definition in 3.1. The derivation shows how to obtain the buffer size at BS and an empirical IC value for the traffic source. We assumed a reliable data link layer protocol between BS and MH. The handoff method is VC multicasting. We assume that the cells are multicast to both the previous cell and the new cell.

T_1, the time required to send the greet message to the base station and to process at BS, is the sum of wireless channel access time, greet message transmission time, and the bound protocol processing time on MH.

$$T_1 = MH\_GREET + (Packet_{control} * 8 / BW_{wl}) + POH_{MH} \quad (1)$$

T_2, the time required for the base station to acknowledge the greeting from mobile host, is the sum of ACK packet transmission time, and the processing time in BS.

$$T_2 = (Packet_{control} * 8 / BW_{wl}) + POH_{BS} \quad (2)$$

T_3 is the total delay in handoff.

$$T_3 = T_1 + T_2 + POH_{MH} + POH_{BS} + MH\_ACC_{wl} + (Packet_{data} / BW_{wl}) + Jitter_w \quad (3)$$

After the greeting, if the new base station already had the next cells required, the base station immediately transmits the required cell. If it doesn't have the proper cell, the buffer is flushed and higher level protocol handles this error. Therefore the buffer size required for the down link traffic is the size of buffer when base station had a proper cell. The buffer size is the product of total delay and traffic bandwidth.

$$B_{BS} = T_3 * BW_w / 8 \quad (4)$$

The number of ATM cells between two marking cells, IC, depends on the quality of the wireline handoff and rerouting scheme. If there are no loss of ATM cells after the handoff and rerouting, IC can be as big as 2^{16} - 1. If there are many loss of ATM cells, IC value should be reduced to the minimum value to guarantee that there is, at least, one marking cell in the FIFO queue for the identification of ATM cells.

$$IC_{min} = B_{BS} / Cellsz_{ATM} \quad (5)$$

And we regarded the system overhead in handoff, OVs, is the sum of the duration of a packet receiving time and $Jitter_w$. as the system characteristics are amortized in the duration of a packet receiving time, and $Jitter_w$ actually has an effect on the handoff overhead.

$$OV_s = Duration\_Time_{PKT\_RCV} + Jitter_w \quad (6)$$

The actual value is calculated by (1)-(5). The value of the overhead, OVs, is simulated.

$$T_1 = 6.79 \text{ (ms)}$$

$$T_2 = 1.79 \text{ (ms)}$$

$$T_3 = 17.37 \text{ (ms)}$$

$$B_{BS} = 1.41 \text{ (Kbytes)}$$

$$IC_{min} = 26$$

The overhead of the marking cell approach in bandwidth is 3.8% at the worst case. IC value can be increased to reduce the overhead. When the simulation is done with the IC value of 128, there was no unidentifiable marking cell.

3.3 Simulation

3.3.1 Simulation result

We assume that the backbone network is the DS-3 network with raw bandwidth of 44.736 Mbps. One frame of a DS-3 link is 125 us. There are twelve ATM cells per frame. 96000 ATM cells delivered per second. The simulation was done to measure the handoff overhead for the network layer handoff method and the marking cell method in the micro cellular system. The user mobility is modeled with Poisson distribution. Each simulation is performed for 120 handoffs. The traffic source is the simulated FTP trace [6] and an MPEG trace [8]. We have segmented generated FTP packets with the MTU size of 512 bytes and 1024 bytes. As the interarrival time of FTP packet is not modeled, we have assumed the packet interarrival time is normally distributed, not to exceed the reserved bandwidth. In order to test both the network layer approach and the marking cell approach under the same condition, the same random distributions are used for each approach.

The FIGURE 3. is the simulation of FTP trace with 512 byte and 1024 byte MTU sizes. The most notable difference between the data link layer handoff and the network layer handoff is its jitter ratio. The time overhead of the data link layer approach is quite regular and less bursty, while the time overhead of the network layer approach is bigger and bursty.

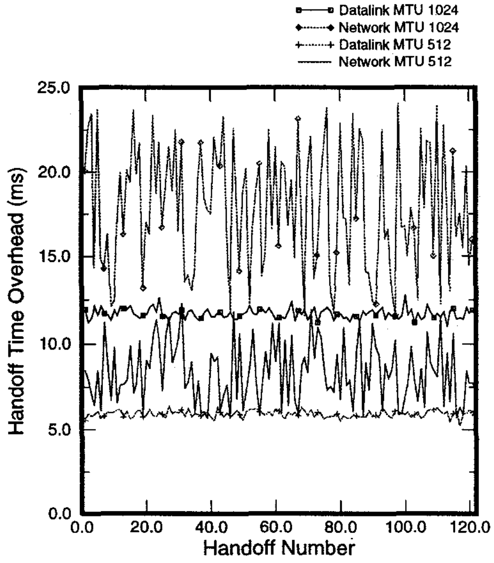

FIGURE 3. Overhead Time Comparison in FTP

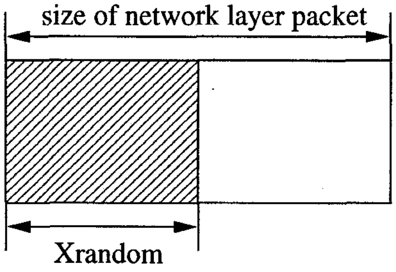

FIGURE 4. Network Layer Receiving Model

The overhead of the network layer approach is worsen as the size of MTU increases. The network layer handoff model in FIGURE 4, explains the relation of the overhead and the MTU size. The X_{random} is a random parameter, indicating how much portion of the currently receiving packet is received. As MH has to receive again X_{random} amount data after the handoff, the X_{random} portion becomes the source of overhead and jitter. Since the average value X_{random} increases in proportion with the MTU size, the FTP trace with 1024-byte MTU size shows a bigger overhead than 512-byte MTU trace in FIGURE 3.

FIGURE 5 is the simulation of the MPEG trace. The network layer unit of MPEG trace is a Convergence Sublayer Protocol Data Unit (CS-PDU). The size of CS-PDU varies extensively according to the content of MPEG movie.

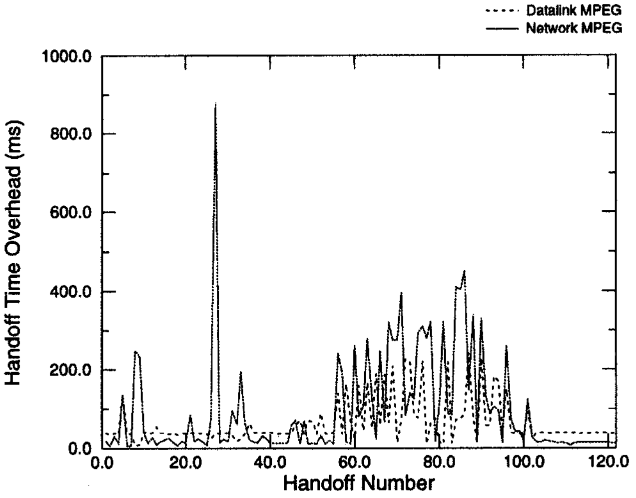

FIGURE 5. Overhead Time Comparison in MPEG

As a network layer unit is varying in sizes, MH would suffer the various lengths of jitters. FIGURE 4 shows that the network level approach sometimes handoff faster than data link level approach. This is because that MH in the network layer support happened to be receiving a small size CS-PDU, while MH in data link layer support was receiving bigger CS-PDU. And in the first 50 or 60 handoffs, the datalink layer approach has a bigger overhead. This is caused by the wireless MAC layer protocol. As we assumed that MH was assigned a fixed amount of bandwidth and that BS transmitted data by the request of MH, MH had to wait a certain amount of time if there is no data in BS. This is an extra overhead in measuring Duration_Time$_{PKT\_RCV}$. The big glitch around 30th handoff of network layer handoff scheme is caused by the late arrival several ATM cells. In the network layer approach, CS-PDU can't be identified until all cells arrive. If even one packet arrives late, whole CS-PDU can't be identified and can't be transmitted, while the data link approach has no such an effect. TABLE 2 is the comparison of the performance in each approach.

TABLE 2. Comparison of Performance Index

Simulation	Mean Delay Overhead	Delay Overhead Variation
Network Layer Appr. (1024 MTU)	17.832 ms	3.646 ms
Marking Cell Appr. (1024 MTU)	11.720 ms	0.291 ms
Network Layer Appr. (512 MTU)	8.352 ms	1.757 ms
Marking Cell Appr. (512 MTU)	5.925 ms	0.204 ms
Network Layer Appr. (MPEG)	93.412 ms	129.33 ms
Marking Cell Appr. (MPEG)	59.027 ms	54.354 ms

TABLE 2 shows that the marking approach can handoff faster than the network layer approach. The marking cell approach also has much less delay overhead variation, which would cause less jitters in MH. The network layer approach has a bigger delay overhead variation. This variation becomes worse when the Network layer packet size varies a lot, as in the case of MPEG traffic. Since we have reserved the enough buffer space at BS, there was no packet loss during the simulation.

4. Conclusion

We have presented a new data link layer handoff mechanism to support fast handoff in the ATM networks. The key observation which motivates our work is the fact the current handoff schemes need the identification of data in order to synchronize or forward the data when the user moves. Since this identification is done at the network layer, there are redundant overheads in the schemes. We also noticed that the lack of sequence number in the header fields of ATM cell does not mean ATM cell is unidentifiable in the ATM layer. We have inserted marking cells in the ATM cell stream and made ATM cells identifiable in the ATM layer. We analyzed and simulated the handoff time for both the network layer and the marking cell approach in the VC multicasting handoff environment. The simulation results show that the marking cell approach can reduce handoff delay by 50% and delay variance by an order of magnitude.

5. References

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