Abstract

This paper describes an ATM network with wireless access for multimedia communications. The network has been specifically designed to serve as a testbed on which evaluate the impact that different handovers techniques have on multimedia traffic. The access network has been completely emulated, where the behaviour of the radio channel is emulated by software and its transmission functions supported by an Ethernet LAN. The paper discusses the design decisions that have been taken, provides an example showing how a handover takes place and provides preliminary results on frame jitter at the mobile terminal and on handover execution times.

Keywords: WATM, Testbed, Handover, Multimedia, Radiochannel.

1. INTRODUCTION

In December 1995 the ATM Forum started working on the definition of the Wireless ATM (WATM) architecture. This work has been carried out in co-operation with the European Telecommunications Standards Institute (ETSI). In particular, ETSI study group RES10 has been working on the definition of the radio interface. ETSI has taken advantage of previous work done for HIPERLAN II (High PErformance Radio LAN), which defined an interface in the 2.5 Ghz band, with an effective bandwidth of 20 Mbit/s and a maximum cell diameter of 50 m.

Two possible scenarios have been defined for the provision of interconnection via the radio interface: Fixed Terminals (FT), that is without mobility support, and Mobile Terminals (MT). Figure 1 shows how the different functional entities are related for the second scenario. As can be observed, specific switching units called End-user Mobility Supporting ATM Switch (EMAS) have been defined for the support of mobility. The EMAS can be of two types: Edge (EMAS-E) which incorporate Radio Ports (RP) and Network (EMAS-N) which do not have RP. The EMAS-E can make use of external traffic concentrators called Access Points (APs) where the RPs can be attached. As can also be observed, the interconnection among the EMAS is provided by wired links. The MTs access the network via the RPs, while the FTs are connected to the EMAS-N.

Figure 1 also shows the protocol architecture that has been defined for WATM. A Wireless Access Layer (WAL) that provides access to the radio interface has been defined for both planes. The WAL includes the Physical (PHY), Medium Access Control (MAC) and Data Link Control (DLC) layers. The MAC layer manages the access to the shared medium in a way such that the user requested QoS can be guaranteed. To achieve this, the MAC layer incorporates a scheduler that allocates the available resources among the contending connections [1]. The DLC provides an error free service for the loss sensitive traffic.

At user plane, the same layers defined for a fixed ATM network have been defined here, but replacing the PHY layer for the WAL.

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At the signalling plane, some additional functionality is required to support mobility. These functions will be included in the new signalling protocols: the PNNI+M at the network to network interface, and the UNI+M at the user to network interface. Two approaches have been proposed for the PNNI+M and the UNI+M definitions: to add new mobility modules to the existing PNNI and UNI, or to redefine the PNNI and the UNI. Both types of approaches are now being the object of an intense debate [2]. The PNNI’ protocol supports the functions defined for a conventional PNNI interface but additionally supports the transport of mobility information [3]. Figure 1 also shows that new protocols like the Access Point Control Protocol (APCP) have been added. The APCP allows an EMAS-E to access the state of the radio resources.

2. DESCRIPTION OF THE TESTBED

Our experimental testbed is composed of a real wired ATM network and an emulated wireless network, as shown in Figure 2a. The wired part is built up from an ATM switch and three workstations. One workstation is a fixed terminal and runs the multimedia application server. The other two workstations emulate two base-stations, that is, two APs as defined in the WATM general configuration. The wireless part is emulated over an Ethernet LAN segment. The two base-stations (APs) are connected through the emulated radio-channel to mobile multimedia terminals. These terminals are emulated by PCs running Linux.

The ATM switch is a Fore ASX-200BX with an aggregated capacity of 2.5 Gbps, four output ports of 155 Mbps, a non-blocking time division multiplexing cross-connect with a delay of 10 microseconds, and output distributed shared memory buffers with capacity for 32768 cells. The workstations are connected to the ATM network through ForeRunner SBA-200E ATM cards at 155 Mbps and with UNI3.1 and AAL5 support.

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The testbed protocol architecture is shown in figure 2b. The following protocol layers have been developed:

- A Radio-Channel emulation layer (RC_emul). This layer is located at the base-station and at the MT. It emulates the behaviour of the radio-channel link, by estimating the transmission conditions as a function of the distance from the MT to the base-station. The distance is computed in real-time according to a predefined trajectory. This information is used to introduce a varying transmission delay, and to provide real-time information about the quality of the radio interface. RC_emul entities use an UDP-socket interface to communicate through the Ethernet network.

- A Network Access Protocol layer (NAP). This layer is present only at the wired part: server and base-stations. It provides the additional functionality required to all those signalling procedures needed for handover, that is, it emulates some of the UNI+M extra features.

- A Synchronisation and HandOver layer (SHO). This layer is present in all network elements except the switches, that is, in the server, in base-stations and in the MT. It implements the handover procedures and guarantees the continuity of the user data flow at the MT.

![Testbed configuration](image)

Figure 2. Testbed network elements and protocol architecture.

At the application layer, we run a free distributed experimental video-conference
application that implements a standard UDP-socket interface.

3. HANDOVER SCHEME

The execution of a handover has two parts: the wired and the wireless. The wired part implements the re-routing of the wired part of the connection. Specifically, the server to the ‘old’ base-station segment must be substituted by a new segment from the server to a ‘target’ base-station. The wireless part implements the setting of a new radio data circuit from the MT to the target base-station, and the removal of the data circuit with the old base-station.

The different alternatives and the solution implemented in the testbed are described below. Handover schemes in both sides have been chosen trying to keep the transmission jitter as low as possible.

The alternatives for the wired part of the handover are:

- Total connection set-up. It consists in setting up a new connection from the server to the target base-station. It does not require special signalling but involves the fixed terminal, which in general, is not desired.

- Nearest common node re-routing [4]. The connection is re-routed only from the ‘nearest common node’ to the involved base-stations. The nearest common node is the node that is common to both connections — the connection before and the connection after the handover —, and that is the closest to both base-stations. This type of re-routing involves the minimum number of nodes. The problem is that it requires specific signalling features, which include the capability of changing information of standing connections at the switches dynamically. This capability is not externally provided by the Fore switch.

- Virtual connection tree [5]. It requires the setting of a multicast connection from the server to all the base-stations in the coverage area. The handover is made by selecting the appropriated leaf of the multicast connection. This is logically the most costly solution, and it would be considered only if none of the other solutions could achieve the jitter requirements.

- Path extension. It consists in the extension of the connection by the inclusion of the target base-station. It is simple to implement, but it can add arbitrary delays to the connection as more handovers are made. Besides, the resulting connections could be far from optimal, and they could even include loops.

- Left initiated joint [6]. The target base-station incorporates itself to the standing connection by adding a leaf to it. During a short period, the connection becomes a multicast one: from the server to both base-stations. When the handover is finished, the old base-station will remove itself from the connection. This method requires the left initiated ‘add leaf’ and ‘drop leaf’ signalling features, which have been included in the UNI 4.0 [7].

On the wireless part, the handover has the following alternatives [8] [9]:

- In regard to how the decision to initiate a handover is made, the following alternatives exist: i) measuring the received signal power. The handover is initiated when the power received from a target base-station is higher than the current power; ii) measuring the received signal power with a threshold. This is identical to the previous one, except that the handover is initiated only when the current power is under a certain threshold; iii) Measuring the received signal power with hysteresis. Now the handover is initiated only when the difference between the power of the signals received from a target base-station and the current one is over a ‘hysteresis gap’. This avoids repeated handovers when the MT progresses on the cell border; d) Measuring the received signal power with hysteresis and threshold. This is similar to the one before, but adding the condition that the signal power has to be under a threshold.

- In regard to how the new radio-channel is set-up, the following alternatives exist: i) hard handover. In this case the MT can only maintain a radio-channel at one time. During a hard handover, there is a time gap during which the MT is waiting for the target radio-channel, and has no connection at all; ii) soft handover. In this case the MT can maintain simultaneously two radio-channels: one with the old base-station and other with the target base-station.

- In regard to who initiates the handover, the following alternatives exist: i) MT-controlled handover. The handover is initiated by the MT, which monitors the strength of the signals received from the base-stations of all neighbour cells; ii) network-controlled handover. The handover is initiated by the current base-station, which keeps track of the power received from the MT. When it is under a certain threshold, it polls to the neighbour base-stations to find out
the best candidate; iii) MT-assisted handover.
The handover is initiated by the current base-station, but is the MT which provides the signal power levels received from the current base-station and from the candidate base-stations.

- In regard to which base-station supports the main part of handover dialogue, the following alternatives exist: i) backward handover. In this case is the old base-station which supports the main part of the handover dialogue; ii) forward handover. In this case is the target base-station which supports the main part of the handover dialogue.

The design decisions taken for our testbed are now summarised:

- On the wired part, we selected the left initiated joint option. This seems to be the better solution for multimedia services, at a reasonable cost. It allows to minimise the jitter, it is not resource wasteful and does not require specific signalling. In our case, we used an ATM switch supporting UNI 3.1 only, so we added some UNI 4.0 features, mainly the capability for the left initiated ‘add leaf’ and ‘drop leaf’ from multicast connections. In the testbed, this functionality is provided by the NAP layer.

- On the wireless part, the handover decision is based on ‘signal power with hysteresis’ criterion because it minimises the number of handovers.

- The radio-channel is set-up by soft handover, which is the best solution for multimedia traffic because it allows to minimise the jitter. The synchronisation of the user traffic is done at the MT. This simplifies the implementation because it does not require that the target base-station stores any user data during handover.

- The handover process is initiated by the MT because this the technique that requires less dialogue, so it reduces the overall handover time and signalling load.

- The handover protocol has been implemented as a forward handover. During a handover the MT contacts the target base-station to make it aware of its decision. From this point on the target base-station controls the handover at the wired part. When this wired part process ends, the MT contacts the old base-station to indicate that it can now initiate a leaf drop from the connection. As can be seen, the handover can be implemented without dialogue between both base-stations.

With these options, the overall handover procedure is as follows:

1. The MT keeps track of the power received form the current base-station and from its neighbours.
2. When the hysteresis condition is fulfilled, the MT sets a radio-channel with the target base-station and request a handover sending the necessary information.
3. The target base-station requests its addition to the data connection as a leaf.
4. When the leaf is set, the target base-station starts to send data to the MT.
5. The MT starts to receive data simultaneously from both base stations.
6. When the data flow coming from the target base-station overlaps the data flow coming from the old base-station, the MT requests to the old base-station to initiate a leaf drop from the connection.
7. The old base-station initiates a leaf drop from the connection. When the leaf is dropped, the handover is finished.

4. RADIO-CHANNEL EMULATION

The radio-channel is emulated by adding a protocol layer, which adds the delays caused by propagation and retransmission at the DLC radio layer. The parameters used to calculate these delays are obtained from a propagation and loss model and from the distance to the base-station. The MT maintains the distance to the base-stations by following a virtual user-defined trajectory. The power values estimated are also used to generate handover requests.

The total delay of the radio-channel is the sum of three components:

- The transmission delay. It is equal to the frame size divided by the bit rate, plus the propagation delay. Because the propagation delay is negligible for normal size cells, the transmission delay depends on the frame size only.

- The DLC delay. It depends on the number of retransmissions due to errors, and it is a random value. The bit error probability is derived from the loss model. Knowing the bit error probability and the characteristics of the error-correcting code used by the DLC, the frame loss probability can be calculated. For each frame, the number of retransmissions is generated according to a probability function derived from the frame loss probability. The RC_emul layer retains each frame according to the number of retransmissions computed for it. When a frame is retained, any following frames must wait until
the first one is delivered, this guarantees that they will be delivered in order.

Radio-channel losses are estimated by the Walfish-Bertony propagation model for urban environments [10]. To allow soft handover, the radio-channel should implement spread spectrum with Code Division Multiple Access (CDMA). Therefore, we have used a CDMA model for the calculation of transmission parameters of the radio-channel emulation. We have assumed a direct-sequence BPSK spread spectrum, so the bit error probability can be obtained from the Rappaport expression [11].

- The MAC delay. This is a random delay that is applied only to those connections that have not reserved bandwidth at the radio interface, which applies only to the signalling traffic. This delay has effect during the handover only.

5. PRELIMINARY RESULTS

The system described above has been tested with different types of traffic, including a real-time video application. To illustrate the effect of the radio-channel and handovers on the end-to-end connections, we present in this section some measures of the frame jitter of the traffic received by the MT. Because the traffic pattern generated by a video application is very random, the jitter measurements shown here correspond to a synthetic traffic pattern. In this experiment the server sent 2 Kbyte frames every 20 ms., which corresponds to 1.2 Mbps approximately. During the experiment the mobile executed several handovers.

Figure 3a shows the inter-arrival time of a 15,000 frames sequence. The line at 20 ms. shows that most of the frames arrived with no jitter. The other visible lines above and below the 20 ms. line indicate inter-arrival times of 20±3, 20±12 and 20±15 ms. These values include the impact of the data synchronisation algorithm and the most probable combinations of radio-channel retransmissions. Inter-arrival times below 20 ms. are caused mainly by the clustering effect produced by the DLC operation, which retains all frames when a previous one is being retransmitted. During handover, the synchronisation algorithm can also contribute to the frame jitter but its contribution is negligible. There are also some scattered points at different times, which are due to random effects, like message processing time and operating system scheduling delays.

Figure 3b shows a histogram of the inter-arrival times. This plot has been zoomed, clipping the main peak at 20 ms., making more visible the rest of the peaks. The peaks at 32 ms. — and their symmetric ones at 8 ms. — are due to frame retransmissions at the radio interface. A frame retransmission takes about 12 ms.: a frame transmission time plus twice the radio-channel propagation delay. Peaks at 21, 22 and 23 ms. are due mainly to the operating system scheduling time granularity.
Although the soft handover, as it is implemented, has almost no effect on the traffic, its duration is important for network resources dimensioning. Figure 4 plots the duration of 25 handovers executed during the experiment described above. The total handover time is the sum of four components:

1. T1 is the set-up time of a control radio-channel between the MT and the target base-station.
2. T2 is the set-up time of the new data radio-channel.
3. T3 is the re-routing time at the fixed network.
4. T4 is the time it takes to synchronise the data flow.

The average of each of these components for 25 handovers are also plotted. The total handover time is about 213 ms., which is similar to the one obtained for a DECT system, but much shorter than in GSM [8].

6. CONCLUSIONS AND FUTURE WORK

A testbed has been implemented to study the impact of handovers and retransmissions on multimedia communications. The solution designed for handover at the wired side of the network is based on the leaf initiated join procedure. The solution designed for handover at the wireless side of the network is based on the following techniques: i) MT initiated handover; ii) forward handover; and iii) soft handover. Measurements taken for frame inter-arrival times show that this handover solution introduces no additional jitter. Besides, the duration of handovers have been also measured. Results show that the execution of a handover process takes a short time, which is similar to the one obtained in DECT systems and much shorter than in GSM.

Future work will focus on the implementation and evaluation of other handover techniques.

REFERENCES


