A New AAL Protocol for Time-Critical UDP Traffic over Wireless ATM Networks

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Abstract

This paper describes the design and performance of a new AAL protocol (AAL-UDP) for UDP/IP over wireless ATM networks. The wireless links are characterized by higher error rates and burstier error patterns in comparison with the fiber-based links for which ATM was designed. The purpose of AAL-UDP is to support time-critical UDP traffic over wireless ATM networks. The key ideas in the design consist of no discard at AAL level and header protection with sequence number mechanism. The UDP/IP header is repeated for reliability, because it contains the most important information such as address and port number. The proposed AAL has been validated using a software emulator on the simulated wireless ATM network. The simulation results show that the AAL-UDP provides significant improvement in throughput as well as application-level performance compared to the conventional AAL 5 case.

Key Words: Wireless ATM, AAL-UDP, Packet Discard, Sequence Number, Header Repetition, Cell Error Rate.

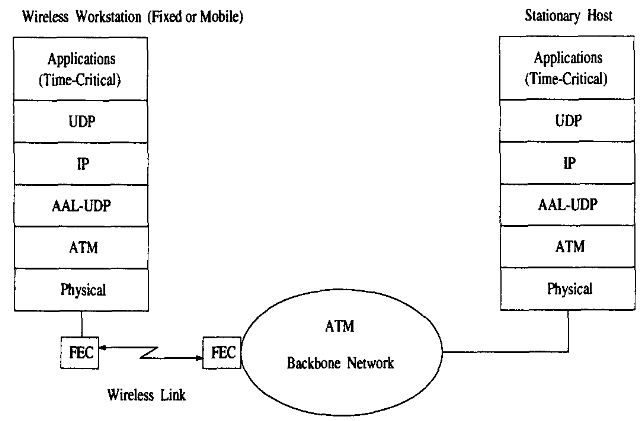

Figure 1: System Architecture for Time-Critical Traffic over Wireless ATM

1 Introduction

AAL (ATM Adaptation Layer) 5 has been selected by the ATM Forum [1] for the transport of video and audio packets in ATM networks. The reasons for this selection are: the wide acceptance of AAL 5 from both the computer and telecommunication industries, no requirement for extra hardware, (possibly) easy extension to support VBR video and audio transport, and effective error handling. However, the use of AAL 5 cannot be considered as the optimal solution for wireless ATM networks in terms of error characteristics, because there can be still severe discard of AAL packets due to payload error checking at AAL level. Moreover, AAL 5 has larger delay and overhead due to the 32-bit CRC, and AAL 5 cannot identify which cell is lost.

Because of the fading effects and interference, the wireless links are characterized by higher error rates and burstier error patterns in comparison with the fiber-based links. Thus, channel errors on the wireless channel will be typically beyond the error correcting capability of the ATM HEC (Header Error Control), making it necessary for more powerful FEC (Forward Error Control) schemes. Since time-critical traffic cannot tolerate retransmission delays, retransmission-based ARQ (Automatic Repeat Request) schemes are not considered here. As shown in Figure 1, FEC is used for each wireless link segment to raise the channel BER (Bit-Error-Rate) from its raw level to a level acceptable to higher layer protocols.

UDP (User Datagram Protocol) provides a datagram service on top of IP (Internet Protocol). Since UDP does not do retransmission and its protocol overhead is minimal, it is more adequate for time-critical traffic than TCP (Transmission Control Protocol). However, when UDP is used with AAL 5 over wireless ATM networks, this causes significant drop-outs in voice and video communications, because there is severe loss of UDP/IP packets due to discard at AAL level [9]. For time-critical UDP traffic, the end-to-end quality should be controlled by higher layers and should not be hindered by payload error checking and discard at AAL level.

If there is no error checking on the payload at lower layers, application software will receive the user data without discarding, even if the data contains a certain amount of errors. In general, however, time-critical traffic is error-tolerant to some degree because of the limitations of human ears or eyes. Moreover, error control methods can be employed at this high level to satisfy QoS (Quality of Service) requirements. For example, FEC attempts to correct transmission errors at the expense of bandwidth by adding redundancy, and error concealment techniques like interpolation attempts to ameliorate the impact of the errors on the reception quality of video or voice to users.

In this paper, we propose a new AAL, *AAL-UDP*, to support time-critical UDP traffic over wireless ATM networks to overcome the problems mentioned above. In the next section, we describe a detailed design of the AAL-UDP, followed by a concatenated FEC scheme for wireless ATM networks. In Section 4 we present the simulation models and performance evaluation results from a software emulation of the AAL-UDP. Finally, we conclude the paper by highlighting our contribution.

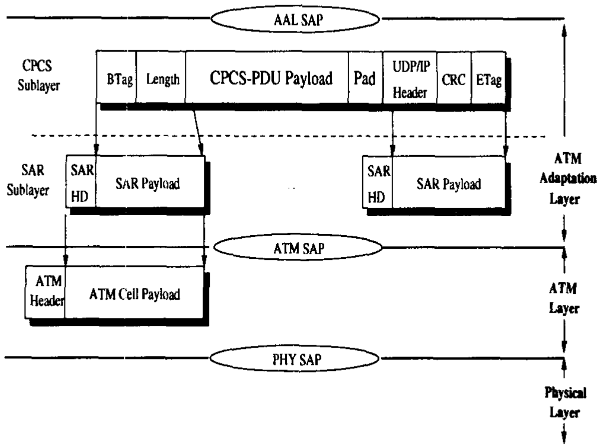

Figure 2: Packet Formats of AAL-UDP

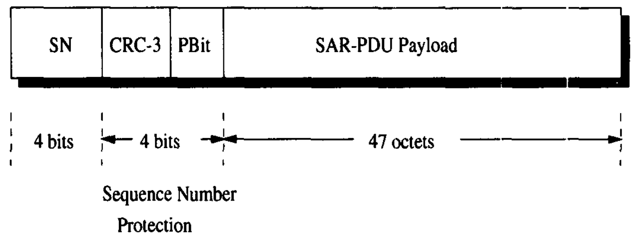

Figure 3: SAR-PDU Format of AAL-UDP

2 The AAL-UDP Design

The AAL-UDP supports time-critical voice and video traffic (such as electronic white board and voice) over wireless ATM networks. The design is based on the standard AAL 5, because lower-quality video and audio streams may be carried over ATM using AAL 5 with no time stamps at the ATM level (assuming a "buffer fullness" is used for timebase reconstruction).

In AAL-UDP, we need to know which exact cell is lost in order to provide time concept for reconstructing AAL packets at the receiver. Note that AAL 5 is not able to distinguish which cell of the CPCS-PDU is lost, since it does not use any sequence number mechanism. Thus, AAL 5 is not appropriate for time-critical traffic. The AAL-UDP provides the sequence number mechanism to detect missing cells.

The AAL-UDP will not perform packet discard at AAL level as a result of errors in the payload of AAL packets. Furthermore, even if there is a certain amount of cell discards or errors in the AAL packet, AAL packets will not be discarded as long as the destination IP address is valid.

The structure of the AAL-UDP design is given in Figure 2.

2.1 The Common Part Convergence Sublayer (CPCS)

The format of the CPCS-PDU is presented in Figure 2. The transmitter inserts the Beginning Tag (*BTag*) and the End Tag (*ETag*), one octet for each, in order for the receiver to identify each CPCS-PDU. The *Length* field (2 octets or 16 bits) consists of two parts: the first 12 bits indicate the length of the entire CPCS-PDU in unit of the SAR-PDU payload (47 octets) and the remaining 4 bits indicate the length of the UDP/IP header in 4 octets (UDP header is fixed as 8 octets, and IP header is usually 20 octets but it may be larger because of the options). These fields (tags and length) can be used as an additional mechanism to capture the lost or misinserted cells. The *Payload* field consists of (1-64K) octets of information. The *Pad* field (0-46 octets) is used to ensure that the total length of the CPCS-PDU is a multiple of 47 octets because one octet header will be added at the SAR sublayer. This field is located before the UDP/IP header field so that the pair of UDP/IP header and CRC fields is not split into two different cells. The *UDP/IP Header* is repeated for reliability because it contains the most important information (host address and port number) for packet delivery. Another reason is that if there is an error in the IP header, the entire IP packet is discarded due to the checksum error. For error detection, the *CRC-16* field (2 octets) is used for the UDP/IP header only.

The UDP/IP header is transmitted twice at a time, with the hope that at least one of them arrives at the destination correctly. If at least one of two UDP/IP headers is correct, the entire UDP/IP packet can be delivered to the destination, even if there is some error on the user data. The duplicated UDP/IP header is placed at the end of the CPCS-PDU far from the original header at the beginning of the CPCS-PDU, so that burst errors may not disturb both headers at the same time.

At the receiver, the second UDP/IP header is checked first with the CRC bits for error detection. If there is no error, the original UDP/IP header will be replaced by the second one and passed up to the upper layer. On the other hand, if an error is found in the second UDP/IP header, the AAL-UDP passes up the UDP/IP packet intact to the upper layer without discarding the packet, as long as the destination IP address is valid. Then the IP layer determines whether to accept the packet or not, based on the checksum field in the IP header. After that, the UDP layer just delivers the data part of the UDP packet to the destination port without error checking.

2.2 Segmentation and Reassembly Sublayer (SAR)

As shown in Figure 3, the format of SAR-PDU is the same as AAL 1. There is a sequence number (SN) field of 4 bits to detect missing cells and fill them with dummy ones for a timing purpose. For the sequence number protection, the SAR header includes CRC check bits (3 bits) and a parity bit (1 bit).

3 The Concatenated FEC Scheme for Wireless ATM

In wireless ATM, channel errors will be typically beyond the error correcting capability of the ATM HEC, making it necessary for more powerful FEC schemes. In recent literature, the concatenated FEC scheme has been proposed [3, 10] for wireless ATM by using a convolutional inner code and an RS (Reed-Solomon) outer code with interleaving. In particular, a rate-1/2, constraint-length 7 convolutional code with

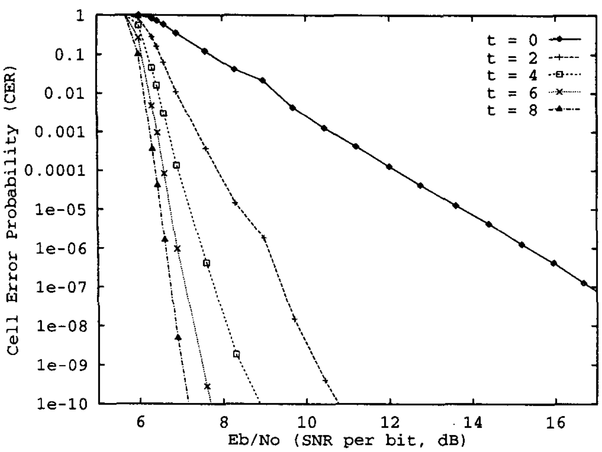

Figure 4: CER Performance of the Concatenated FEC Scheme on AWGN Rayleigh Fading Channel

QPSK modulation is used as an inner code, since it is an industry standard in digital cellular systems.

When the Viterbi decoder suffers a decoding error, the resulting codeword usually differs from the transmitted word by a few consecutive trellis branches. As a result, although the input to the Viterbi decoder is corrupted by random noise, the output of the decoder tends to have burst errors. This burst will typically be beyond the error correcting capability of HEC, causing a cell loss. Hence, the RS code with its inherent burst-error correcting capability is used as an outer code to deal with burst errors out of the Viterbi decoder.

Since the constraint length is chosen to be 7 for convolutional coding, the most frequently occurring error bursts at the output of the Viterbi decoder will have length 7 or 8. Thus, 8-bit symbols are chosen for the RS code to handle these burst errors with a single RS symbol. The RS codeword takes the form of (n, k) over GF(2^8), where n is the number of code symbols and k is the number of information symbols. Since the minimum distance of this code is $n-k+1$, it can correct i symbol errors and j symbol erasures at the same time as long as $(2i + j) \leq (n - k)$ is satisfied. For the concatenated coding scheme, the RS decoder processes the output stream of the Viterbi decoder. In this case, there are no symbol erasures and further the symbol error rate at the input of the RS decoder is the same as the symbol error rate at the Viterbi decoder output, $P_{s,Vit}$. Therefore, the word error rate for a codeword of n symbols at the output of the RS decoder that is capable of correcting up to t symbol errors is given by [7],

$$P_w = \sum_{i=t+1}^{n} \binom{n}{i} P_{s,Vit}^{i}(1 - P_{s,Vit})^{n-i} \tag{1}$$

In particular, if the information size of the RS code is chosen to be 53 symbols which is one ATM cell, the word error rate P_w corresponds to the CER (Cell Error Rate), where the CER is defined as the percentage of errored or lost cells to total cells transmitted.

Since there is a symbol interleaver between two decoders, the symbol errors at the input of the RS decoder are assumed to be independent, assuring that a given error burst from the Viterbi decoder affects no more than one symbol in an RS codeword. The evaluated results from Eq. (1) are presented in Figure 4 for the concatenated code of a rate-1/2, constraint-length 7 convolutional inner code and an $(n, 53)$ RS outer code for various error correcting capabilities (t = 2, 4, 6, 8 symbols) on an AWGN Rayleigh fading channel with QPSK modulation. For comparison, the convolutional code alone case (t = 0) with the rate-1/2 and constraint-length 7 is also shown in Figure 4. The results show that the more error-correcting capability, the steeper performance curve with more coding gain, and consequently the concatenated coding scheme can offer sufficient error performance for wireless ATM.

The desired CER can be obtained by properly choosing input parameters of the concatenated FEC scheme such as SNR (Signal-to-Noise Ratio) and t values. For example, the CER 10^{-6} can be generated when the SNR value per bit is about 6.9 dB and the t value is set to 6. Since time-critical traffic is error-tolerant to some degree, the desired CER can be relaxed so that power or bandwidth to be consumed by the FEC scheme can be saved, which is quite important in that the wireless link has a much lower bandwidth and mobile terminals are usually battery-powered. Although this relaxation causes residual errors after FEC correction, the AAL-UDP will minimize the effect of residual errors by providing correct delivery to the destination and performing no discard at AAL level.

4 Simulation

In this section, we evaluate the performance of AAL-UDP for UDP/IP over wireless ATM networks by conducting two types of simulation. Section 4.1 presents the throughput of AAL-UDP, while Section 4.2 describes how the AAL-UDP affects on the application-level performance.

4.1 Throughput

we develop two simulation models:

- *Configuration 1*: UDP/IP and AAL 5 as the conventional approach.
- *Configuration 2*: UDP/IP and AAL-UDP as our new approach.

The implementation of the simulation models was carried out on a Sun Sparcstation by using a software emulator written in C. In our experiment, there are two cases for the payload length of the UDP/IP packet: voice and video. The voice packet has the payload length of 318 bytes, and the video packet has the payload length of 1600 bytes [8].

The wireless channel is modeled as a Rayleigh fading channel to be combined with AWGN (Additive White Gaussian Noise) generator. For simulation parameters, the carrier frequency is 2.4 GHz ISM band and the data rate is DS1 (1.544 Mbits/s). By adjusting the SNR value and Doppler shift, specific error situations can be generated in terms of CER (Cell Error Rate).

We compare the normalized throughput at the UDP layer between AAL 5 and AAL-UDP as a function of CER for voice packets and video packets, respectively, as shown in Figure 5(a) and (b). The normalized throughput is defined as the ratio of throughput to offered load. The AAL-UDP provides almost a perfect normalized throughput 1.0 until CER of 10^{-2} for both voice and video packets, while the throughput of AAL 5 degrades rapidly as the CER becomes higher.

For example, the normalized throughput of AAL 5 is about 0.92 for voice packets at the CER of 10^{-2}, whereas it is as low as 0.65 for video packets at the same CER. Since video

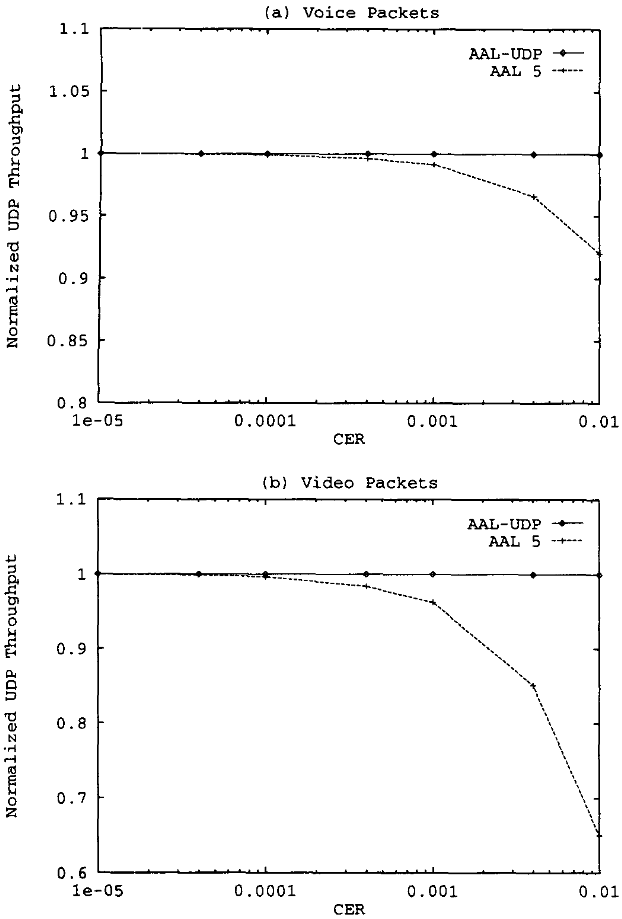

Figure 5: Comparison of Normalized UDP Throughput for AAL 5 vs. AAL-UDP

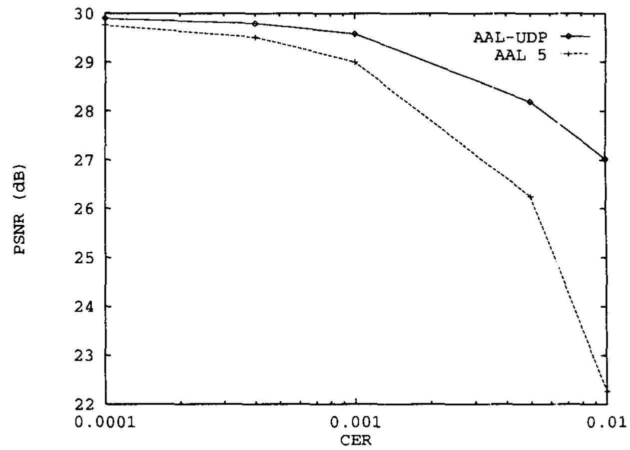

Figure 6: Application-Level Performance in terms of PSNR for MPEG video

packets have a high probability of packet discard in AAL 5 for higher CER ranges due to the larger packet size, the throughput of AAL 5 for video packets drops more substantially compared to the voice case. However, the AAL-UDP can still offer high throughput even for higher CER ranges, because it performs no discard at AAL level and further provides the header protection to increase the probability of correct delivery of UDP/IP packets to the right destination.

4.2 Application-Level Performance

The purpose of this simulation is to assess the impact of AAL-UDP on the application-level performance over wireless ATM networks. Since MPEG has been accepted by the ATM Forum as the standard for digital video transmission, MPEG is used to carry time-critical video traffic as an application for our simulation. For MPEG video data, a video sequence (*Flower Garden*) has been chosen. The video sequence consists of 150 picture frames and has a resolution of 352 by 240.

When the MPEG application transmits the video sequence by UDP/IP packets over wireless ATM networks, the application-level performance is evaluated using the PSNR (Peak Signal-to-Noise Ratio) as a function of CER for AAL5 and AAL-UDP, respectively. The PSNR is normally used as an objective measure of video quality. In general, the video quality begins to visually noticeable for a difference of more than 0.5 dB and is clearly visible for a difference of more than 1 dB [2].

Figure 6 presents the video quality as the application-level performance in terms of the PSNR in dB for AAL 5 and AAL-UDP as a function of CER. The results show that the AAL-UDP outperforms the AAL 5 case for all CER ranges, especially for higher CER values as in a wireless ATM channel. For example, the difference of video quality between AAL 5 and AAL-UDP is visually noticeable around 10^{-3} of CER, and further it is clearly visible beyond this point toward higher CER values because the PSNR values of AAL-UDP are greater than AAL 5 by 1 dB or more. This improvement is due to the fact that the AAL-UDP performs no discard at the AAL level and increases the probability of correct packet delivery to the destination by header protection.

4.3 Discussions

If there is at least one wireless section in the end-to-end ATM path from the source to the destination, the AAL-UDP can be clearly justified as a better solution compared to AAL 5 from the simulation results. The AAL-UDP provides less delay and higher throughput for higher error ranges like in a wireless ATM path.

On the other hand, let us discuss the case where the end-to-end ATM path does not contain any wireless section just like in traditional wireline ATM networks. In order to check if the AAL-UDP can be still applied in this case, we compare between AAL 5 and AAL-UDP in two different perspectives. First, consider the normalized throughput at the UDP level. For wireline ATM networks, the CER range falls below 10^{-5} when considering the BER values of fiber links as 10^{-9} or 10^{-10}. Therefore, there is no difference in terms of throughput at the UDP level regardless AAL 5 and AAL-UDP over wireline ATM networks, because they provide the same normalized throughput 1.0 as shown in Figure 5.

Second, the overhead ratio is compared at the AAL level between AAL 5 and AAL-UDP in terms of bandwidth efficiency. For voice packets, the total overhead size in AAL 5 is 38 bytes (i.e., 30-byte pad plus 8-byte trailer of AAL 5), while the total overhead size in AAL-UDP is 77 bytes (i.e., 3-byte header of AAL-UDP, 43-byte pad, and 31-byte trailer of AAL-UDP including the UDP/IP header). For

Packet Types	AAL 5	AAL-UDP
Voice Packets	10% (38/384)	18% (77/423)
Video Packets	3% (52/1680)	4% (64/1692)
MPEG-2 Packets	10% (24/240)	23% (66/282)

Table 1: Overhead Ratio Table of AAL 5 Vs. AAL-UDP for Different Packet Types

video packets, the total overhead size in AAL 5 is 52 bytes (i.e., 44-byte pad plus 8-byte trailer of AAL 5), while the total overhead size in AAL-UDP is 64 bytes (1.e., 3-byte header of AAL-UDP, 30-byte pad, and 31-byte trailer of AAL-UDP including the UDP/IP header).

Table 1 presents the overhead ratio (%) of AAL 5 and AAL-UDP with respect to the total AAL-PDU size for voice packets, video packets, and MPEG-2 transport packets each. In summary, the AAL-UDP cannot be justified as a good solution for wireline ATM networks, because it consumes a lot of bandwidth due to the larger overhead compared to the AAL 5 case. Here, we propose to use two different AAL protocols for time-critical UDP traffic: AAL 5 for wireline ATM networks and AAL-UDP for wireless ATM networks. In this approach, a mechanism is required in the ATM connection setup phase to detect whether there is any wireless section in the end-to-end ATM path from the source to the destination, in order to determine which AAL type will be used for this connection.

5 Conclusions

For time-critical traffic, the end-to-end quality should be controlled by higher layers and should not be hindered by payload error checking or discard at the AAL layer. Therefore, a primitive protocol is appropriate at the AAL level, while more sophisticated algorithms for error control will be employed at higher layers to meet QoS expectations. In this paper, we proposed a new AAL protocol, *AAL-UDP*, to support time-critical UDP traffic over wireless ATM networks. The key ideas in the design consist of no discard at AAL level and header protection with sequence number mechanism. The simulation results show that the AAL-UDP provides significant improvement in throughput as well as application-level performance for UDP/IP over wireless ATM networks, compared to the conventional AAL 5 case.

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