An Adaptive Redundancy Technique for Wireless Indoor Multicasting

Youshi Xu and Tingting Zhang
Institution of Information Technology
Mid-Sweden University
S-851 70 Sundsvall
Sweden
Email: Tingting.zhang@ite.mh.se

Abstract

In this paper we present an adaptive technique that combines forward error correction (FEC) with channel state estimation (CSE) and automatic repeat request (ARQ) for packet loss recovery in wireless indoor multicast systems. The novel aspect of the technique is its ability to achieve significant packet throughput with high data reliability, to avoid feedback acknowledgement (ACK) implosion, as well as to reduce and flexibly to limit delay for real-time applications. We suggest a set of criteria to measure the multicast performance and simulate a simple two-state indoor channel model. The simulation results show that the proposed technique significantly improves the communication quality and channel efficiency, to compare with traditional FEC and ARQ techniques.

1. Introduction

In multimedia communications, it is often to distribute a data stream to multiple radio teminals. For example, in a video conference, the voice and pictures are sent to all participants located at different sites of a wireless indoor network. When the network access bandwidth is low, it may impossible for the source to send these data to each participant individually one by one. A better solution is that the source sends the same copy of the data simultaneously to all the addressed users in the network. This technique is called multicasting or point-to-multipoint communications.

Reliable multicast over wireless indoor networks remains a challenging research problem. Most protocols for reliable multicast use either forward error correction (FEC) techniques that result in large bandwidth requirement or automatic repeat request (ARQ) techniques that result in feedback acknowledgement (ACK) implosion. In this paper we present an adaptive technique that combines FEC with ARQ and channel state estimation (CSE) for packet loss recovery. This technique is based on the concept of hybrid ARQ schemes [3, 4, 5] and on the fact that indoor radio links are slowly time variable [1, 2, 7, 8] which implies that the estimation of channel state is possible.

The remainder of this paper is structured as follows. Section 2 outlines the adaptive technique for recovery of the lost packets in indoor multicast environment. In Section 3, we suggest and define a set of criteria to measure the multicast performance. In section 4, a two state channel modal is described, and some simulation results and comparisons are given. The simulation results show that the proposed technique significantly improves the communication quality and channel utilization efficiency, to compare with traditional FEC techniques and ARQ techniques. Finally, we conclude the paper in Section 5.

2 Outline of the adaptive technique

In the section we describe the proposed adaptive redundancy technique for recovery of erased packets through the unreliable multicast links in wireless indoor networks. We first outline a two dimensional concatenated coding scheme for bit-error correction in packets and packet-erasure resilience in blocks.

Information data is segmented into blocks. Suppose that a block information has $K \times k$ bits and is set to a two-dimensional array. Further let $k = k'm$ and every m bits form a symbol in finite field GF(2^m). The encoding is performed in two stages: outer (column) encoding and inner (row) encoding. Typically, the outer-codes are Reed-Solomn (RS) codes over GF(2^m). Each column of information symbols in GF(2^m) is encoded into a codeword of $C_0(N, K)$, where $N = K+R$ (number of redundancy symbols). Totally there are k' outer codewords in a block. Then, add packet header of length h bits to each row, encode each row into a codeword of $C_{in}(n, k+h)$ code, and form a packet of length n. Totally there are K information packets

0-7695-0722-0/00 $10.00 © 2000 IEEE

and R redundancy packets in a block. A binary BCH code can possibly be used as an inner code for simultaneous error correction and error detection.

The decoding is also performed in two stages: inner (row) decoding and outer (column) decoding. If the number of errors in a packet is less or equal to a predetermined value, the information data can correctly be found, and fed to the corresponding row of information array, otherwise, the packet is declared erasures. Then it is outer decoding to recover the erasures. As a maximum distance separable (MDS) code, the code C_0 is able to recover all erased information symbols if K symbols of N are received [14].

The packets are transmitted in order $i = 1, 2, ...N$, but this does not mean that all the redundancy packets will be transmitted. If all the terminals have correctly received K packets regardless of message packets or redundancy packets, all the remaining redundancy packets will not be sent and can be discarded. Then, a new block of packets is transmitted. Therefore, the R is called maximum number of redundancy packets.

The proposed multicast scheme is shown in Figure 1. When a packet numbered i is transmitted, the preset number (r) of redundancy packets which will be sent is attached to the packet header. For a user, if the received packet can be correctly decoded the data is fed to the block decoder, otherwise it is set as erasures. Consequently, if the total number of correct packets is $s = K$, all the erased packets can be recovered and the user will wait for the next block of packets. Otherwise, the user must determine whether an NAK is needed. Now, let us suppose that i packets have been transmitted and s packets correctly decoded. Assume that the remaining packets $(K + r - i)$ might correctly be received. If $s + (K + r - i) \geq K$, i.e., $s + r - i \geq 0$, which implies erasures can be recovered, an NAK to the sender is not needed. Otherwise, the user sends an NAK.

In wireless LANs, the round-trip propagation time between terminals and base stations is on the order of a few microseconds and even less than one microsecond. In many cases, for example, if time division multiple access (TDMA) is applied in the system, an acknowledge (ACK or NAK) for the current packet can arrive at the base station before the beginning of the next packet transmission, or at most one packet later. In our studies, we have assumed immediate NAKs are possible. One packet delay would have little effect on performance. A few packets delay, for instance, in some CDMA systems, can works well by a little change of the scheme. This is shown in Section 4.

If the base station receives one or more NAKs, the preset number of redundancy packets will increase by one and the updated number r will be attached to the header of the next packet. If the packets of planned number have been transmitted, i.e.,$i = K + r$, and no NAKs arrive at the base station, the block is end because all the terminals can recover all the erased information packets. The block is also end, if all the redundancy packets of maximum number R, i.e.,$i = N$, have been transmitted. In the case, some terminals may loss some packets. Most real-time applications can not tolerate long time delay associated with the maximum number of redundancy packets, which may cause packet loss at some terminals. For some applications which can tolerate long time delay, the R can be designed large enough to achieve a full reliable multicast. Of course, terminals in badly poor link condition can be detached from the multicast group.

When a new block starts, the initial preset number (R_0) of redundancy packets will be updated as shown in Equation (1).

$$R_0 \leftarrow \begin{cases} r & \text{if } R_0 < r \leq R \\ R & \text{if } r > R \\ \lfloor \frac{R_0}{2} \rfloor & \text{if } r = R_0 \neq R \end{cases} \quad (1)$$

If the final transmitted packet number ($i = K + r$) in the previous block was greater than the original planned number ($K + R_0$), it implies that the links have now become worse than expected. Thus we initial $R_0 = r$ for the new block. If $R_0 = R$ (maximum) and NAKs were received after the transmission of the last redundancy packet, which implies that some terminals were in very poor link condition and message packet loss occurred, we still set initial $R_0 = R$. If no any NAKs were received in the previous block, it implies that the links may be better than expected. Then, the R_0 is reduced.

During the transmission of a block, the planned number of redundancy packet is adapted to the worst link and multicasted to all the terminals. This will significantly reduce the number of NAKs. Since the base station is interested in whether there are NAKs after a packet transmission, not interested in which and how many terminals send the NAKs, therefore the feedback packets can be very short and very simple, and the feedback collisions or lost NAKs from hidden terminals have little or no effect on performance.

3 Criteria of Multicast Performance

When studying the various protocols for point-to-point data link control, we are interested in the characteristics of channel throughput, delay, acknowledgement and data integrity. In the section, we suggest and define a set of criteria to measure the performance of point-to-multipoint communications, specially indoor multicast.

In practice, since there is a limited number of redundancy packets, R, erased message packets at some terminals might not be recovered. The data integrity can be measured using two parameters: average packet loss rate L_{av} and peak packet loss rate L_{pk}, which are defined as follows.

Assume that the number of terminals in the system is M, the number of transmitted blocks is Ω and large enough, the number of information packets in each block is K. After the transmission of the j^{th} block, the number of unrecovered information packets in the j^{th} block at the terminals can be written as $\{l_{j1}, l_{j2}, ..., l_{jM}\}$ with $0 \leq l_{ji} \leq K$. The average packet loss rate can be defined as

$$L_{av} = \frac{1}{\Omega M K} \sum_{j=1}^{\Omega} \sum_{i=1}^{M} l_{ji} \quad (2)$$

The peak packet loss rate is defined as

$$L_{pk} = \frac{1}{\Omega K} \sum_{j=1}^{\Omega} \max\{l_{j1}, l_{j2}, ...l_{jM}\} \quad (3)$$

which should be much greater then Lav and reflects the worst links. If both $L_{av} = 0$ and $L_{pk} = 0$, it is called full reliable multicast, otherwise called conditional reliable multicast.

Since the signal propagation time within a cell can be neglected, the channel throughput η is defined as the average number of packets successfully transmitted to all the addressed terminals in a unit of packet time. For the proposed protocol, that is

$$\eta = \frac{K}{K + R_{av}} \quad (4)$$

where K is the number of information packets and R_{av} the average number of redundancy packets in a block transmission. For some applications, for example, real-time multicasting, we can not guarantee that all the terminals will successfully receive all the information packets. In the case, conditional throughput η_{cd} can be adopted to measure the performance. The conditional means that the packet loss rate is less than a give value, as defined before, and the R_{av} is a statistic value under the condition of packet loss rate.

The channel utilization efficiencies U for full reliability and U_{cd} for conditional reliability are consequently

$$U = \eta \frac{k}{n} \quad (5)$$

$$U_{cd} = \eta_{cd} \frac{k}{n} \quad (6)$$

where k is the number of information bits in a message packet, n the packet length in bits.

Generally, the packet delay is defined as the time interval between the time at which a packet arrives at the sender and the time at which it is received by the users. In the paper, we assume that a block of data is stored in the buffer of the sender and the maximum number of redundancy packets R is given. This implies that both the engaged channel capacity and the maximum delay are limited.

In the proposed protocol, if some of information packets are erased, they might be recovered after redundancy packets of required number are successfully received. Therefore we instead define the average block latency T_{av} and maximum block latency T_{mav} in WLANs as

$$T_{av} = K + R_{av} \quad (7)$$

$$T_{mav} = K + R \quad (8)$$

We apply two parameters to measure the acknowledgement collision levels: average NAK density and peak NAK density, denoted as λ_{av} and λ_{pk} respectively. The λ_{av} is defined as the average number of NAKs per terminal after each packet transmission. Let the j^{th} block transmission contains N_j packets. For the i^{th} packet transmission, there are λ'_{ji} NAKs. Suppose total number of blocks is Ω and the number of terminals M. The λ_{av} can be calculated by

$$\lambda_{av} = \frac{1}{\Omega M} \sum_{j=1}^{\Omega} \frac{1}{N_j} \sum_{i=1}^{N_j} \lambda'_{ji} \quad (9)$$

It is easy to image that there are few NAKs at the beginning of transmitting a block of packets and more NAKs at the later transmissions. Moreover, when all the terminals are in good link condition, there may be no any NAKs during a block transmission period, on the contrary, there are many. To reflect the ununiform, we now define the peak NAK density λ_{pk}. We first find the maximum number of NAKs in the set, $\lambda'_j = \max\{\lambda'_{j1}, \lambda'_{j2}, ..., \lambda'_{jN_j}\}$, for $j = 1, 2, ..., \Omega$. The peak NAK density is defined as the mean of λ'_j per terminal under the condition of $\lambda'_j \neq 0$, that is

$$\lambda_{pk} = \frac{1}{\Omega' M} \sum_{j=1}^{\Omega} \max\{\lambda'_{j1}, ..., \lambda'_{jN_j}\} \quad (10)$$

where Ω' is the number of blocks with $\lambda'_j > 0$.

4 Simulation Results and Comparisons

We study the performance of our proposed protocol for multicasting over wireless local area networks using computer simulations. Events in the simulations occur on a single cell and with no power control under consideration.

4.1 Channel model

Indoor path loss has been shown by many researchers to obey the distance power law [1, 13]. The average received power (a_0 in dB) at a mobile station is

$$a_0(d) = a(d_0) - 10\alpha \log(\frac{d}{d_0}) + X \quad (11)$$

where d is the distance between the base station and the mobile station, d_0 reference distance, $a(d_0)$ received power at reference distance, α average path loss index, X location dependent variable.

The location-dependent variable describes the random shadowing effects which occur over a large number of measurement locations which have the same sender-to-receiver separation, but have different levels of received power. The location-dependent variable X in dB is a zero-mean Gaussion distributed random variable with a standard deviation of σ dB [10].

In practice, the values of α and σ are computed from measured data. The typical values are $\alpha = 1.5 - 2.4$, $\sigma = 4 - 10$ dB for factories [12], and $\alpha = 2.7 - 5.0$, $\sigma = 4.3 - 16$ dB for multifloor buildings [13]. The experiments have shown that indoor channel is temporal stationary only during short intervals of time [1, 3]. Due to the motion of people and equipment in most indoor environments, the channel is nonstationary in time of large scale, i.e., the channel is statistically changed, even when the terminal is location-fixed. To reflect the temporal variation, the received power as a random process can be represented by

$$a(d,t) = a_0(d) + Y(t) \tag{12}$$

The temporal variation $Y(t)$ normally occurs in bursts lasting several or tens seconds with dynamic range of about 17 dB, even 30 dB. The bursts of fadings form Poisson arrival processes with different rate, and have exponentially distributed durations [11]. Therefore the channel can be modelled as a two-state Markov chain, shown in Figure 2. We assume that a terminal link is always in one of the two states: the stationary (Sta) and the unstationary (Uns). Let Ts and Tu be the mean durations in stationary state and in unstationary state respectively. let τ be transmission time of a packet, and $\tau \ll T_s, \tau \ll T_u$, so that no more than one transition occur in a packet transmission period. Then the transition probabilities are respectively

$$\sigma = 1 - \exp(-\frac{\tau}{T_s}) \tag{13}$$

$$\beta = 1 - \exp(-\frac{\tau}{T_u}) \tag{14}$$

In stationary state, we can simply assume that the value of temporal variation is zero, that is $Y(t) = 0$ dB. In the unstationary state, the received signal suffers fadings in bursts. The temporal fading shows a good fit to Rician distribution, if the carry frequency is around 1GHz [2, 3, 12]. It is also indicated that it is nearly Rayleigh distributed, if it is measured at 60 GHz [5]. In the paper, we use Rayleigh distribution to simulate the bursty fadings. The Rayleigh probability densing function for amplitude envelope r is given by

$$f_R(r) = \begin{cases} \frac{r}{w^2}\exp(-\frac{r^2}{2w^2}) & 0 \le r \le \infty \\ 0 & r < 0 \end{cases} \tag{15}$$

where w is the rms value of the received voltage signal before envelope detection. The square of the magnitude of Rayleigh distributed random variable represents the signal power after detectors. To transform the amplitude distribution to power distribution in dB, we set the variable power after detectors $a = 10\log_{10} r^2$ and the average power after detectors $a_0 = 10log_{10}2w^2$. The transformation equation can be written as

$$a - a_0 = 10\log_{10}\frac{r^2}{2w^2} \tag{16}$$

Since $r \ge 0$ and $w \ge 0$, we have

$$r = \sqrt{2}w\exp(\frac{a - a_0}{c}) \tag{17}$$

where $c = 20/ln10 = 8.686$. Taking the derivative, we have

$$\frac{dr}{da} = \frac{\sqrt{2}w}{c}\exp(\frac{a - a_0}{c}) \tag{18}$$

From probability theory [6], the probability density function of a can be achieved from $f_R(r)$ by

$$f_A(a) = f_R(r)|\tfrac{dr}{da}| = \frac{2}{c}\exp[\frac{2(a - a_0)}{c}]\exp\left\{-\exp[\frac{2(a - a_0)}{c}]\right\} \tag{19}$$

From Equation (12) we know $Y(t) = a(d,t) - a_0(d)$, therefore the probability density function of Y is

$$f_Y(y) = \frac{2}{c}\exp(\frac{2y}{c})\exp[-\exp(\frac{2y}{c})] \tag{20}$$

For simplicity, we assume that all the terminals within the cell have the same value of average interference and noise, N_{av} in dB. Thus the signal-to-noise ratio γ at a mobile station is

$$\gamma(d,t) = a(d_0) - 10\alpha\log_{10}(\frac{d}{d_0}) + X + Y(t) - N_{av} \tag{21}$$

If we denote the SNR at the reference distance as

$$\gamma(d_0) = a(d_0) - N_{av} \tag{22}$$

then the average SNR at a receiver can be written as

$$\bar{\gamma}(d) = \gamma(d_0) - 10\alpha\log_{10}(\frac{d}{d_0}) + X \tag{23}$$

and Equation (21) can be represented by

$$\gamma(d,t) = \bar{\gamma}(d) + Y(t) \tag{24}$$

where $Y(t) = 0$ at the stationary state; otherwise, the distribution function is given in (20).

In our simulation, we set $d_0 = 1$ m, $\gamma(d_0) = 70$ dB,$\sigma = 10$ dB, $\alpha = 3$, $T_u = 10$ seconds, $T_s = 60$ seconds and $\tau = 2$ ms. It is also assumed that the radius of a cell is $d_{max} = 50$ meters and all the mobile stations are uniformly distributed in the cell. For simplicity, we assume that the random value of $Y(t)$ is mutually independent for each and one of the receivers in the system, but it is fixed during a packet transmission period for a receiver.

4.2 Simulation example

As known, the packet erasure rate depends on both the channel state, i.e. signal-to-noise ratio γ, and the design of bit-error correction codes in packets. How to design the bit-error correction codes and erasure recovery codes for optimum channel efficiency is beyond the scope of the paper. In the simulations, if the average SNR in a packet is greater than some threshold γ_0 we assume the packet is correctly decoded without errors, otherwise, the packet is erased. This assumption is reasonable for multiple receivers. It is obvious that if a terminal is located in radio-silent zone, i.e., the average SNR in (23) is less than the threshold γ_0, the communication between the terminal and the base station is almost impossible. In our simulation, this possibility is excluded by re-placing the terminal on a new random location so that $\bar{\gamma}(d) > \gamma_0$.

We first determine how the number of terminals and their mobility within a cell affect the packet loss rate. In the simulation, we give that threshold of SNR is $\gamma_0 = 6$ dB, the block size is equal to the number of information packets in a block, that is, $N = K$ with no redundancy packets, which implies that the erased packets are also lost. From a number of simulations, the statistic results show that the number of users within a cell has little or no effect on average packet loss rate.

Figure 3 plots the average packet loss rate (L_{av}) and peak packet loss rate (L_{pk}) as a function of the threshold of signal-to-noise ratio (γ_0) respectively. In the simulation, we give that the number of terminals is $M = 50$ and no redundancy packets are applied to recover the erasures. As expected the loss rate increases as the threshold increases. However, reducing the threshold implies increasing the error correction capability in a packet, which results in significant decreasing the channel efficiency. This should further be studied.

Usually, packet loss rates $L_{av} \leq 10^{-5}$ and $L_{pk} \leq 10^{-3}$ are expected. This figure shows that the packet loss rate is extremely high which are not acceptable for any applications. Therefore some techniques must be used to recover the erased information packets, for example, redundancy packets proposed in the paper.

Figures 4 shows the packets loss rates as functions of the maximum number of redundancy packets, if our proposed protocol is applied. In the simulation, we assume $K = 32$, $\gamma_0 = 6$ dB and $M = 50$. It is obvious that the packet loss rates L_{av} and L_{pk} decrease as maximum number of redundancy packet increases. From the simulation example, it can be found that if we set the maximum redundancy $R = 49$, then a full reliable multicast $L_{av} = 0$, $L_{pk} = 0$ can be achieved; if we set $R = 32$, then a conditional reliable multicast can be achieved with $L_{av} = 0.1061 \times 10^{-4}$ and $L_{pk} = 0.5295 \times 10^{-3}$.

4.3 Comparisons

To compare multicast performance between different techniques, we also simulate a forward error correction (FEC) protocol and an automatic repeater request (ARQ) protocol. The advantage of FEC over the proposed technique is that no any feedback message is needed. Since the number of transmitted redundancy packets is fixed, many of them are often wasted. In the simulation of FEC, we chose $R = 32$. To simplify the comparison, we simulate the selective-repeat ARQ with NAKs, window size $K = 32$, and given that the maximum retransmissions of an original packet is not greater than 2. The redundancy packets are retransmitted packets in ARQ. Other parameters are all the same for the three schemes, e.g. $\gamma_0 = 6$ dB and $M = 50$. The simulation results are listed in Table 1.

The table shows that to achieve a conditional reliable multicast with $L_{av} = 10^{-5}$ and $L_{pk} = 0.53 \times 10^{-3}$, the traditional FEC always needs 32 redundancy packets and the proposed one averagely need 8.04 which will improve the channel throughput by a factor of 1.6. It can be seen that the proposed protocol reduces the packet loss rates L_{av} and L_{pk} with a factor of 24.3 and 21.9 respectively and NAK densities λ_{av} and λ_{pk} with a factor of 5.1677 and 1.438 respectively, although the new protocol uses a little less redundancy ($R_{av} = 8.0409$) than the traditional ARQ ($R_{av} = 8.8987$).

In the above simulations, we assume that the NAKs for the current packet arrive at the base station before the next packet is transmitted. Suppose now that the NAKs have a delay of one packet transmission time. If the base station receives an NAK, it increases the planned number of redundancy packets by 2. The simulation results of the proposal protocol with this small change are follows $R = 32$, $R_{av} = 8.5179$, $L_{av} = 0.1049 \times 10^{-4}$, $L_{pk} = 0.5245 \times 10^{-3}$, $\lambda_{av} = 0.0005$, $\lambda_{pk} = 0.0194$, $\eta_{cd} = 0.7898$, $T_{av} = 40.5179$, $T_{max} = 64$. To compare with the values on column New in Table 1, it can be found that the delay has just a little effect on the performance. If the delay is greater than one packet transmission time, the problem can be solved in the same way.

Table 1. Performance comparison between the proposed technique (New) and traditional FEC and selective repeat ARQ.

	FEC	New	s-ARQ
maximum redundancy R	32	32	53
average redundancy R_{av}	32	8.0409	8.8987
average loss rate L_{av} (10^{-4})	0.1061	0.1061	2.5794
peak loss rate L_{pk} (10^{-3})	0.5292	0.5295	11.6
average NAK density λ_{av}	0	0.0012	0.0062
peak NAK density λ_{pk}	0	0.0203	0.0292
throughput η_{cd}	0.5	0.7992	0.7824
average block latency T_{av}	64	40.0409	40.8987
maximum block latency T_{mav}	64	64	85

5 Conclusion

In this paper, the indoor radio channel is briefly reviewed and a simulation model for the slow time-variable channel is introduced. We have suggested and defined a set of criteria, such as conditional throughput, peak loss rate, peak NAK density and etc., to measure the performance of multicasting. Based on the concept of hybrid automatic-repeat-request, we have proposed a coding scheme of bit-error correction in packets and packet-erasure resilience in blocks to achieve reliable and efficient point-to-multipoint data transmissions. By dynamically adapting the number of redundancy packets, the problem of acknowledgement collision in multicasting systems can be avoided. Our results for a particular design example show that (1) To compare with the selective repeat FEC systems which do not need feedback ACKs, this proposed protocol will significantly improve the channel throughput by a factor of 1.6 for the same packet loss rates,$L_{av} = 10^{-5}$, $L_{pk} = 0.53 \times 10^{-3}$. (2) To compare with the traditional ARQ protocols, this proposed protocol will reduce the packet loss rates L_{av} and L_{pk} with a factor of 24 and 22 respectively and NAK densities λ_{av} and λ_{pk} with a factor of 5.2 and 1.4 respectively, although the proposed protocol uses a little less redundancy. (3) This scheme can potentially solve the problems of feedback ACK implosion and temporal hidden terminals, but an entry in each packet header is needed to announce the planned number of redundancy packets, which is simple to be implemented and consumes little bandwidth. Therefore, the proposed technique is possible to play a useful role in wireless multicast systems.

Acknowledgments

The authors world like to thank Dr. Carl-Gustav Löf, Mr. Conny Flemin and Mr. Tobias Borg for useful discussions and a part of simulation work during the study.

References

[1] J. B. Andersen, T. S. Rappaport and S. Yoshida. Propagation measurements and models for wireless communications channels. *IEEE Communication Magazine,* 42-49, January 1995.

[2] R. J. C. Bultitude. Measurement, characterization and modelling of indoor 800/900 MHz radio channels for digital communications. *IEEE Communication Magazine,* 25(6):5-12, June, 1987.

[3] R. J. C. Bultitude, S. A. Mahmoud and W. A. sullivan. A comparison of indoor radio propagation characteristics at 910 MHz and 1.75 GHz. *IEEE Journal on Selected Area in Communications,* 7(1):20-30, January 1989.

[4] R. H. Deng and M. L. Lin. Type 1 hybrid ARQ system with adaptive code rates. *IEEE Transactions on Communications,* 43(2/3/4):733-737, Feb./Mar./Apr. 1995.

[5] H. Hashemi. The indoor radio propagation channel. *Proceedings of the IEEE,* 81(7) July 1993.

[6] A. Leon-Garcia. *Probability and random processes for electrical engineering.* (second edition), Addison-Wesley, 1994.

[7] S. Lin and P.S.Yu. A hybrid-ARQ scheme with parity retransmission for error control of satellite channels. *IEEE Transactions on Communications,* 30(7):1701-1709, July 1982.

[8] J. Nonnenmacher, E. W. Biersack and D. Towsley. Parity-based loss recovery for reliable multicast transmission. *IEEE/ACM Transaction on Networking,* 6(4), August 1998.

[9] K. Pahlavan and A. H. levesque. *Wireless information networks.* John Wiley&sons,105-108, 1995.

[10] T. S. Rappaport. *Wireless communications.* 44-106 and 123-128, IEEE press, 1996.

[11] A. A. M Saleh and R. A. Valenzula. A statistical model for indoor multipath propagation. *IEEE Journal on Selected Areas in Communication,* 5(2):128-137 February 1987.

[12] Teodore S. Rappaport. Indoor radio communications for factories of future. *IEEE Communication Magazine, May,* 15-24, 1989.

[13] Scott Y. Seidel and Teodore S. Rappaport. 914 MHz path loss prediction models for indoor wireless communications in multifloored buildings. *IEEE Transactions on Antennas and Propagation,* 40(2):207-217, Feb. 1992.

[14] Youshi Xu. A maximum likelihood erasure-decoding scheme for concatenated codes. *IEE Proceedings I,* 139(3):236-239, June 1992.

Notations:
i: the ith row or packet.
r: the preset number of redundancy packets
s: the number of packets correctly received

Base station:

1. Input a data block ($K \times k$);
2. Outer encoding: $C_0(N, K)$;
   Initial $i = 0, r = R_0$;
3. Do while $i < N$ and $i < K + r$,
   (a) $i \leftarrow i + 1$;
   (b) Inner encoding (packet): $C_{in}(n, k + h)$;
   (c) Send Packet i;
   (d) If NAK (one or more) is received,
   then $r \leftarrow r + 1$;
4. Reset R_0 by Equation (1)
5. Go to step 1.

Mobile Station:

0. Initial s = 0;
1. Receive Packet i;
2. Inner decoding;
3. If correct,
   then Data to buffer of outer decoder;
   $s \leftarrow s + 1$;
   else Packet is erased;
4. If $s + r - i < 0$ and $i \leq N$,
   then send NAK;
5. If $s = K$ or $i = N$,
   then Outer decoding;
   Stop for next block;
   back to step 0;

Figure 1. An algorithm for the adaptive redundancy scheme

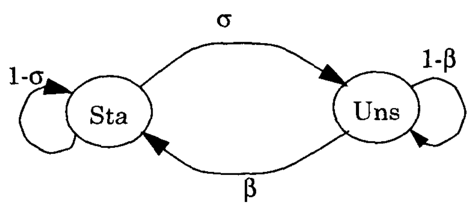

Figure 2. Two-state channel model

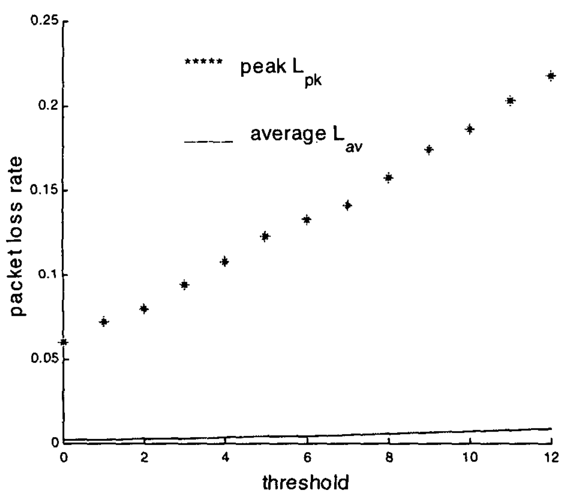

Figure 3. Packet loss rates versus SNR threshold given $M = 50$ and no redundancy packets.

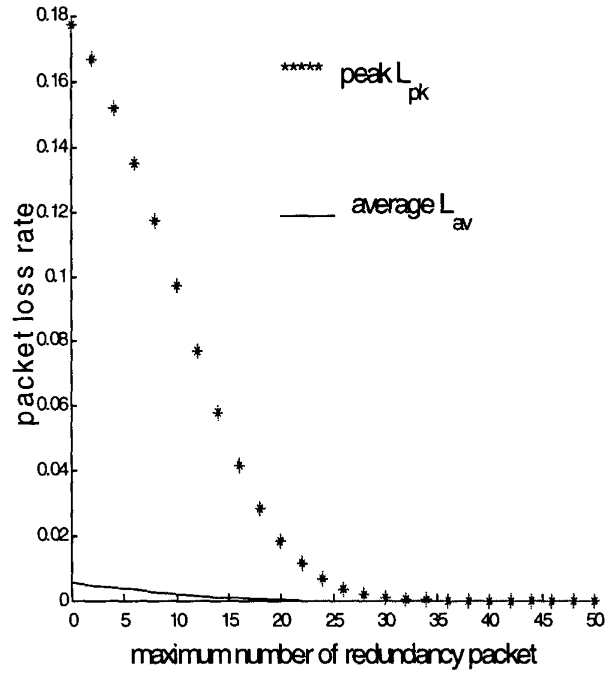

Figure 4. Packet loss rates versus the maximum number of redundancy packets for $K = 32$, $\gamma_0 = 6$ dB, $M = 50$.