An Adaptive Redundancy Technique for Wireless Indoor Multicasting

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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.

Keywords — multicasting, FEC and ARQ, channel state estimation, indoor wireless LAN.

I. INTRODUCTION

In multimedia communications, it is often to distribute a data stream to multiple radio terminals. 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 II outlines the adaptive technique for recovery of the lost packets in indoor multicast environment. In Section III, we suggest and define a set of criteria to measure the multicast performance. In section IV, 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 V.

II. 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-erasuance 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. Future 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-Solomon (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 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 a 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) ≥ K, i.e., s + r - i ≥ 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 microsecond 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. A one packet delay would have little effect on performance. A few packets delay, for instance, in some CDMA systems, can work well by a little change of the scheme. This is shown in Section IV.

If the base station receives only 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. j = N, have been transmitted. In the case, some terminals may lose 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 (R0) of redundancy packets will be updated as shown in Equation (1). If the final transmitted packet number (i = K + r) in the previous block was greater than the original planned number (K + R0), it implies that the links have now become worse than expected. Thus we initial R0 = r for the new block. If R0 = 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 R0 = R. If no any NAKs were received in the previous block, it implies that the links may be better than expected. Then, the R0 is reduced.

During the transmission of a block, the planned number of redundancy packet is adapted to the worst link and multicast 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.

### 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 × k);
2. Outer encoding: C0(N, K);
   - Initial i = 0, r = R0;
3. Do while i < N and i < K + r,
   (a) i ← i + 1;
   (b) Inner encoding (packet): Cint(n, k + h);
   (c) Send Packet i;
   (d) If NAK (one or more) is received, then r ← r + 1;
4. \[
   R0 \left\{ \begin{array}{ll}
   r & \text{if } R0 < r \leq R \\
   R & \text{if } r > R \\
   \left[ \frac{K}{P} \right] & \text{if } r = R0 \neq R
   \end{array} \right.
   \]
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 ← s + 1;
   - else Packet is erased;
4. If s + r - i < 0 and i ≤ N, then send NAK;
5. If s = K or i = N, then Outer decoding;
   - Stop for next block;
   - back to step 0;

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Fig. 1. An algorithm for the adaptive redundancy scheme
III. 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 $\Omega$ 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

$$L_{av} = \frac{1}{\Omega MK} \sum_{j=1}^{\Omega} \sum_{i=1}^{M} l_{ji}$$

(2)

The peak packet loss rate is defined as

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

(3)

which should be much greater then $L_{av}$ 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 $\eta$ 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

$$\eta = \frac{K}{K + R_{av}}$$

(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 $\eta_{cd}$ 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

$$U = \eta \frac{k}{n}$$

(5)

$$U_{cd} = \eta_{cd} \frac{k}{n}$$

(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 $T_{max}$ in WLANs as

$$T_{av} = K + R_{av}$$

(7)

$$T_{max} = K + R$$

(8)

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

$$\lambda_{av} = \frac{1}{\Omega M} \sum_{j=1}^{\Omega} \frac{1}{N_{ji}} \sum_{i=1}^{N_{ji}} \lambda_{ji}'$$

(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 $\lambda_{pk}$. 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 $\lambda_{j}'$ per terminal under the condition of $\lambda_{j}' \neq 0$, that is

$$\lambda_{pk} = \frac{1}{\Omega M} \sum_{j=1}^{\Omega} \max\{\lambda_{j1}', ..., \lambda_{jN_{j}}'\}$$

(10)

where $\Omega'$ is the number of blocks with $\lambda_{j}' > 0$.

IV. 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.

A. 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

\[
a_0(d) = a(d_0) - 10\alpha \log\left(\frac{d}{d_0}\right) + X \tag{11}
\]

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, \( \alpha \) 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 Gaussian distributed random variable with a standard deviation of \( \sigma \) dB [10].

In practice, the values of \( \alpha \) and \( \sigma \) 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 fades 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 \( T_s \) and \( T_u \) be the mean durations in stationary state and in unstationary state respectively. Let \( \tau \) be transmission time of a packet, and \( \tau \leq T_s, \tau \leq T_u \), so that no more than one transition occur in a packet transmission period. Then the transition probabilities are respectively

\[
\sigma = 1 - \exp\left(-\frac{\tau}{T_s}\right) \tag{13}
\]
\[
\beta = 1 - \exp\left(-\frac{\tau}{T_u}\right) \tag{14}
\]

![Fig. 2. Two-state channel model](image)

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 fading 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 fading. The Rayleigh probability density function for amplitude envelope \( r \) is given by

\[
f_r(r) = \begin{cases} \frac{r}{w} \exp\left(-\frac{r^2}{2w^2}\right) & 0 \leq r \leq \infty \\ 0 & r < 0 \end{cases} \tag{15}
\]

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 \( a_0 = 10 \log_{10} 2w^2 \). The transformation equation can be written as

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

Since \( r \geq 0 \) and \( w \geq 0 \), we have

\[
r = \sqrt{2w} \exp\left(\frac{a - a_0}{c}\right) \tag{17}
\]

where \( c = 20/\ln 10 = 8.686 \). Taking the derivative, we have

\[
\frac{dr}{da} = \sqrt{2w} \exp\left(\frac{a - a_0}{c}\right) \tag{18}
\]

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

\[
f_a(a) = f_r(r) \left| \frac{dr}{da} \right| = \frac{2}{c} \exp\left(\frac{2(a - a_0)}{c}\right) \exp\left\{-\exp\left(\frac{2(a - a_0)}{c}\right)\right\} \tag{19}
\]

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

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

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 \( \gamma \) at a mobile station is

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

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

\[
\bar{\gamma}(d) = \gamma(d_0) - 10\alpha \log_{10} \left(\frac{d}{d_0}\right) + X \tag{23}
\]
\[ \gamma(d, t) = 5(d) + Y(t) \]  

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 \( d_{max} = 60 \) 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.

**B. Simulation example**

As known, the packet erasure rate depends on both the channel state, i.e. signal-to-noise ratio \( \gamma \), 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 \( \gamma_0 \) 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 \( \gamma_0 \), 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 \( \gamma(d) > \gamma_0 \).

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 \( \langle L_{av} \rangle \) and peak packet loss rate \( \langle L_{pk} \rangle \) as a function of the threshold of signal-to-noise ratio \( \langle \gamma_0 \rangle \) 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 \( L_{pk} = 0.5295 \times 10^{-3} \).

**C. Comparisons**

To compare multicast performance between different techniques, we also simulate a forward error correction (FEC) protocol and an automatic repeat 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 \( \lambda_{av} \) and \( \lambda_{pk} \) 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 sta-
tion receives an NAK, it increases the planned number of
redundancy packets by 2. The simulation results of the pro-
posal 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.

V. Conclusion

In this paper, the indoor radio channel is briefly reviewed
and a simulation model for the slow time-variable chan-
nel 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 re-
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 signifi-
cantly 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 densi-
ties $\lambda_{av}$ and $\lambda_{pk}$ with a factor of 5.2 and 1.4 respectively, al-
though the proposed protocol uses a little less redundancy.
(3) This scheme can potentially solve the problems of feed-
back 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.

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rison of indoor radio propagation characteristics at 910 MHz and

Table I

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<th>FEC</th>
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<td><strong>density $\lambda_{pk}$</strong></td>
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<td>0.0203</td>
<td>0.0292</td>
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<tr>
<td><strong>throughput $\eta_{cd}$</strong></td>
<td>0.5</td>
<td>0.7992</td>
<td>0.7824</td>
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<td><strong>average block</strong></td>
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<tr>
<td><strong>latency $T_{av}$</strong></td>
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<td>40.0409</td>
<td>40.8987</td>
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<tr>
<td><strong>maximum block</strong></td>
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<tr>
<td><strong>latency $T_{max}$</strong></td>
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<td>64</td>
<td>85</td>
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