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Dynamic Resource Allocation in Time-Varying Ultra Wide Band Channels

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Abstract— **When considering applications requiring QoS it is crucial to adopt in the MAC adaptive resource allocation strategies in order to reduce system load. This paper proposes an analytical procedure for optimizing resource allocation in the case of mobile Ultra Wide Band (UWB) users in an indoor environment. The proposed analysis incorporates both the user mobility model and the UWB time-varying channel model. The proposed resource allocation algorithm is based on an iterative algorithm which is capable of minimizing required capacity according to both the state of the channel and the requested QoS. Results of simulations show that within reasonable speed limits which can be quantified, the algorithm is capable of providing the system with an excellent fit of assigned resource vs. required QoS.**

Keywords- Quality of Service (QoS), Resource Allocation, Error Protection, Medium Access Control (MAC), Ultra Wide Band (UWB), Time-Varying Channels.

I. INTRODUCTION

Two basic problems in the design of Medium Access Control (MAC) protocols with Quality of Service (QoS) are the efficient management of the resource, and the need for fulfilling QoS requirements despite the unpredictable behavior of the channel. When the MAC is capable of adapting resource allocation to the time-varying nature of the channel, the above problems can be strongly mitigated. In the ideal scenario where all the transmitted data units correctly reach destination, the MAC module is able to fulfil QoS constraints by simply reserving a portion of the available capacity for each source which is admitted in the system. When considering a channel which introduces errors on the transmitted units, mechanisms of either retransmission (i.e. Automatic Repeat on reQuest, ARQ) or error correction (i.e. Forward Error Correction, FEC), or both, may be needed in the MAC in order to improve system efficiency [1]-[3]. ARQ mechanisms are based on the repetition of corrupted MAC Protocol Data Units (MACPDUs) [4]. FEC schemes introduce redundancy in each MACPDU in order to provide the receiver with the capability to correct a certain number of errors [5]. FEC schemes may also be combined within ARQ mechanisms, giving rise to the so-called Hybrid ARQ [6]. ARQ based solutions introduce delays which might be incompatible with real-time features. On the other hand, FEC approach has the drawback of requiring overhead transmission, and therefore introduces efficiency loss. In this paper, we propose an algorithm which optimizes transmission efficiency at the MAC level by adapting the error protection to both channel status and QoS constraints. Performance of the proposed algorithm in the case of a slowly time-varying Ultra Wide Band (UWB) channel is discussed. A simple mobility model for the receiver is introduced, and the capability of the proposed algorithm to fulfil QoS for different values of the receiver speed is evaluated. The paper is organized as follows. Section II illustrates the reference scenario. Section III describes the proposed algorithm for optimizing error protection and resource allocation at the MAC level. Section IV introduces the receiver mobility model. Section V describes the empirical model which is introduced for emulating propagation over the UWB channel. Section VI evaluates performance of the proposed algorithm in the case of a slowly time-varying UWB channel and, finally, Section VII contains the conclusions.

II. REFERENCE SCENARIO

Each traffic source is characterized by two sets of parameters. The first, denoted *Tspecs*, collects parameters describing source traffic activity. In particular, we assume that *Tspecs* consists of the Dual Leaky Bucket (DLB) parameters described in [7], i.e. the peak rate of the flow p (bits/s), the average rate of the flow r (bits/s), the token buffer dimension b (bits), and the maximum source packet size M (bits). The second set of parameters, denoted *Qspecs*, defines two QoS requirements: the maximum tolerable end-to-end delay D_{MAX} (s), and the minimum percentage of packets F (0÷100) required at destination within D_{MAX}. Note that the same parameters are used for both real-time and non-real-time services, with no explicit need for defining classes of traffic. The proposed MAC protocol uses fixed-size MACPDUs of L_{PDU} bits, composed of a fixed-size header of L_H bits and a fixed-size payload of L_P bits. The header contains the information used by the MAC for managing the transmission of a MACPDU. It may contain error detection codes such as CRC, but no FEC. The payload conveys bits originating from source packets segmentation, and redundancy bits eventually introduced by the FEC. In other words, we assume that the introduction of corrective overhead is realized by removing the corresponding bits from the MACPDU payload. The payload is therefore composed of two parts: a FEC field of L_{FEC} bits and an effective payload for user data of $L_{EFF}=L_P-L_{FEC}$ bits. Note that while L_{FEC} and L_{EFF} may vary in different MACPDUs, L_{PDU}, L_H, and L_P are fixed. With reference to the ARQ, a Selective Repeat (SR) strategy is implemented in order to avoid unnecessary re-transmissions which could affect simulation results. Resource allocation is based on the definition of a MAC frame of D_F seconds. Dsys is the maximum system delay introduced by the MAC for the transmission of a single MACPDU. We assume a slowly time-varying channel characterized by a Bit Error Rate (BER) indicated by p_b. We also assume that the transmitter knows the exact value of p_b by estimation of the reverse channel. We discard incorrigible errors in the header field, and restrict the present analysis to error protection on the payload.

III. OPTIMIZATION OF ERROR PROTECTION

All packets generated by the source are stored in a buffer before being transmitted. We assume the MAC module to extract packets at constant rate C_b (bits/s). The C_b value must be evaluated at admission time according to both size of the buffer, and the delay constraint which is required by the source. Given a source activity model (i.e. for given *Tspecs*), the trade-off between delay and capacity can be expressed in analytical terms by introducing two functions: the Delay function $\Delta(Tspecs,C_b)$ and the Capacity function $\mathrm{X}(Tspecs,D)$. The Delay function evaluates the end-to-end delay D when the MAC reserves the capacity C_b:

$$\Delta(Tspecs,C_b)=\begin{cases}\left[\dfrac{p-C_b}{p-r}(b-M)+M\right]\cdot\dfrac{1}{C_b}+D_{sys} & when\ p>r\\ M/C_b+D_{sys} & when\ p=r\end{cases} \tag{1}$$

The Capacity function evaluates the capacity in bits/s which is necessary for guaranteeing a maximum end-to-end delay D:

$$\mathrm{X}(Tspecs,D)=\begin{cases}\dfrac{p\cdot b-r\cdot M}{(D-D_{sys})\cdot(p-r)+b-M} & when\ p>r\\ M/(D-D_{sys}) & when\ p=r\end{cases} \tag{2}$$

The lower bound for C_b is given by the average emission rate r, since $C_b<r$ causes overflow on the source buffer. The value of C_b should be however increased if the corresponding delay $D_0=\Delta(Tspecs,r)$ is higher than D_{MAX}. The rule for evaluating C_b in absence of any error protection mechanism is:

$$C_b=\mathrm{X}\left(Tspecs,\min\{D_{MAX},\Delta(Tspecs,r)\}\right) \tag{3}$$

When considering the presence of an error protection mechanism, the rule in (3) must be modified in order to take into account the presence of ARQ, i.e:

$$C_b(N_R)=\mathrm{X}\left(Tspecs,\min\{D_{MAX}-N_R\cdot RTT,\Delta(Tspecs,r)\}\right) \tag{4}$$

where N_R represents the maximum number of retransmissions allowed by the ARQ scheme, and RTT is the estimated round-trip-time, i.e. the time necessary for the retransmission request plus the time needed for the retransmission of a MACPDU. Note that the capacity in (4) is expressed as a function of N_R, which must satisfy:

$$0\le N_R\le N_R^{(\max)}=\left\lfloor\frac{D_{MAX}-D_{sys}}{RTT}\right\rfloor \tag{5}$$

One can thus define the vector **C** containing the values of capacity corresponding to the different N_R values:

$$\mathbf{C}=[C_b(0),C_b(1),\ldots,C_b(N_R^{(\max)})]^T \tag{6}$$

After a source packet is completely extracted from the buffer, the MAC proceeds with packet segmentation, i.e. the separation of the original packet into different MACPDUs. The number of MACPDUs which are generated from a single packet depends on both the size of the original packet, and the size of the available payload L_{EFF} on each MACPDU. The L_{EFF} value depends on the amount of FEC which is introduced on each MACPDU. Two basic problems must be solved however in order to design the FEC for each MACPDU. First, the MAC must be capable of calculating the L_{FEC} value which guarantees a desired level of protection on each transmitted data unit. Then, the MAC must evaluate how much protection is required on each MACPDU for fulfilling the QoS constraint given by F.

The introduction of redundancy bits inside the payload guarantees a certain corrective capability k to each transmitted data unit, i.e. up to k binary errors can be corrected at the receiver after transmission on the noisy channel. For a given value of k, the probability P_L to lose a MACPDU during transmission is thus given by:

$$P_L=1-\sum_{j=0}^{k}\binom{L_P}{j}\cdot p_b{}^{j}\cdot(1-p_b)^{L_P-j} \tag{7}$$

By applying the DeMoivre-Laplace theorem, the expression in (7) can be overturned as follows:

$$k\approx\left\lceil L_P\cdot p_b+\sqrt{2\cdot L_P\cdot p_b\cdot(1-p_b)}\cdot erfc^{-1}(2\cdot P_L)\right\rceil \tag{8}$$

where *erfc(x)* is the complementary error function. Depending on the adopted FEC scheme, the corrective capability in (8) will be guaranteed by L_{FEC} bits of FEC. In the case of a Reed-Solomon FEC code using a word length of 8 bits, one has:

$$L_{FEC}=\min\left\{8\cdot\left\lceil\frac{L_P}{8}\right\rceil,16\cdot k\right\} \tag{9}$$

The following step is the evaluation of the P_L value which satisfies the constraint given by F. This problem cannot be solved in a closed form, since the relationship between F and P_L depends on the number of MACPDUs which are generated from the source packet, and this quantity depends in turn on the L_{FEC} value. We propose thus an iterative algorithm based on successive approximations. Each time a source packet is ready for segmentation, the MAC evaluates the actual p_b value and initializes the algorithm by calculating the following values:

$$P_E(p_b)=1-(1-p_b)^{L_P} \tag{10}$$

$$P_{L,0}(F,M)=1-\left(\frac{F}{100}\right)^{\frac{1}{\lceil M/L_P\rceil}} \tag{11}$$

where $P_E(p_b)$ in (10) represents the MACPDU loss probability in absence of any protection mechanism, i.e. with no FEC and no ARQ, and $P_{L,0}(F,M)$ in (11) is the target packet loss probability P_L for the case $L_{EFF}=L_P$, i.e. $L_{FEC}=0$. If $P_{L,0}(F,M) \geq P_E(p_b)$ all MACPDUs can be transmitted without protection, i.e. $L_{EFF}=L_p$ and $N_R=0$, since the channel produces an average packet loss rate which is acceptable in terms of QoS fulfilment. Oppositely, when $P_{L,0}(F,M)<P_E(p_b)$, an error protection scheme becomes necessary in order to increase MACPDU robustness against errors. In this case, the following iterative steps are introduced for computing L_{FEC} as a function of the number of retransmissions N_R:

A) The MAC evaluates the target MACPDU loss probability in absence of FEC:

$$P_{L,A} = \left[1-\left(\frac{F}{100}\right)^{\frac{1}{\lceil M/L_P \rceil}}\right]^{\frac{1}{(1+N_R)}} \tag{12}$$

If $P_{L,A} \geq P_E(p_b)$, no FEC is required, i.e. $L_{FEC}(N_R)=0$ and the remaining steps of the algorithm are skipped. Otherwise, the *old* target MACPDU loss probability oP_L is set equal to $P_{L,A}$.

B) The new value of corrective capability is evaluated through (8) by considering oP_L as the target packet loss probability P_L. Then, the corresponding L_{FEC} value is computed according to the selected FEC scheme, e.g. by using (9) in the case of a Reed-Solomon FEC code. Given L_{FEC}, the MAC evaluates the new MACPDU loss probability nP_L:

$$nP_L = \left[1-\left(\frac{F}{100}\right)^{\frac{1}{\lceil M/(L_P-L_{FEC}) \rceil}}\right]^{\frac{1}{(1+N_R)}} \tag{13}$$

C) nP_L and oP_L are compared.

- If $nP_L \neq oP_L$, the target MACPDU loss rate has been affected by the new FEC size. In this case, oP_L is set equal to nP_L and the algorithm must be repeated from step B). The procedure converges thanks to the non-linear dependence on L_{FEC} of the function in (13).
- If $nPL = oPL$ the algorithm is concluded since FEC has been correctly designed.

Once L_{FEC} is fixed, one can evaluate the effect of segmentation due to the presence of both a MACPDU header and a FEC field. In particular, the MAC module creates a vector $\mathbf{C_{EFF}}$ containing the effective values of the capacity $C_{eff}(N_R)$ which is required at the MAC level for a given N_R:

$$\mathbf{C_{EFF}} = \left[C_{eff}(0), C_{eff}(1), \ldots, C_{eff}(N_R^{(\max)})\right]^T = \mathbf{\Psi}\,\mathbf{C} \tag{14}$$

where:

$$\mathbf{\Psi} = \begin{bmatrix} \psi(0) & 0 & \cdots & 0 \\ 0 & \psi(1) & \cdots & 0 \\ \vdots & \vdots & \ddots & \vdots \\ 0 & 0 & \cdots & \psi(N_R^{(\max)}) \end{bmatrix} \tag{15}$$

$$\psi(N_R) = \frac{L_{PDU}}{M} \cdot \left\lceil \frac{M}{L_P - L_{FEC}(N_R)} \right\rceil \tag{16}$$

The final step consists in expressing the effective capacity in terms of the number N_{PDU} of MACPDUs per frame which should be reserved for the source:

$$\mathbf{N_{PDU}} = \left[N_{PDU}(0), N_{PDU}(1), \ldots, N_{PDU}(N_R^{(\max)})\right]^T = \mathbf{\Phi}\,\mathbf{C_{EFF}} \tag{17}$$

where:

$$\mathbf{\Phi} = \begin{bmatrix} \phi(0) & 0 & \cdots & 0 \\ 0 & \phi(1) & \cdots & 0 \\ \vdots & \vdots & \ddots & \vdots \\ 0 & 0 & \cdots & \phi(N_R^{(\max)}) \end{bmatrix} \tag{18}$$

$$\phi(N_R) = \frac{D_f}{L_P + L_H}\left(1+\sum_{j=1}^{N_R}\left\{\left[1-\left(\frac{F}{100}\right)^{\frac{1}{\lceil M/(L_P-L_{FEC}(N_R)) \rceil}}\right]^{\frac{j}{(1+N_R)}}\right\}\right) \tag{19}$$

The sum in (19) takes into account the average number of MACPDUs per frame which will be retransmitted by the ARQ mechanism. The vector $\mathbf{N_{PDU}}$ contains the average number of MACPDUs per frame for a source with Q_{specs} which generates traffic according to T_{specs}. In order to optimize transmission efficiency, the MAC must simply select the lowest $N_{PDU}(N_R)$ component of $\mathbf{N_{PDU}}$. The result of such a selection allows the MAC to evaluate the FEC size for each MACPDU, the maximum number of retransmissions which must be allowed by the ARQ mechanism, and the average number of MACPDUs which must be allocated on each frame, i.e. the minimum amount of resource which is required for fulfilling the QoS. Note that the selection of N_R depends on the actual p_b value; The procedure should therefore be repeated at each observed bit error rate variation.

IV. Mobility Model

In order to verify the robustness of the proposed algorithm in the case of a slowly time-varying channel, we simulated the mobility of the receiver in a multipath environment. A rectangular area of $L_A{\cdot}L_B$ m^2 is considered. This area contains a fixed grid of $N_A{\cdot}N_B$ *spatial points*. The distance between two adjacent spatial points of the grid, i.e. the spatial resolution of the area, is equal to dL (m). The transmitter is located in the centre of the area, while the receiver moves with uniform speed v (m/s). The receiver moves along a path composed of a sequence of n_S rectilinear segments. The length L_j of the j-th segment is a random variable uniformly distributed between L_{min} and L_{max}. The j-th segment consists of a sequence of L_j/dL spatial points. The receiver remains in the same spatial point for a time $dt=dL/v$, then moves to the next one in the segment. The direction D_j of the j-th segment is a random variable which can assume one of 8 possible values, corresponding to the 8 possible directions on the grid (North, North-East, East, South-East, South, South-West, West, and North-West). All lengths and directions are independent of each other. When the receiver reaches the border of the rectangular area, it remains in the same spatial point until the next change of direction. Moreover, the distance d between transmitter and receiver cannot be smaller than a fixed value D_{min}. When the receiver reaches a spatial point at distance D_{min} from the transmitter, it starts moving on a circular path at the same distance D_{min} from the transmitter until the next change in direction.

V. The UWB Channel Model

We assume that each spatial point of the grid introduced in Section IV is characterized by a specific value of the BER. In particular, we assume the j-th spatial point of the grid to be characterized by the BER value $p_b^{(j)}$ given by:

$$p_b^{(j)} = \frac{1}{2} erfc\left(\sqrt{SNR^{(j)}/2}\right) \tag{20}$$

where $SNR^{(j)}$ is the signal to noise ratio in the j-th spatial point. The value of $SNR^{(j)}$ is computed by fixing the reference signal to noise ratio SNR_0 at distance d_0=1m, and then by introducing the statistical path-loss model for the UWB indoor channel which is described in [8]:

$$\begin{aligned} PL^{(j)}(dB) = & [PL_0 + 10 \cdot \mu_\gamma \cdot \log_{10} d^{(j)}] \\ & + [10 \cdot n_1 \cdot \sigma_\gamma \cdot \log_{10} d^{(j)} + n_2 \cdot \mu_\sigma + n_2 \cdot n_3 \cdot \sigma_\sigma] \end{aligned} \tag{21}$$

where PL_0 (dB) is the average path loss at d_0, $d^{(j)}$ is the distance between the transmitter and the j-th spatial point of the grid, μ_γ is the mean value of the path-loss exponent, σ_γ is the standard deviation of the path-loss exponent, μ_σ is the mean value of the small scale fading in dB, σ_σ is the standard deviation of the small scale fading in dB, and n_1, n_2, n_3 are three independent zero-mean Gaussian random variables with unitary standard deviation. Common values for PL_0, μ_γ, σ_γ, μ_σ, and σ_σ, are given by specific tables corresponding to different propagation scenarios (LOS Commercial, LOS Residential, NLOS Commercial, and NLOS Residential). One obtains:

$$SNR^{(j)}(dB) = SNR_0(dB) - PL^{(j)}(dB) + PL_0 \tag{22}$$

VI. Simulation

Performance of the proposed algorithm was verified by simulating the mobility of the receiver for different values of the constant speed v. The reference scenario consists of a rectangular room with L_A=L_B=10m. The spatial resolution is dL=0.05m, i.e. N_A=N_B=200. The dL value is equal to the spatial resolution used for developing the presented model for the UWB channel [8]. L_{min} and L_{max} are fixed to 4m and 5m, respectively. D_{min} is equal to 2m. Path loss is evaluated through the model in (21), with the parameters given in Table 1. The reference signal to noise ratio at d_0=1m is set to SNR_0=30dB. Three different traffic sources are taken into account for the simulations. All these sources generate constant bit rate traffic with the following *Tspecs*: p = r = 512 kb/s, M=400 bits, b=2000 bits. *Qspecs* parameters for the three sources are listed in Table 2. The first source (*source A*) represents a typical real-time application and is characterized by the lowest values for both D_{MAX} and F. The third source (*source C*) represents a typical non-real-time application and is characterized by the highest values for both D_{MAX} and F. The second source (*source B*) represents an intermediate case. In such a scenario, performance of the algorithm was verified by simulating the movement of the receiver along two representative paths (Figs. 1 and 2). In both cases, the total distance covered by the receiver is D_{tot}=60m. For each spatial point belonging to these paths, the distance between transmitter and receiver is evaluated and the corresponding BER value is estimated according to (20) (Figs. 3 and 4). Robustness of the proposed algorithm to BER variations is verified by performing different simulations with increasing values of the receiver speed v. For each v value, we calculate the percentage of source packets delivered to destination within D_{MAX} and compare this quantity with the requested QoS parameter F. Results of simulation are presented in Fig.5 for *Case 1*, and in Fig.6 for *Case 2*. In both cases, we observe that the algorithm is capable of guaranteeing the required QoS as far as the speed of the receiver remains below a specific threshold v_{max}. In particular, we find v_{max}=1.1m/s for *Case 1* and v_{max}=0.6m/s for *Case 2*. When v is higher than v_{max}, we observe a decrease in performance which is not acceptable in terms of QoS fulfilment, leading to conclude that the proposed algorithm is capable of guaranteeing the requested QoS up to a certain limit of mobility degree. In a scenario with increased mobility, QoS cannot be guaranteed because of the lack of correlation between channel condition at segmentation time and channel condition at transmission time. The estimated values of v_{max}, are however compatible with the adopted indoor WLAN scenario.

VII. Conclusion

An analytical approach for optimizing resource allocation at the MAC layer for traffic sources requiring QoS was proposed. The resulting algorithm maximizes transmission efficiency by selecting and dimensioning an error protection mechanism which takes into account both channel status and QoS constraints. In the case of propagation over a slowly time-varying UWB channel, the proposed algorithm shows to be capable of guaranteeing QoS fulfilment by adapting error protection to channel performance.

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TABLE I. PARAMETER VALUES IN PATH LOSS MODEL

Path Loss Parameter	Symbol	Value
Average Path Loss at d_0	PL_0	47.2 dB
Mean value of the Path Loss exponent	μ_γ	1.82
Standard deviation of the Path Loss exponent	σ_γ	0.39
Mean value of the small scale fading	μ_σ	1.5
Standard deviation of the small scale fading	σ_σ	0.6

TABLE II. QSPECS PARAMETERS

Qspecs parameter	Symbol	A	B	C
Maximum tolerable end-to-end delay	D_{max}	0.5 s	2 s	5 s
Minimum tolerable percentage of packets received within D_{max}	F	99.9%	99.99%	99.999%

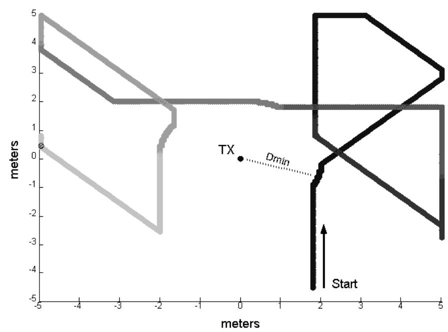

Figure 1. Path of the receiver in *Case 1*

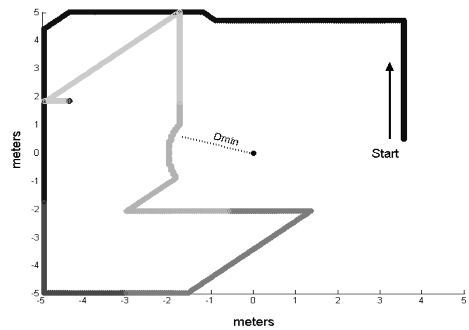

Figure 2. Path of the receiver in *Case 2*

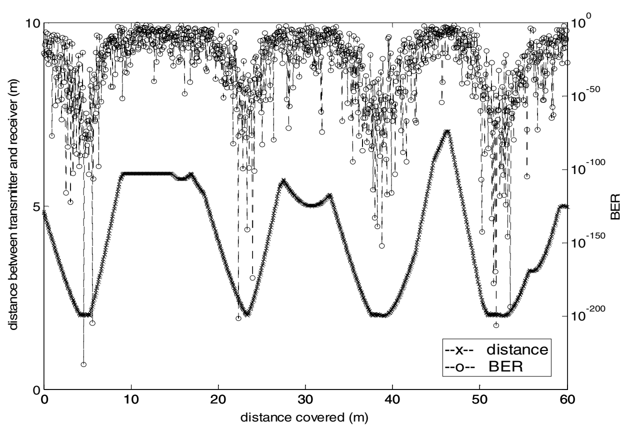

Figure 3. Distance between transmitter and receiver (crosses) and BER values (circles) throughout the path of *Case 1* (see Fig.1).

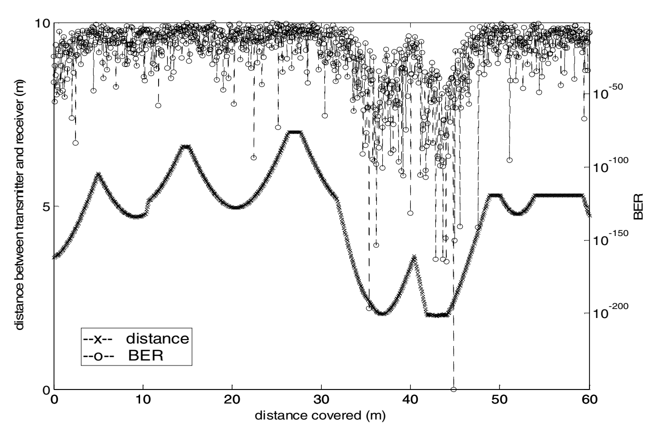

Figure 4. Distance between transmitter and receiver (crosses) and BER values (circles) throughout the path of *Case 2* (see Fig.2).

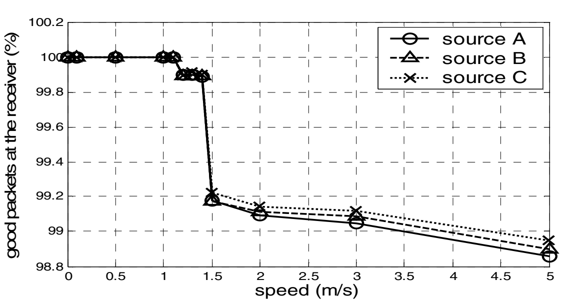

Figure 5. Percentage of received good packets vs. receiver speed for *Case 1* (see Fig.1). Circles are for *Source A*, triangles are for *source B*, and crosses are for *source C*.

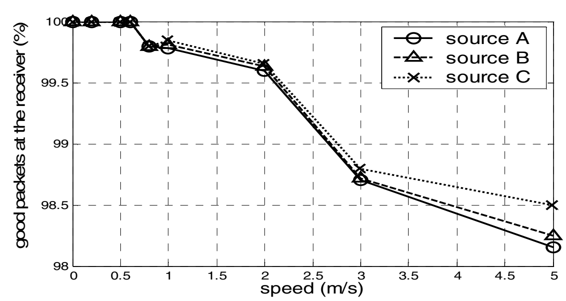

Figure 6. Percentage of received good packets vs. receiver speed for *Case 2* (see Fig.2). Circles are for *Source A*, triangles are for *source B*, and crosses are for *source C*.