THE IMPACT OF TIME VARYING CHANNELS ON MAC LAYER IN IEEE 802.11 NETWORKS

Ali Nassar and Michel Kadoch
École de Technolgie Supérieure (ÉTS)
1100 street Notre-Dame west, Montreal, QC, Canada, H3C 1K3
E-mail: alinassar@gmail.com, michel.kadoch@etsmtl.ca

Abstract— **Channel variation plays a crucial role in data transmission. This variation may affect wireless network protocols like IEEE 802.11 and hence the throughput. We propose the combination of a simple channel estimation method based on basis expansion model with channel prediction to improve the performance of the IEEE 802.11 network's MAC protocols in a time-varying channel environment. When many users try to access the medium, the time separation (delay) between two successive transmissions belonging to the same user may vary due to the fairness mechanism used by the medium access protocols. During this delay, the channel may change and efficient channel modeling and estimation methods should be applied. Time varying channel's estimation and prediction have been proposed in order to mitigate the effect of channel variation on the system's throughput. This proposition mitigates also the impact of pilot signals loss, due to collisions, on the system's throughput. Pilot signals are known to the receiver and they are used to predict the future values of the time-varying channel for the following data reception period.**

Keywords— **Time varying channels, IEEE 802.11, distributed coordination function(DCF).**

I. Introduction

In many telecommunication applications, scattering and users mobility are the major sources of propagation channel variation, while line of sight may not be the dominant component of the received signal. Besides, recent new generation networks standards demand higher performance in an environment with terminal mobility. For example, Long Term Evolution (LTE) standards demand high performance for speeds of 15 to 120 km/h[1].

Different models have been suggested to mitigate the effect of channel variability. Channel can be expanded over a basis of complex exponentials [2], where it can be expressed as a superposition of time varying bases with minimum number of time invariant coefficients, which has the advantage of reducing the spectral leakage. Channel variation is relative to data transmission rate. For time varying channels, periodic pilot signals are essential.

Lately, IEEE 802.11 wireless Local Area Networks (LAN) have grown rapidly due to their high data rates and because they operate in unlicensed spectrum. IEEE 802.11 is based on the Orthogonal Frequency Division Multiplexing (OFDM) technology. In OFDM, the bandwidth is divided into subcarriers. This subdivision solves the distortion problem caused by frequency selective channels, while channel still may vary with time due to terminal mobility. For successful reception, receiver should have information about the new values of the channel.

In IEEE 802.11, many users try to access the medium. MAC protocols may guarantee a fair use of the medium between users. However, this leads to delays between transmissions of the same user to take place. Therefore, the problem arises when the terminal moves from one point to another causing channel variations. So, long delays may cause faster channel variations, and hence efficient channel estimation and prediction are required.

In a single user scenario, channel can be considered always known because the pilot signal is always available and no collisions take place. Many works discuss the issue of time varying channels such as [6],[5] and [2]. In the IEEE.802.11 scenario, pilot signals may collide with other users transmissions. These collisions in addition to the delays due to access protocols may cause higher data loss due to the absence of channel information.

We propose the combination of an efficient channel estimator based on basis expansion model (BEM) with a statistical channel model to predict and estimate the propagation channel in the absence of pilot signals due to collisions.

According to [5], time varying channel is characterized based on a set of basis functions that modulate certain time invariant coefficients. The objective of this work is to estimate these invariant coefficients during the pilot reception and to use them to predict and estimate the time varying channel in the absence of pilot signals during the following data reception. If the following pilot signal collides, this estimation may still be valid. Collisions may happen due to the nature of the protocol DCF and retransmissions. The proposed method of coefficients estimation and prediction aims to mitigate the impact of pilot signals collisions and delays on the throughput of the IEEE 802.11 network. We will show also the impact of the time varying channels on the system's throughput.

The system model will be described in section II, where, time varying channel characterization and signal model will be explained in sections II-A and II-B respectively. Section II-C describes the DCF protocol. Channel coefficients prediction and extrapolation are described in section III. Simulations and results are shown in section IV. Throughputs were compared for different simulation scenarios, and

ISBN: 978-1-61804-018-3

results have been depicted as a function of different system parameters such as channel coherence time.

II. System Model

A. Channel Characterization

The continuous-time model of transmission over time varying channel can be expressed as

$$y(t) = \int_{-\infty}^{+\infty} h(t;\tau)x(t-\tau)d\tau + \omega(t) \quad (1)$$

where $x(t)$ is the transmitted signal, $h(t;\tau)$ denotes the time varying channel impulse response and ω(t) is the zero mean additive white Gaussian noise. When the channel is limited in Doppler spread and the input signals are band-limited to $|f| \leq (F_t + F_d)/2$, where F_t is the transmission bandwidth and F_d is the bandwidth of the random process associated with variation in t (Doppler spread), the sampling theorem establishes

$$h(t;\tau) = \sum_k h_k(\tau)\phi(F_d t - k) \quad (2)$$

where

$$h_k(\tau) = h(k/F_d;\tau) \quad (3)$$

is the impulse response of a stationary channel, and $\phi(t)$ is the interpolating function. This means that $h(t;\tau)$ can be perfectly interpolated in the time domain using the set of functions $\{\phi(F_d t - k)\}$. On the other hand, the stationary coefficients $\mathbf{h_k}(\tau)$ incorporate all relevant Channel State Information (CSI) and can be shown to be band-limited in the delay variable τ to $|f| \leq F_t/2$. Hence, they may be in turn interpolated as

$$h_k(\tau) = \sum_q h_{k,q}\phi(F_t\tau - q) \quad (4)$$

Finally, the complete channel expansion reduces to

$$h(t;\tau) = \sum_{k,q} \mathbf{h}_{k,q}\phi(F_t\tau - q)\phi(F_d t - k) \quad (5)$$

from which we obtain the set of coefficients $\{h_{k,q}\}$ whose statistics completely characterize the channel. In this paper we focus on the flat fading case. This gives rise to one-dimensional problem applicable to the frequency subdivisions (carriers) of the OFDM scheme, where each frequency carrier is considered a fading channel by itself. In this case, the channel response simplifies to

$$h(t) = \sum_k h_k\phi(F_d t - k) \quad (6)$$

B. Signal Model

The simplified input-output relation in the flat fading case when transmitting a pilot signal can be set as:

$$y(t) = h(t)x_p(t) + \omega(t) \quad (7)$$

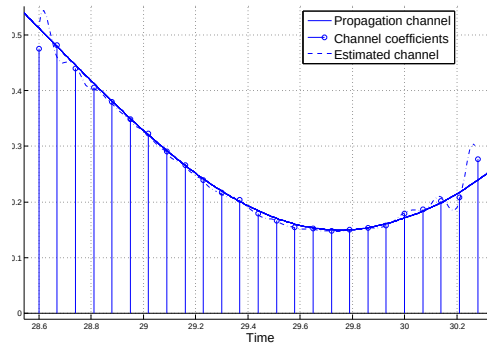

Fig. 1. Channel estimation by the method BEM.

where $y(t)$ is the received signal at the time instance t, $h(t)$ is the channel response at the instance t, $x_p(t)$ is the transmitted pilot signal, and $\omega(t)$ is the additive white Gaussian noise received at the time instance t. Here, we only consider the flat fading channel case. The discrete notation of the M received samples can be written as follows:

$$\begin{aligned} \mathbf{y}(M) &= \mathbf{x}_p(M) \odot \mathbf{h}(M) + \boldsymbol{w}(M) \\ &= \mathbf{X}_p\mathbf{h}(M) + \boldsymbol{w}(M) \\ &= \mathbf{X}_p\mathbf{S}\mathbf{a} + \boldsymbol{w}(M) \end{aligned} \quad (8)$$

Where $\mathbf{x}_p$ is the pilot signal vector, $\mathbf{x}_p \in \Re^{M\times 1}$, the $\odot$ operator is the dot product operator(element by element multiplier), $\mathbf{X}_p$ is an $M \times M$ matrix containing the elements of the vector $\mathbf{x}_p$ in its diagonal, while the rest of elements is set to zero. $\mathbf{a}$ is the $Q \times 1$ channel coefficients vector, and $\mathbf{S}$ is a long vertical matrix formed by Q columns, each column contains a shifted version of the channel interpolating pulse $\phi(t)$. The vector $\mathbf{a}$ contains the coefficients we want to estimate. So, by applying the Mean Square Error (MSE) to equation (8), we can write:

$$\mathbf{a} = (\mathbf{X}_p\mathbf{S})^\dagger\mathbf{y} \quad (9)$$

while † is the matrix pseud-inverse operator.

C. Distributed Coordination Function (DCF)

The DCF medium access method is based on the Carrier Sensing Multiple Access/Collision Avoidance principle (CSMA/CA) where a terminal wishing to transmit senses the medium and waits for a period of time known as Distributed Interframe Spacing (DIFS), whose duration is 128 microseconds, then it transmits if the medium is still free (not occupied by other users). In order to reduce collision probability between terminals' transmissions, the terminal is allowed to transmit only if its carrier sensing mechanism determines that the medium has not been occupied for at least DIFS time plus a random backoff interval. If the transmitted packets are correctly received, the receiver sends an acknowledgment (ACK) after a fixed period of time referred to as SIFS (Short Interframe Spacing) whose duration is 28 microseconds. SIFS is designed to be smaller than DIFS so ACK frame is given higher priority. If the ACK is not received by the transmitter, a collision is assumed to have occurred. The transmitter attempts to resend the packets when the channel is free. The backoff

ISBN: 978-1-61804-018-3

time is randomly chosen from the interval [0,CW], where CW refers to contention window. After each collision, the CW is doubled. Figure (2) shows the DCF method and figure (3) shows the DCF method when a collision takes place.

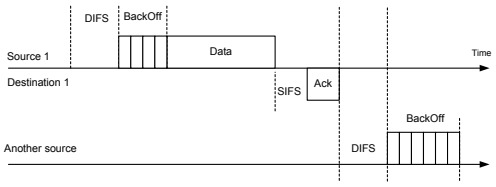

Fig. 2. The DCF used in IEEE 802.11.

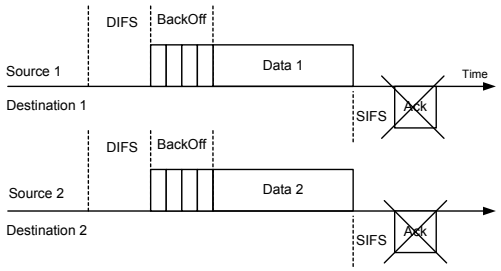

Fig. 3. DCF where a collision happens and data of both sources should be retransmitted.

III. Channel coefficients prediction

In the previous sections, channel coefficients were estimated within the observation period, *i.e.* the duration of pilot signal reception. In a medium without terminal mobility, the estimated coefficients remain constant during the subsequent time slots corresponding to data reception. While in DCF with terminals mobility, when a terminal waits for its turn to transmit, the propagation channel may change. So, the coefficients estimated during pilot reception are no longer valid but they can be used to predict the channel coefficients for the following data reception periods in order to avoid retransmissions of pilot signals and hence throughput decrease.

There are many methods for signal prediction. Here, we present briefly the prediction by extrapolation [4]. For a set of observations, h, known for all $t < T$, the predicted value at $t = T + k$ is denoted by $\hat{h}_T(k)$ and it can be expressed as the conditional expectation as follows:

$$\hat{h}_T(k) = \mathbb{E}(h_{T+k}/h_T, h_{T-1}, h_{T-2}, ...) \qquad (10)$$

where k is the prediction horizon. In our case, the set of observations is the set of channel coefficients estimated during pilot reception. First, these observations are used to find the prediction filter's weights. Figure (4) shows the prediction filter. Once these weights are known, future channel coefficients can be predicted. The values of the filter weighting vector α are calculated recursively by the method mean square error (MSE) during the pilot signal reception.

For an ARMA model, we can write the successive predicted values as:

$$\begin{aligned}
\hat{h}_T(1) &= \alpha_1 h_T \\
\hat{h}_T(2) &= \alpha_1 h_{T+1} + \alpha_2 h_T = \alpha_1 \hat{h}_T(1) + \alpha_2 h_T \\
\hat{h}_T(3) &= \alpha_1 h_{T+2} + \alpha_2 h_{T+1} + \alpha_3 h_T \\
&= \alpha_1 \hat{h}_T(2) + \alpha_2 \hat{h}_T(1) + \alpha_3 h_T
\end{aligned}$$

where $\{\alpha_i, i = 1, 2, ..., p\}$ are the prediction filter's weights. Here, we fixed the order of the filter to 3, *i.e.* $p = 3$.

The advantage of prediction by extrapolation method is that it is based on a good model, so the channel can be better characterized.

Figure (5) shows channel coefficients estimated during pilot reception, while the dotted line represents the channel to be estimated during the following data reception period. These estimated coefficients were used to find the filter's weights. Filter's weights jointly with the known channel coefficients are to be used to predict the future values of the time varying channel denoted by the dotted line in the figure. Figure (6) shows the result of coefficients' prediction during data reception. Those coefficients were predicted by the aid of the estimated coefficients shown in figure (5). The prediction error depends on the number of coefficients

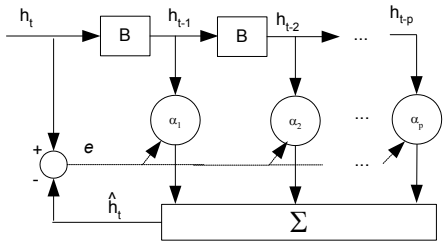

Fig. 4. Prediction filter used during pilot reception in order to estimate channel coefficients for the sequent data reception.

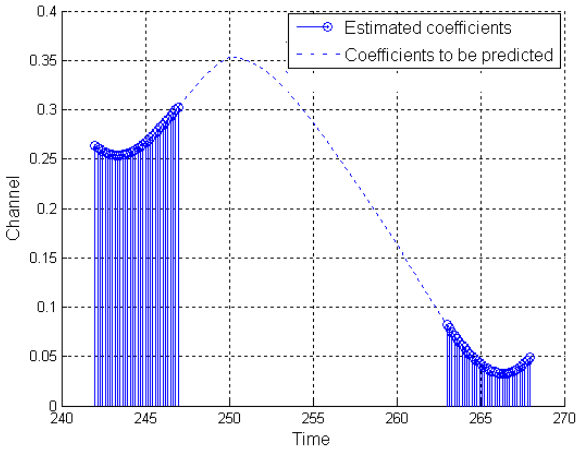

Fig. 5. Channel coefficients estimated during pilot reception. The dotted line represents the channel to be estimated during data reception.

used to find the filter's weights and on the horizon of the prediction. Higher prediction horizon causes higher prediction error.

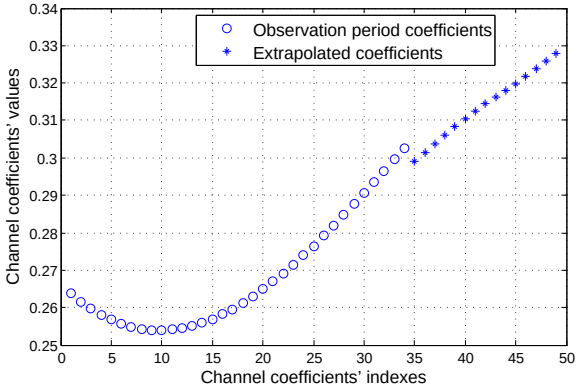

Fig. 6. Channel coefficients estimated during pilot reception. The estimated coefficient are used to predict the future channel coefficients during data reception period.

IV. Simulation and results

To validate our previous analysis, we simulated several scenarios using Matlab simulator and we compared the results obtained when applying our proposition with other results obtained without channel prediction. We considered first the case where collided packets, including pilot ones, are not retransmitted. For the purpose of simplicity, we use here the term packet to indicate a transmission that may actually contain more than one packet. According to the previous analysis, data packets received after pilot signal collisions are not decoded because the channel can't be estimated even though those data packets are collision free. We consider the transmission of L data packets following each pilot signal. If the pilot signal collides, the receiver will not be able to decode some of, or in the worst case all, the L following data packets. Figure (7) shows the propagation channel where collisions occur to data and pilot signals and the value of L in this case is four, $L = 4$. Table I shows

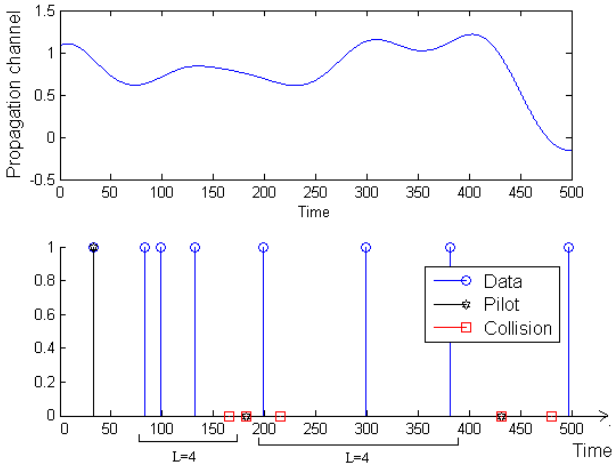

Fig. 7. Pilot and data transmissions with respect to time varying channel. Each pilot signal is followed by four data transmissions, $L = 4$

the result of simulation obtained for two users sharing the same medium with no packet retransmission allowed and without the application of channel estimation and prediction. In order to calculate the throughput, we consider the collision-free data packets whose pilot signal has been also successfully received. Those packets are denoted in the table by successful reception. The throughput of the first user is calculated as follows:

$$\eta_1 = \frac{\text{Total packets} - \text{Total collisions} - \text{Packets without CSI}}{\text{Total packets}} = \frac{52 - 15 - 9}{52} = 53.9\% \quad (11)$$

In the previous equation, we subtracted the packets without Channel State Information (CSI). Those packet were correctly received while their corresponding pilot signal has collided, hence those packets can not be decoded.

By applying our proposition for time varying channel estimation and prediction discussed in section III, data packets received without collision while their corresponding pilot signal has previously collided, can still be decoded because estimated channel coefficients and channel's model permit to predict channel values during data reception period. So, the value of *packets without CSI* was set to zero in equation (12) and hence, the throughput increases. The new throughput, when applying channel prediction, can be calculated as follows:

$$\eta_{1new} = \frac{\text{Total packets} - \text{Total collisions} - \text{Packets without CSI}}{\text{Total packets}} = \frac{52 - 15 - 0}{52} = 71.2\% \quad (12)$$

user, y	Total packets	Collision-free packets	Total collisions	Number of pilots	Pilot collisions	Packets without CSI	Successful reception	Throughput, η_y	New throughput, η_{ynew}
1	52	37	15	11	3	9	28	53.9 %	71.2 %
2	51	36	15	11	5	12	24	47.1 %	70.6 %

TABLE I
Simulation results obtained for DCF applied to a system composed of two users without channel prediction and retransmissions are not allowed

In reality, when collisions happen, packets are retransmitted. Packet's retransmission increases time between pilot signals. However, retransmission is not the only source of inter-pilot signals' time increase. Larger number of users sharing the same medium causes also larger delay between pilot signals, so our proposition becomes a good solution for channel variation problem in the IEEE 802.11 network.

ISBN: 978-1-61804-018-3

When time between pilot signals is much higher than channel coherence time, T_c, channel prediction and estimation becomes more difficult.

Table II shows simulation results obtained for a system composed of five users, $N = 5$. Channel estimation and prediction have been applied to this scenario. We notice that throughput values have been also improved even delays have been increased. This is because pilot signals are retransmitted in case of collision, and this permits the periodic update of channel coefficients that are needed to predict channel values for the following data reception period. In addition, under slow time varying channels, predicted values become valid for longer data reception periods.

From the previous discussion, longer data reception period is equivalent to increasing L, where L is the number of data packets transmitted between two consecutive pilot signals belonging to the same user. We can find that the throughput increases when the parameter L increases. We notice that for $L = 1$, the throughput is almost equal to 50%. The justification is that after each pilot signal, we transmit only one data packet. This means that the useful packets are one out of two, which is equivalent to 50%. As we mentioned before, if the delays caused by the DCF protocol are much higher than the channel coherence time (T_c), throughput will no longer be improved even if L increases. Figure (8) shows the throughput of a system composed of only one user in function of L and for different values of T_c. When T_c is much greater than the time between two pilot signals (ΔT), all data packets will be correctly received while no collisions happen because in this case we have only one user in the system that can access to the medium at any time with no delay. So, the throughput increases when L increases. So, for very large values of L, the throughput becomes closer to 100%. While, the maximum value of L is limited by the channel coherence time, T_c. Figure (8) shows also that when the channel's coherence time decreases, the throughput decreases after certain value of L.

user, y	Total packets	Collision-free packets	Total collisions	Number of pilots	Throughput, η_y
1	55	39	16	7	70.91%
2	57	38	19	7	66.67%
3	38	24	14	5	63.16%
4	67	48	19	9	71.64%
5	16	5	11	1	31.25%

TABLE II

Simulation results obtained for DCF applied to a system composed of five users, where retransmission of the collided packets including pilot ones, is permitted.

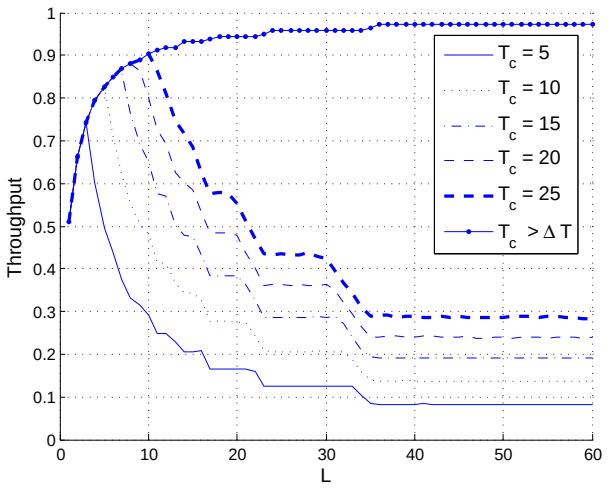

Fig. 8. The throughput as a function of the channel coherence time (T_c). Throughput decreases for smaller values of T_c.

V. Conclusions

In this paper, we proposed the combination of the protocol DCF of IEEE 802.11 with efficient channel estimation and prediction methods. The channel estimator we used is based on the basis expansion model (BEM). The primary goal of this paper is to study the impact of time varying channels on the DCF protocol. The performance of the system was degraded by the increase of the number of data packets when the channel varies quickly or when the number of users increases. We demonstrated that the throughput has been improved when applying BEM's channel estimation method with channel prediction. Simulation results confirm the theoretical analysis exposed throughout this paper.

References

[1] "3GPP Long Term Evolution. System Overview", Product Development, and Test Challenges", Application Note, Agilent Technologies.

[2] Giannakis, G.B.; Tepedelenlioglu, C.; "Basis expansion models and diversity techniques for blind identification and equalization of time-varying channels", Proceedings of the IEEE Volume 86, Issue 10, Oct. 1998.

[3] Ali Nassar, Josep Sala.; "Flat Fading Time-Variant MIMO Channel Estimation Based on Raised Cosine under Orthogonal codes Transmission", Mosharaka Multi-conference on Communications, computers and applications, MM-CCA, September 2007.

[4] Simon S. Haykin; "Introduction to adaptive filters", New York, N. Y. : Macmillan 1984, ISBN 0029494605

[5] T. Zemen, F. Mecklenbräuker, "Time-Variant Channel Estimation Using Dicrete Prolate Spheroidal Sequences", *IEEE Transactions on signal processing*, vol. 53, N°9, September 2005.

[6] D. Slepian,"Prolate spheroidal wave functions, Fourier analysis, and uncertainty-V:The discrete case", Bell Syst. Tech. J.,vol. 57 no. 5 pp. 1371-1430, May-June 1978

[7] Wei Wang, Soung Chang, Victor Li,"Solutions to performance problems in VoIP over a 802.11 wireless LAN", *IEEE Transactions on vehicular technology*, Vol. 54, No. 1, January 2005.

[8] Stefania Sesia, Issam Toufik, Matthew Baker,"LTE the UMTS Long Term Evolution, from theory to practice.". 2009, John Wiley & sons, Ltd. ISBN: 978-0-47069716-0