Modelling Chain for Throughput Estimation in Wireless Networks

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Abstract—The motivation for our work was a feasibility study of the use of wireless technologies, more concretely IEEE 802.11, in industrial control systems. The considered application for configuration files upload to a system component puts requirements on the throughput of the established connection. To estimate the TCP throughput under the large range of varying conditions an analytical model has been developed. The applied approach is a feed-forward modelling chain, where the complex behavior of the communication system is split into several sub-models that are tied together by the overall model design. Defining multiple sub-models enables their individual development primarily with the purpose of complex reduction. The developed model allows to take into account a variety of parameters, such as distance between a transmitter and a receiver, channel conditions, modulation schemes, in-bound interference, cross-traffic from other users etc. Analytical results are compared with the simulation results (using ns-2 simulator) and measurement results (obtained from a specially constructed test-bed).

I. Introduction

Wireless technology is expected to impact to a great extend industrial control and monitoring. Elimination of wires as the physical layer to carry data is often referred to as cable replacement. Cable replacement has clear advantages in installation and maintenance cost reduction, and ease of installation. However, error characteristics of wireless links and communication latency makes this task nontrivial. Although we are witnessing fast deployment of wireless networks, their applicability in industrial environments has been limited so far. Typically, industrial applications have a stringent requirements on e.g. throughput, delay and system reliability. At the same time, for cost-efficient solution it is desired to use commercial off-the-self protocols and components such as 802.11 WLAN or Bluetooth for wireless data transmission and IP/TCP protocol suit.

The goal of this work is to develop an evaluation methodology and testing framework that aims at evaluating different requirements for the communication system with wireless links. Our focus is on throughput estimations. The need for such evaluation has raised during our work for SAFEDMI project [1]. However we believe that the applicability of the developed approach is much broader than a single system and it can be successfully applied in many scenarios where estimation of application layer throughput in wireless systems is required.

The SAFEDMI project is dealing with the development of a safe wireless communication solution for the Automatic Train Control (ATC) system. A wireless connection is used for file upload/download from a train component, namely Driver-to-Machine Interface (DMI), to a Maintenance Center. The system possesses dependability requirements. Dependability of a wireless connection is understood in the way that if the connection has been established, the file transfer have to be completed within a predefined time. In the particular case of an ATC system the time threshold has been set to 5 min. Given the size of the transferred files, the time threshold can be translated into throughput requirements. Thus, a throughput estimator should be developed for a designed communication stack (in this case, for 802.11/IP/TCP stack).

Many scenarios and applications require throughput estimation. For example, in [2] analytical model for TCP optimization using FEC and ARQ is derived. Throughput efficiency modelling and evaluation of cross-layer protection strategies for video transmission over 802.11 WLANS is done in [3]. One can continue this list. In this work we are developing a unifying modelling approach that we refer to as the feed-forward modelling chain. The approach to model the complex behavior of a communication system is to split it into several sub-models that are tied together by the overall model design. Defining multiple sub-models enables their individual development primarily with the purpose of complexity reduction. The already developed models for the individual layers of layered system architectures can be chosen as sub-models or the individual layers can be further subdivided into smaller and less complex systems.

The subsequent sections present a detailed description of our modelling approach, including the sub-models and their interrelations. The developed model is further validated using simulation results and the measurement results obtained from a test-bed.

II. Analytical Model

Figure 1 provides a high level overview of model components and interfaces. The definition of sub-models can partially be based on layers in the protocol stack. This allows a starting point in already studied models such as channel models. The physical environment plays an important role in relation to the performance of the communication, since physical phenomena
deteriorate channel conditions. The purpose of the first sub-model is to estimate Signal-to-Noise Ratio (SNR) due to phenomena existing in wireless channel, such as path loss, shadowing and multipath fading. Depending on the modulation scheme used, the second sub-model maps SNR to bit error rate (BER). BER influences both frame losses (FER) on layer 2 and delays due to retransmissions. Additionally, delay to access the channel depends on the amount of the cross-traffic. The modelling analysis considers 802.11b specifics for these sub-models. FER translates into packet error rate (PER) on IP layer. Delay and PER are used as input for the next model. The output of this sub-model is the TCP throughput.

Fig. 1. Chain of blocks used in the modeling of the link

A. SNR Estimation

For SNR estimation we assume the following input parameters are known: distance \(d\) between the transmitter and the receiver; wavelength \(\lambda\) of the used radio; transmitter and receiver antenna gain (respectively \(G_t\) and \(G_r\)); transmission power \(P_t\); receiver bandwidth \(B\); noise floor of the receiver \(N_f\). The output parameter is the Signal Power per energy bit \(\gamma \equiv E_b/N_0\).

As given by [4], the noise at the receiver \(N\) is the sum of the thermal noise and the noise figure at the receiver:

\[
N = kTB + N_f
\]

where \(k\) is the Boltzmann constant, \(T\) is the effective temperature in Kelvin (commonly 293\(^K\) which corresponds to 20\(^\circ\)C, the ambient temperature) and \(B\) is the receiver bandwidth. The noise figure \(N_f\) of the receiver is defined as the magnitude of the noise added by the receiver itself. Both terms \(B\) and \(N_f\) can be found in the hardware specifications of the receiver. In order to obtain numerical results, it is assumed that \(B = 20MHz\) and \(N_f = 3.7dB\). With these values the total noise is \(N = -96.8dBm\). A background noise level of around \(-97dBm\) has been observed on the test bed, so the calculated figure is assumed to be representative of the real world. The path loss \(L\) is given by

\[
L(d) = -10\log(G_tG_r) + 10\beta\log\left(\frac{4\pi d}{\lambda}\right)
\]

where \(\beta\) is the path loss exponent. The transmission power of a standard 802.11b transceiver is \(P_t = 18dBm\), through a 2dBi dipole antenna. It is assumed that the transmitter and receiver are both of this type. The path loss exponent is assumed to be \(\beta = 3\) based on [5], this seems as a reasonable value for an environment with propagation impairing walls as should be expected on a train. The Carrier to Noise Ratio (CNR) \(C_{dB}\) is then:

\[
C_{dB} = P_t - L - N
\]

The signal power per energy bit can be then found as

\[
\gamma \equiv \frac{E_b}{N_0} = \frac{10C_{dB}/10B}{R_b}
\]

where \(R_b\) is the bit rate.

B. BER Estimation

The BER estimation block assumed that the following input parameters are known: K-factor \(K\) of Rician fading channel; modulation scheme; SNR \(\gamma\). The output parameter is BER \(P_b\). In [6] performance of 802.11b system in terms of BER with Rician/ Rayleigh fading is presented including analysis of the Doppler shift caused by velocity of transmitter and receiver, and the multipath interference due to reflections and diffractions from terrains in the radio service coverage area. The following formulas for average BER are derived for different modulation schemes:

1) DBPSK modulation (employed in 1 Mbps operation mode):

\[
P_b = \frac{1 + K}{2(1 + K + \gamma)} e^{-K\gamma/(1+K+\gamma)}
\]

where \(\gamma\) is an instantaneous SNR. The average SNR \(\bar{\gamma}\) is defined, for the Ricean channel as \(\bar{\gamma} = 2(1+K)\sigma^2\gamma\) with \(2\sigma^2\) being the mean square power of the signal component subject to fast fading.

2) DQPSK modulation (employed in 2 Mbps operation mode):

\[
P_b = \int_0^{\infty} \left[ Q_1(a,b) - \frac{1}{2} I_0(ab)e^{-(a^2+b^2)/2} \right] \cdot \frac{2(1+K)}{\gamma} \left[ -K - \frac{1+K}{\gamma} \right] I_0\left( \sqrt{\frac{4K(1+K)\gamma}{\gamma}} \right) d\gamma
\]

where \(Q_1(a,b)\) is the Marcum Q-function, \(I_0(ab)\) is the modified Bessel function of the first and zero order, and parameters \(a\) and \(b\) are defined as

\[
a = \sqrt{2\gamma(1-\sqrt{1/2})}, \quad b = \sqrt{2\gamma(1+\sqrt{1/2})}
\]

3) CCK modulation (employed in 5.5 and 11 Mbps operation modes):

\[
P_b = \int_0^{\infty} \frac{2(1+K)}{\gamma} e^{-K\gamma/(1+K+\gamma)/\gamma} I_0\left( \sqrt{\frac{4K(1+K)\gamma}{\gamma}} \right)
\]

\[
\left[ 1 - \int_X^{\infty} \left( \frac{1}{\sqrt{2\pi}} \int_y^{\infty} e^{-y^2/2} dy \right)^{M/2-1} e^{-z^2/2} dz \right] d\gamma
\]

where \(X = \sqrt{2E_b/N_0}\) and \(M = 4\) for 5.5 Mbps and \(M = 8\) for 11 Mbps.
C. FER Estimation

For the FER estimation the input parameters are: BER; Payload size, \(s_{data}\); Number of 802.11b neighboring stations, \(n\); Current size of the contention window, \(W\). The output parameter is the FER.

Frame losses can be result of bad channel conditions that are quantified as a BER, or collisions with other 802.11 stations that are trying to transmit in the same channel (collision probability \(P_{cj}^{f}\)).

\[
P_{f} = P_{cj}^{f} + P_{cp}^{p}
\]

We assume that no FEC is used (this is the case for 802.11b standard) and in case of a single bit flip the whole frame is lost. Thus, FER due to the channel conditions can be expressed as:

\[
P_{cj}^{f} = 1 - (1 - P_{b}^{DBPSK})^{hdr}(1 - P_{b}^{mdl})^{pld}
\]

where \(P_{b}^{DBPSK}\) is given by the DBPSK modulation of the header as expressed in eq.(5), \(P_{b}^{mdl}\) given by the corresponding modulation used by the payload and expressed by eq.(8). Equation (10) accounts for the fact that different modulation schemes are used to send the packet header and the packet payload.

In order to quantify the effect of collisions on FER, we need to describe the cross-traffic. We assume the saturated traffic conditions when the transmission queue of each station is assumed to be always nonempty. We assume that there are \(n\) contending stations. The current size of contention window is denoted as \(W\). It is assumed that each of the contending stations has the same current size of the contention window (CW). The key approximation of this modelling approach is that each packet collides with a constant and independent probability \(p\). We do not incorporate in our model the fact that in case of ongoing transmission the back-off counter will be frozen and restarted in the next contention opportunity. Instead, we assume that the collision probabilities are the same during any contention period. Under these assumptions, \(P_{cp}^{p}\) can be found to be

\[
P_{cp}^{p} = 1 - (1 - p)^{n} - np(1 - p)^{n-1}
\]

where \(p = 1/W\), since the transmission slot is chosen uniformly and independently by each station among \(W\) slots.

Note that the transmission of the L2 Acknowledgement packet is always collision free. ACK packet can be lost due to the bad channel conditions, however due to the small size of the ACK packet the probability of its loss is much smaller compared with the probability to loose a data packet and therefore, it is omitted in our calculations. We assume that ACK is always received correctly.

D. L2 delay and retransmissions

For L2 delay estimation input parameters are: FER; Current size of the contention window. The output parameters are: Average number of retransmissions; L2 delay.

We denote as \(\Delta_{L2}\) a random delay for a single packet transmission (or the packet retransmission). This delay consists of the time a station spends contending for the channel (including the time needed to decrease back-off counter to zero and transmission time of other stations that wins the contention before the counter reaches zero) and the packet transmission time. Let \(w\) be a randomly chosen initial value of the back-off timer from interval \([0, W]\). Then delay can be found as

\[
\Delta_{L2} = \sum_{j=1}^{W} \Delta_{j} + w\Delta_{slot} + \Delta_{dp}
\]

where \(\Delta_{j}\) is a random variable indicating frozen time of the back-off timer between \(j\)-th two consecutive back-off timer decrementing instants; \(\Delta_{slot}\) is the slot duration and \(\Delta_{dp}\) is the time needed to transmit a data packet and an acknowledgement. A r.v. \(\Delta_{j}\) is a discrete random variable that can take on only two values: it is either 0 in case when the current slot is idle or it is equal to \(\Delta_{dp}\) if the slot is detected as busy and another station has started a transmission. Its probability mass function is given as \((1 - 1/W)^{n}\) for \(\Delta_{j} = 0\) or \(1 - (1 - 1/W)^{n}\) for \(\Delta_{j} = \Delta_{dp}\).

We are interested in estimating an average delay. Taking expectations from both left and right parts of the previous expression, we obtain

\[
E[\Delta_{L2}] = E[w]E[\Delta_{j}] + E[w]\Delta_{slot} + \Delta_{dp}
\]

where the expected values for \(w\) and \(\Delta_{j}\) are

\[
E[w] = \frac{W}{2}; \quad E[\Delta_{j}] = T[1 - (1 - 1/W)^{n}]
\]

In 802.11 standard the number of L2 packet retransmission is limited. We denote the maximum number of retransmissions as \(m\). Typically, this number depends on a particular implementation and lies in range \(m = 3...7\).

Considering retransmissions, the total delay experienced by a packet is

\[
\Delta_{t} = \Delta_{L2}N
\]

where \(N\) is a number of retransmissions.

Expectation of \(N\) can be found as

\[
E[N] = \frac{1 - P_{f}^{pp}}{1 - P_{f}^{p}}
\]

And the expected value of the total delay is \(E[\Delta_{t}] = E[\Delta_{L2}] \cdot E[N]\).

One should note that there exist other modelling approaches for WLAN 802.11 delay and throughput estimations where some of the stated above assumptions are relaxed. For example, in [7] the analytical evaluation of the saturated throughput is derived modelling behaviour of a single station as a Markov chain. However, it requires a numerical solution of a nonlinear system of equations.
E. PER estimation

The input parameters are: FER; Maximum number of retransmissions. The output parameter is the PER. Due to the limited number of retransmission attempts, a packet will be dropped if the number of retransmissions exceeds a predefined threshold. Therefore, PER is related to FER in the following way:

\[ P_p = P_f^m \] (17)

F. RTT estimation

For RTT estimation the input parameters are: L2 delay; Delay for wired BD-DMI connection; Buffer size. The output parameter is the RTT. The time to transmit one data packet and the TCP acknowledgement includes the time for two transmissions (MC-BD and BD-DMI) and a reply, also consisting of two transmissions. The transmission delay over a wired link BD-DMI can be assumed to be constant. The processing time is negligible small compared with the time it takes to transmit a packet and therefore, it is omitted in calculations. However, the queuing time can not be ignored and as we will show now it is a significant part of a RTT. The queue size is limited, we denote it as \( M \). In our measurements and simulations \( K = 46 \) packets. The departure process is essentially determined by number of retransmissions needed to send a L2 packet, and thus, it can be approximately considered to be exponential. The arrival process has the same nature as a departure process: each time a TCP ACK packet is received, a new packet is put in the queue. Thus, the system behaviour can be described as M/M/1/K queue. Since it takes approximately the same time to send a packet from MC to DMI and from DMI to MC, we assume that the arrival and departure processes have the same rate. In this case the mean number of packets in the queue is given by

\[ E[Q] = K/2 \] (18)

It means that before a packet can be transmitted, it has to wait on average until \( K/2 \) packets that are in the queue are transmitted. Thus the RTT \( \Delta \) is defined by:

\[ \Delta = \Delta t \cdot K \] (19)

G. TCP throughput estimation

TCP throughput estimation assumes that the following input parameters are known: PER; RTT; Maximum size of TCP congestion control window; Average time-out duration; TCP window increment. The output parameter is the TCP throughput.

In [8] a simple model of the steady state throughput as a function of a loss rate and round trip time for a bulk transfer TCP flow is derived. It takes into account TCP’s fast retransmission mechanism and the affect of the timeout on the TCP throughput. The following formula gives approximation for the throughput measured in the number of packets received during one time-unit:

\[ R \approx \min \left( \frac{W_{tcp}}{\Delta}, \frac{1}{\Delta \sqrt{\frac{2P_p}{3}} + \Delta_0 \min \left( 1, \frac{3}{\sqrt{3}bP_p^2} \right)} \right) \] (20)

where

\[ F = P_p(1 + 32P_p^2) \] (21)

Where \( W_{tcp} \) is the congestion control window size, \( \Delta_0 \) is average single time-out duration, and \( b \) is increment of a window. \( W_{tcp} \) is a fixed parameter depending on the implementation. TCP uses RTT estimations and based on these estimations its timeout is modified accordingly. The low-pass filter is used to obtain a smoothed RTT estimator. Given this smoothed estimator, RFC 793 recommended the retransmission timeout value be set to \( \Delta_0 = \alpha \Delta \), where \( \alpha \) is a delay variance factor with a recommended value of 2. This is an approach we adopt in our calculations. Due to the fact that we are dealing with average values, we do not incorporate a low-pass filter in our model.

III. RESULTS

Figures 2-4 illustrates the presented above analytical approach for throughput estimation. Figure 2 gives Eb/No as a function of a distance between MC and BD. Figure 3 maps Eb/No into BER. Finally, throughput as a function of a BER is presented in Figure 4. In all figures three different curves corresponds to different data rates: 1Mbps, 5.5 Mbps and 11 Mbps. The figures corresponds to the case when there is no collisions with the other 802.11 stations, i.e. \( n = 0 \). Additionally, the suburban outdoor environment or indoor environment with some LOS power is assumed, modelled by Rician fading channel with K-factor of 10.

In order to validate the developed model the obtained analytical results are compared with the simulation and measurement results. For simulations ns-2 version 2.29 was used with a patch that adds the Rician model to any path-loss model available. The detailed description of the chosen parameters for the channel, link layer and TCP parameters can be found in [1].
The specially constructed test-bed [1] aimed primarily at determining achievable performance of 802.11-based wireless link in the SAFEDMI scenario. In this work the subset of the obtained measurements is used as a basis for analytical results verification. The wireless interfaces on two computers are used in two configurations, monitor mode and ad-hoc mode. While the latter is used in the normal experimental setup, monitor mode enables collection of all activities in a given channel. This is used to establish the amount of competing traffic in the chosen channels. For experiments only 802.11b is used. For all experiments channel 9 is used (channel centre frequency: 2.452 GHz). Transmission power of both computers is 18 dBm. Power management and automatic transmission rate adaptation is disabled, as well as RTS/CTS is disabled.

Measurement, simulation and analytical results corresponding to the case of good channel conditions (wireless transmitter and receiver are located close; signal strength about -50dBm) are presented in Table 1. Looking at goodput and Round Trip Time (RTT), there is a good resemblance between the results. As we observed in our studies, the most significant influence to RTT is caused by the waiting time in the queue. The good resemblance in the results indicates that queue size in simulation and analytical model is reasonably defined with respect of the real setup. Another reason is that TCP maximum window size in the simulation and the analytical model, as presumably in the real setup, is limited to 64Kb of unacknowledged segments. This clearly also limits how many packets/frames are put in the queue effectively.

IV. CONCLUSIONS

In many industrial applications the requirements on wireless communication solutions can be formulated in the form of requirements on the minimum throughput of a wireless link. This work has been devoted to analytical modelling of throughput seen at the application layer when the underlying layers of the protocol stack are LL as specified by 802.11 standard and IP/TCP. The presented approach has been originally derived for the case of Automatic Train Control system, but it has a larger applicability to any wireless communication system with a similar protocol stack. The derived model is further validated through simulation and measurement results.

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