Optimization of WiMax modulation scheme with a cross layer erasure code

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Abstract

WIMAX (Worldwide Interoperability for Microwave Access) is a promising new networking technology that potentially offers high speed and wide area wireless access services that complement consistently the 3G and WiFi access networks capabilities. The standard proposes an adaptive modulation scheme which allows WiMax nodes to communicate from various modulation coding schemes according to the link quality. However, the standard does not define a detailed link adaptation algorithm and currently, the most largely used modulation adaptation technique is based on a channel quality lookup table. We argue that this method is not able to make the best adaptation decisions and delivers a sub-optimal goodput in numerous communication contexts. In this paper, we propose a novel cross layer based modulation adaptation mechanism which incorporates the use of adaptive erasure code with the physical layer information to significantly improve the goodput and transmission efficiency. Simulation results show that our proposal adapts more efficiently to real environments and achieves a significant gain on the goodput delivered to mobile nodes.

1 Introduction

The increasing use of portable devices in daily life and the dramatic and continuous growth of the “mobile and wireless Internet” greatly increase the needs of offering higher capacity, higher reliability, and more advanced multimedia services to wireless mobile users. The IEEE 802.16 offers several features that can potentially respond to this acute needs in global wireless access for mobile end systems. The IEEE 802.16 standard defines within its scope four physical (PHY) layers according to the various modes of system operation (i.e. non-line of sight (NLOS) and line of sight (LOS) environments). Specially, the IEEE 802.16e[1] is designed to incorporate the current competitive technologies in communications and digital signal processing to achieve a broadband Internet experience for nomadic or mobile users over a wide or metropolitan area. OFDMA[2] is one of the key techniques which partition the available bandwidth into different sub-carriers to improve the robustness to multipath propagation and bandwidth efficiency. Adaptive Antenna Systems (AAS)[3] and Multiple Input Multiple Output (MIMO)[4] technology are introduced to improve the capacity and coverage together with adaptive modulation and coding (AMC). Forward Error Correction (FEC) [5] and Automatic Repeat request (ARQ)[6] are also considered to ameliorate the transmission reliability. Currently, the extended version of WiMax, the IEEE standardization of 802.16m, which is also called WiMax II is in progress. This emerging standard will incorporate the most promising communication techniques to offer 100 Mbit/s data rates to mobile systems and 1 Gbit/s to fixed end systems. Meanwhile, the standard leaves a lot of open issues which are not detailed, such as bandwidth assignment, detailed link adaptation mechanism, etc. It gives the manufacturers some flexibility to implement more or less efficient mechanisms that could be determinant for the credibility and attractively of WiMax compared to the current and future communication systems. Link adaptation plays an essential role for the efficient use of the radio resources in WiMax networks. IEEE 802.16 defines a radio link control (RLC) framework to enable the implementation of PHY layer adaptation schemes. This framework includes the definitions of the signal indicator message and signal flow for link adaptation. Currently, the largely accepted adaptive modulation mechanism is based on a channel quality lookup table, from which the modulation and coding scheme is chosen according to the monitored channel quality such as SNR, SINR or CINR value. However, this method is not always able to adapt efficiently to different terrain types (SUI_i models)[7] or when different FEC coding schemes are integrated[8]. Therefore, while being simple, such a lookup table based approach entails sub-optimal modulation/coding choices that lower the useful throughput delivered to end systems and impact negatively

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on the transmission efficiency of WiMax networks. In response to this issue, [9] proposes a dynamic threshold link adaptation algorithm for different channel conditions. [8] proposes a cross-layer link adaptation algorithm based on the QoS demand and channel condition. [10, 11] introduce a mean Carrier-to-Interference-Noise Ratio (CINR) based link rate adaptation scheme, the modulation mode will be upgraded or downgraded when the updated mean CINR is out of the required range of the current PHY mode. However, all of these works either significantly modify the standard or are not able to efficiently adapt to real varying channel condition. In this paper, we propose a cross layer–based modulation adaptation mechanism which incorporates high layer (i.e. IP layer) adaptive erasure code with the PHY layer information to significantly improve the goodput and transmission efficiency.

The rest of this paper is organized as follows: in section 2, we introduce our proposal and define an analytical description and evaluation of the proposed mechanism. Section 3 will focus on the validation work through several simulation scenarios. Conclusion will be finally given in section 4.

2 Cross layer FEC based modulation adaptation

There are two main approaches in the current commercial market of WiMax deployment: 1) QoS supported deployment according to the standard. 2) Pre-wimax deployment which assigns the mobile nodes certain time slots and frequencies according to their service level agreement. At lower layers, both of these solutions implement an adaptive modulation scheme which is classically based on channel quality lookup tables. In approach 1, five QoS classes have been defined in the standard; the throughput of Unsolicited Grant Service(UGS) connection is guaranteed and does not depend on the link quality, which means the BS has to assign more time slots or subcarriers to the mobile nodes when SNR degrades and lower modulation/coding scheme is used. This approach can decrease the network capacity offered to other nodes attached to the considered base station. This behavior entails a syndrome similar to the well known performance anomaly issue observed in the context of 802.11 [12]. In approach 2, the base station assigns up to a maximum number of time slots or frequencies to the mobile nodes for data transmission. Since mobile WiMax is a TDD based technology; there is a strong interest in maximizing the goodput for given assigned time slots and frequencies. According to the WiMax standard, transmission reliability can be optionally insured from an ARQ mechanism applied at the MAC layer; however, such mechanism potentially impacts negatively on the end to end delay and may reduce the transmission throughput [13]. In order to obtain a good trade-off between reliability, delay and throughput, we propose to set the ARQ retry timeout to zero (timer not scheduled) to avoid retransmission [16] and still get feedback about the channel state from ACK/NACK messages. Based on a cross layer approach, this low layer feedback information feeds a high layer adaptive FEC mechanism that aims to improve channel reliability while delivering high goodput and low end to end delay. This high layer FEC mechanism is based on Maximum Distance Separable Convolutional Codes [19] which is compatible with the implementation in [20]. Such an approach can be applied both to Up-Link (UL) and Down-Link (DL) flows. In this paper, the analysis focuses on the UL flows.

2.1 Conceptual view of our proposal

Our proposal focuses on a novel modulation decision mechanism which maximizes the goodput of mobile nodes and improves the transmission efficiency. In other words, the approach promoted in this paper aims to replace the default staircase behavior, resulting from the use of a SNR lookup table, by a more “continuous” one that delivers a higher goodput and lowers end to end delay. The improvement is achieved by applying a high level (i.e. network layer) adaptive FEC mechanism which allows correcting packet errors according to the monitoring of packet error rate and the instantaneous SNR at the BS side (for UL flows) and MN side (for DL flows). Our approach is based on a self-learning decision engine that defines a dynamic mapping function which allows the mobile nodes, by considering the current SNR and packet error rate, to select the modulation/coding scheme that delivers the best goodput. Following this decision mechanism described in the next section, the optimal modulation/coding scheme corresponding to the best goodput is then dynamically selected. Compared to the default modulation adaptation mechanism, our proposal takes into account not only the signal to noise ratio, but also the instantaneous packet error rate corresponding to the dynamically varying communication conditions. We will show in the following sections that our high layer adaptive FEC proposal makes it possible to delay the recourse to a lower modulation when signal degrades and delivers a better goodput. Moreover such an approach avoids systematically resorting to a brutal discrete rate reduction entailed by a sudden modulation change.

2.2 Mapping function

In our proposal, the ARQ mechanism is enabled in order to receive feedback of the channel state from ACK/NACK messages. According to the standard, Service Data Units (SDU) are partitioned into fixed-length ARQ blocks (ARQ-BLOCK-SIZE, of which the size is limited to 2040 bytes for fixed WiMax [16] and 1024 bytes for mobile WiMax [1]). At the sender side, the ARQ Block Error Rate (A-BLER) can be directly derived from the returned ACK/NACK information (represented by selective ACK bitmaps) for a continuously monitored sliding window of N transmitted ARQ blocks. Moreover, in order to have an easier and more efficient error control mechanism, we consider that the IP layer fragments its Network-PDU to a maximum size of ARQ-BLOCK-SIZE (1024 bytes in our proposal). That is, each MAC-SDU can be associated by the MAC layer to an ARQ block which is in turn encapsulated into a MAC-PDU, an approach also suggested in [15]. Therefore, as a result, the IP layer packet error rate (PER) is equivalent to the ARQ Block Error Rate and can be calculated by a simple analysis of the returned ACK/NACK information.

In order to define an efficient FEC adaptation policy that aims to dynamically adjust to channel losses, we dynamically update a mapping function defined for the current communication context of the mobile node. This continuously evolving mapping function dynamically maintains, according to the mobile nodes communication experience, the packet error rates (p^i_{SNR}) when a given modulation/coding scheme i is used and a SNR value (defined with a precision step of 0.1dB in our proposal) is observed. For different SUI channels, this mapping database is initially populated with the packet error rate values resulting either from standard IEEE channel models or from simulations. This initial mapping database is dynamically enriched by the hereafter introduced self-learning mechanism that takes into consideration the dynamic channel conditions experienced by the mobile node, so introducing adaptability in the channel modeling structure.

This mapping function maintains an adaptive knowledge database that results from the communication experience of the mobile node. :

$$p^i_{SNR} = f(modulatio\_coding\_i, SNR) \qquad (1)$$

From a practical point of view, we assume that the mobile nodes monitor the packet error rate and are aware of the current Signal to Noise Ratio SNR_{cur} (at the BS side for UL approach) which are either periodically sent by BS or derived from the SNR monitored by mobile node and the transmission power parameters[17]. For the DL flows, SNR_{cur} is the monitored Signal to Noise Ratio that can be obtained from the Channel Quality Indicator field. To avoid oscillations, the SNR variations can be smoothed periodically with an exponential moving average as:

$$SNR = (1-k) * SNR + k * SNR_{cur} \; where \; k \in [0;1] \qquad (2)$$

The resulting smoothed SNR is approximated with a precision of 0.1dB. The instantaneous packet error rate, p_{cur} can be periodically processed by analyzing the ACK/NACK information (represented by selective ACK bitmaps) for a sliding window of N instantaneous transmitted ARQ blocks. Similar to the SNR process, the packet error rate p^i_{SNR} corresponding to the estimated SNR and current modulation is also smoothed and updated periodically by an exponential moving average :

$$p^i_{SNR} = a * p_{cur} + (1-a) * p^i_{SNR} \; where \; a \in [0;1] \qquad (3)$$

This periodic processing of the SNR and the related packet error rates make it possible to establish a dynamic database that contains the packet error rate p^i_{SNR} according to a given modulation/coding scheme i and a given SNR interval. This mapping database allows the mobile node to estimate the goodput it would obtain if it uses a higher or lower modulation/coding scheme (i.e. when using neighbor modulations).

2.3 Adaptive Modulation and coding scheme decision algorithm

In reaction to an observed packet error rate, p_{cur}, in order to insure packet errors correction, we propose to apply instantaneously an adaptive FEC mechanism based on MDS codes [19], of which the FEC ratio is a function of the current packet error rate, $f_{cur} = g(p_{cur})$ to Network-SDUs. If we consider that a group of Network-SDU includes w Network-SDUs, then the number of redundancy packets (with the same packet size of 1024 bytes) for each group is given by:

$$m = ceiling(w * \frac{f_{cur}}{1-f_{cur}}) \qquad (4)$$

The FEC ratio $f_{cur} = g(p_{cur}) = p_{cur}$ is in a first step based on the hypothesis of an uniform packet error distribution that makes possible the full recovery of the lost packets. Note that this hypothesis is reasonable when the FEC ratio processing is applied to short periods that enable an adaptation to non uniform packet error distributions.

In the context of UDP based transmission, within time slots t and subcarriers s assigned by BS, we denote respectively R_c : the current UL transmission rate of a mobile node, R_l: the UL transmission rate if the mobile node uses the next lower modulation/coding scheme, R_h : the UL transmission rate if the mobile node uses the next higher modulation/coding scheme. Then, the UL goodput within the assigned time slots (useful transmission rate successfully received by the BS during t) is given by:

$$G_c = R_c * (1 - f_{cur}) = R_c * (1 - p_{cur}) \qquad (5)$$

	QPSK3/4	16QAM1/2	16QAM3/4	64QAM2/3	64QAM3/4
c_i	1.5	2	3	4	4.5
c_i/c_{i-1}	1.5	4/3	3/2	4/3	9/8

Table 1. Different modulation/coding scheme ratios

The object of our contribution is to choose an optimal combination of the high layer FEC and the default adaptive modulation and coding AMC scheme. This combination results from seeking to use the higher modulation rate while optimizing the goodput (i.e. we look for the FEC ratio needed to correct the errors on the channel link when we use a higher order modulation and coding scheme, under the conditions specified for a lower order one used in default case.) This approach aims to deliver both an optimal individual and collective goodput to the whole set of mobile nodes connected to the considered base station. In order to take an efficient modulation-choosing decision, we compare this resulting rate, G_c, to the goodput obtained if the mobile node uses the next lower or higher modulations/coding schemes. Indeed, we consider the case where the MN uses respectively the next higher and lower modulations/coding schemes with the current smoothed SNR. From the dynamically elaborated database, we can get the corresponding packet error rates $p^i_{SNR}\,(with\,i = l\,or\,i = h)$, the percentage of losses with the current smoothed SNR and the next lower $(i = l)$or higher $(i = h)$ modulations/coding schemes : $p^i_{SNR} = f(i, SNR)$.

Similar to the previous analysis, if we set the FEC ratio $f_i = p^i_{SNR}$, then the UL goodput becomes G_i if the mobile node uses the neighbor modulations/coding schemes:

$$G_i = R_i * (1 - f_i) = R_i * (1 - p^i_{SNR})\,with\,i = h\,or\,i = l \tag{6}$$

The MN will compare G_c to G_l and G_h periodically, the optimal modulation and coding scheme that delivers the higher goodput is chosen and informed to the BS, the UL-MAP [1] is then generated by the BS according to the updated modulation and burst profile. We don't need to compute the exact value of R_c, R_l and R_h, that depend not only of the modulation code but also of several other SLA based parameters, we just need to calculate the ratio between them for the comparison purpose.

The spectral efficiency c is defined as:

$$c = coderate * log_2(M) \tag{7}$$

where $coderate$ and M represent respectively the coding rate and the M-ary phase.

[1]According to the WiMax standard, UL-MAP messages include the burst profile for each uplink user, which specifies the modulation and coding scheme used in the communication channel.

We define the different spectral efficiency c_i as: $c1$ (QPSK1/2), $c2$ (QPSK3/4), $c3$ (16QAM1/2), $c4$ (16QAM3/4) , $c5$ (64QAM2/3), $c6$ (64QAM3/4). Indeed, with the given time slots t and subcarriers s, the transmission rate depends on the spectral efficiency:

$$\frac{R_m}{R_n} = \frac{c_m}{c_n} \tag{8}$$

Table 1 lists the so obtained different modulations ratios.

The algorithm of our proposal is described below:

When (timer runs out) do
{
Fetch the current SNR, calculate p_{cur} and update the database p^i_{SNR};
Process FEC ratio f_i according to the current SNR and the next higher and lower modulations;
Process the comparison of Gc,Gl and Gh;
if (Gc>=Gl) and (Gc>=Gh)
{
Inform BS that MN should keep its current modulation and coding scheme ;
FEC ratio= $f_{cur} = p_{cur}$;
}
else
{
if (Gc<Gl)
{
FEC ratio = $f_l = p^l_{SNR}$;
Inform BS that MN should degrade its coding scheme or modulation rate, $c_i = c_{i-1}$. (UL-MAP is then scheduled by BS);}
else
if (Gc<Gh)
{
FEC ratio = $f_h = p^h_{SNR}$;
Inform BS that MN should upgrade its coding scheme or modulation rate, $c_i = c_{i+1}$. (UL-MAP is then scheduled by BS);}
}
}

Our proposal allows determining an optimal modulation and coding scheme to obtain higher goodput for the mobile nodes with given time slots and subcarriers. The promoted approach can efficiently improve both the individual and collective transmission efficiency in the whole coverage of the considered WiMax base station. The following section will give an analysis on the global gain on the goodput delivered by our proposal, specially when UGS connections are managed by the base station.

2.4 Analysis of Performance Improvements

We will show in this section that the previously introduced FEC based adaptive modulation scheme offers a significant goodput gain to the whole set of connections managed by the base station. Take UGS connection for example, whose bandwidth is guaranteed by the BS. When the mobile node's signal condition worsen, the default modulation mechanism leads him to downgrade his modulation/coding scheme c_i. As a result, the BS has to allocate more time slots or subcarriers to guarantee the UL/DL bandwidth. Our proposal aims to delay this modulation/coding downgrade while guaranteeing the negotiated QoS and the transmission reliability (in other words, such an approach allows improving the default goodput). Indeed, this approach not only preserves the QoS of the considered mobile node but also "saves" time slots used for the other mobile nodes in the wimax BS coverage, and as a result improves the global goodput of the network.

In order to make an analysis of the resulting goodput gain, let us consider N UGS connections of which the respective required UL bandwidths are B_k bps (k=1,..N). When considering a given SNR , their spectral efficiency (bits/symbol) are respectively c^k_{def}(k=1...N) according to the default modulation lookup table[1]. We assume that all the N connections use default modulations that can be improved by our FEC based mechanism, so with the Modulation Decision Algorithm presented in section 2.3, a more efficient modulation/coding scheme $c^k_{new}$is then chosen for the N connections with a FEC ratio f_k. Furthermore, we also assume all the other UL connections (except for the UGS connections) occupy a number of data symbols s_i ($i \in [1, 6]$) which respectively corresponds to the different spectral efficiency $c_i (i \in [1, 6])$.

The total number of symbols per second (N_{sym1}) used for N upload UGS connections considered in the default scheme is then given by:

$$N_{sym1} = \sum\nolimits_{k=1}^{N} \frac{B_k}{c^k_{def}} \tag{9}$$

Since our proposal allows applying redundancy packets (with a FEC ratio of f_k) to the UGS connections, in order to guarantee the bandwidth B_k which is required by MN's applications, the bandwidth assigned (by BS) to the UGS connections becomes $\frac{B_k}{1-f_k}$ (which includes the required bandwidth B_k and the bandwidth for the redundancy packet transmission:$\frac{B_k * f_k}{1-f_k}$). However, although the assigned bandwidth increases, the assigned time slots (or subcarriers) for the N UGS connections decrease thanks to the use of a more efficient modulation/coding scheme c^k_{new}, (where $c^k_{new} > c^k_{def}$).Then, the total number of symbols per second (N_{sym2}) used for the N UGS connections with the proposed FEC based scheme is given by:

$$N_{sym2} = \sum\nolimits_{k=1}^{N} \frac{\frac{B_k}{1-f_k}}{c^k_{new}} \tag{10}$$

Therefore, our approach allows $(N_{sym1} - N_{sym2})$ symbols to be "saved" and to be available for the data transmission of the other mobile nodes. The property $(N_{sym1} - N_{sym2}) > 0$ is guaranteed by the Modulation Decision Algorithm presented in section 2.3 (with $f_{cur} = 0$, since no FEC redundancy packets are applied in default case). If we suppose that these "saved" symbols are used by other nodes with an equivalent spectral efficiency c_{other} bits/symbol, then this leads to a bandwidth gain B_{gain}:

$$B_{gain} = c_{other} * (N_{sym1} - N_{sym2}) \tag{11}$$

In the default case, we assume that the efficiency of the non-UGS connections is modeled by the following mean efficiency :

$$c_{non\_ugs} = \frac{\sum_{i=1}^{6} si * ci}{\sum_{i=1}^{6} si} \tag{12}$$

We denote S, the number of available symbols in each frame for UL data transmission:

$$S = Sym_{UL} * N_{subcarries} - S' \tag{13}$$

Then, the global UL bandwidth in default case is:

$$B_{UL} = \sum\nolimits_{k=1}^{N} B_k + (\frac{S}{T_f} - N_{sym1}) * c_{non\_ugs} \tag{14}$$

Where Sym_{UL} represents the total number of symbol time in a frame used for UL, $N_{subcarries}$ represents the total number of subcarriers used for data transmission. S' represents the symbols that are not used for data transmission (i.e. contention area in UL part). T_f represents the frame duration.

If we suppose that the lower modulation in default case is efficient enough to guarantee a low error rate, so we can define the global gain ratio G as:

$$G = 1 + \frac{B_{gain}}{B_{UL}} \tag{15}$$

2.4.1 Case study

In this section, for a given communication condition, we will estimate the global goodput gain that our approach can deliver to a mixed set of UGS and non UGS connections. Considering a SNR of 14.5dB (SUI_1 environment, Category C: Flat/light tree density), according to the default modulation lookup table, the N uploading UGS connections use the modulation of 16QAM1/2 ($c_{def} = 1/2 * log_2(16) = 2bits/symbol$). Our proposal allows the mobile nodes to use a higher modulation/coding scheme 16QAM3/4 ($c_{new} = 3/4 * log_2(16) = 3bits/symbol$) with a FEC ratio of $f = 25\%$ according to the dynamic mapping database introduced in section 2.2. If we assume that the "saved" symbols are used by other nodes with an equivalent spectral efficiency c_{non_ugs}(the equivalent spectral efficiency for the non-UGS connections in default case), and that B_{UGS} represents the total bandwidth required by the N UGS uploading connections

$$B_{UGS} = \sum_{k=1}^{N} B_k \qquad (16)$$

Then, the global gain G results from equations (11), (13), (14) and (15):

$$G = 1 + \frac{c_{non\_ugs} * (\frac{B_{UGS}}{c_{def}} - \frac{\frac{B_{UGS}}{1-f}}{c_{new}})}{B_{UGS} + (\frac{S}{T_f} - \frac{B_{UGS}}{c_{def}}) * c_{non\_ugs}} \qquad (17)$$

According to [1], We assume that $Sym_{UL} = 12$ for the UL part in each frame, $N_{subcarries}$=1120 for UL data transmission, S' = 288, the frame duration T_f = 5ms.

Fig.1 represents the global goodput gain in function of the percentage of upload UGS bandwidth (B_{UGS}/B_{UL}) and the spectral efficiency c_{non_ugs}.

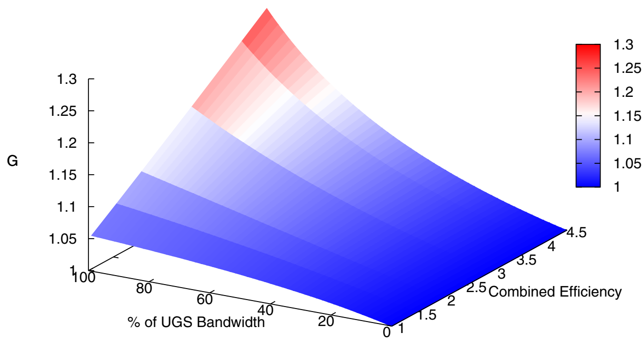

Figure 1. Global Gain

This section has introduced the principal of our proposal which, compared to the default behavior, allows mobile nodes to benefit from higher modulation rates while guaranteeing their transmission reliability. We have shown that this approach delivers a higher goodput compared to the default modulation mechanism which is based on discrete and brutal modulation and rate variations.

Frequency	3.4GHz	Bandwidth	20MHz
Frame Duration	5ms	Symbol duration	102.86us
UL subframe Size	5 Symbols	Frame Preambles	1 Symbol
TTG	106us	RTG	60us
UL Data subcarriers	1120	UL subchannels	70

Table 2. Simulation parameters setting

3 Simulation results

This section gives the simulation results of our proposal as get from OPNET simulations [18]. We assume that a mobile node (UDP connections based) is in the SUI_1 environment (Category C: Flat/light tree density). The Wireless OFDMA 20 MHz bandwidth is used in the considered scenario; the UL Sub-frame size (Sym_{UL}) is set to 5. All the available uplink sub-frames and data subcarriers of the BS are assigned to the mobile node for UL data transmission (i.e. the UL goodput delivered to the mobile node represents the global goodput). Table 2 describes the simulation parameters setting.

3.1 Scenario I: Uniform Movement

We assume that the considered mobile node is moving away from the BS. As a result of this mobility scenario, the SNR monitored at the receiver side (i.e. BS) ranges from 22dB to 4dB (representing all the different modulation/coding schemes $c_i(i \in [1, 6])$.

Fig.2 represents the default and the FEC based goodput experienced by the mobile node. Fig.3 represents the gain of UL goodput with our proposal compared to the default method. Fig.4 represents the FEC ratio at higher layer according to the SNR value. Fig.5 represents the default and FEC based modulation applied in terms of SNR(6: 64QAM3/4; 5: 64QAM2/3; 4: 16QAM3/4; 3:16QAM1/2; 2:QPSK3/4; 1:QPSK1/2).

We observe that the goodput gain is not homogeneously spatially distributed along the radius of the WiMax coverage. In the following, we will estimate the percentage of the WiMax coverage, approximated by a disk, where a gain can be observed. In order to estimate this ratio, we will use the model introduced in [14] for the processing of the distance between the BS and a MN in open air (D) according to the monitored SNR:

$$E = \frac{PE[dBm] + 10log(GE)[dB] + 10log((GR)[dB]) - SNR[dB] - N[dBm]}{20} \qquad (18)$$

$$D = \frac{\lambda * 10exp(E)}{4 * \pi} \qquad (19)$$

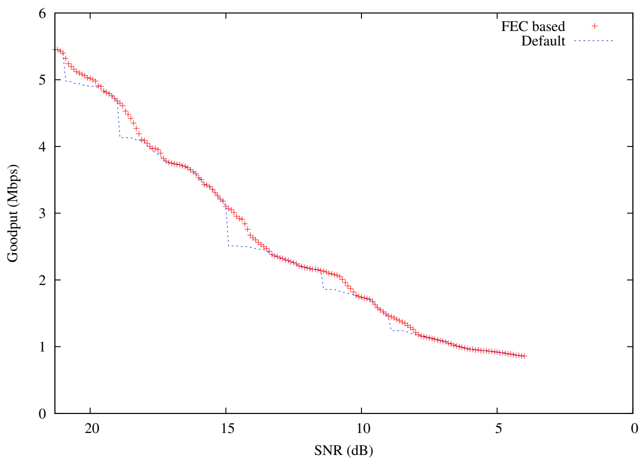

Figure 2. Default and FEC based UL goodput

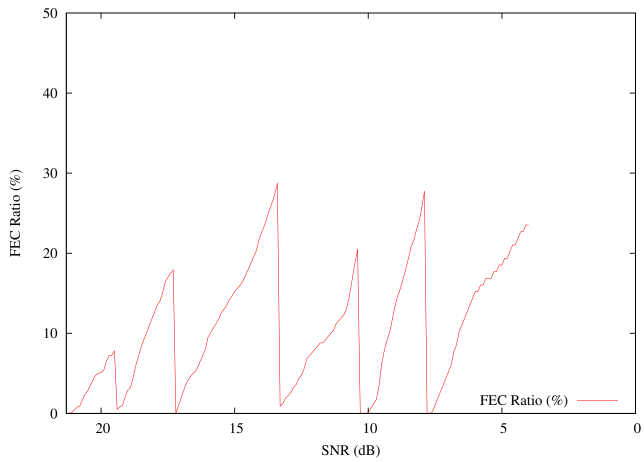

Figure 4. FEC ratio according to SNR

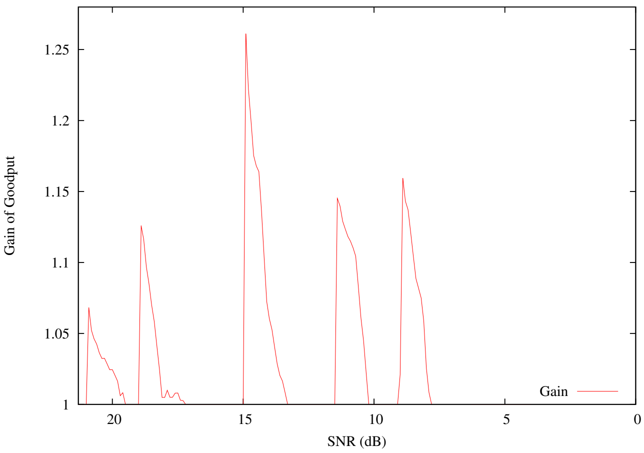

Figure 3. Gain of UL goodput

Where PE is the emitted power, GE is the emitter antenna gain, GR is the receiver antenna gain, N is the thermal noise and λ is the wavelength. If we assume a frequency of 3.4GHz used for the outdoor WiMax in France and the thermal noise is equal to -100.97dBm. According to a maximum allowed Effective Isotropic Radiated Power (EIRP) of 1W, the emitters are assumed to have an emission power PE of 1W. From Fig.3 and equations (18, 19), if we assume that the available SNR for MN is between 4dB and 55dB, we can get goodput gain in the areas where the distance between BS and MN are respectively (2030-2600m), (2980-4030m), (6360-8460m), (12400-15300m), (19500-24000m). So a gain is observed for 26.64% of the whole BS coverage.

3.2 Scenario II: Stop and Start Movement

We assume that a mobile node stays for 60 seconds in a place where the SNR (at the BS receiver side) is around 16dB, then it takes 20 seconds to move to the place where the SNR drops to 14.5dB, and stays there for 100seconds.

Fig.6 shows the SNR evolution, the default and FEC based UL Goodput. We observe that, when SNR drops to around 14.5, the FEC based mechanism offers a higher goodput (16QAM3/4 with FEC applied) compared to the default one where 16QAM1/2 is applied. The goodput gain is around 1.15.

The simulation results show that our proposal always offers a higher UL goodput compared to the default method. Since the FEC mechanism is implemented in the high layer, our proposal does not induce modifying the WiMax standard but simply requires a simple thin interface between the MAC and network layer for the management of the mapping database and high layer adaptive erasure code. This practical consideration makes our proposal compliant with a fast deployment.

4 Conclusion

In this paper we have introduced a new cross layer based modulation adaptation mechanism which leverages on high layer (i.e.IP layer) adaptive erasure code and low layer information such as SNR and packet loss rate to significantly improve the goodput and transmission efficiency of WiMax connections. The proposed self-learning mechanism is “adaptive” to the varying channel conditions. Such an approach also results in a smoother goodput evolution rather than the default coarser evolution entailed by the

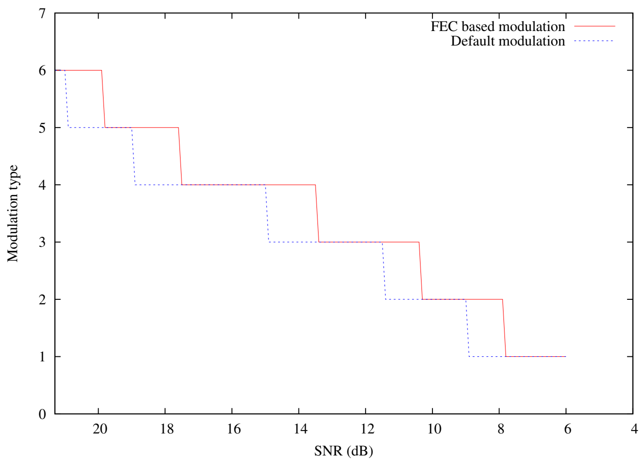

Figure 5. Modulation/coding scheme in terms of SNR

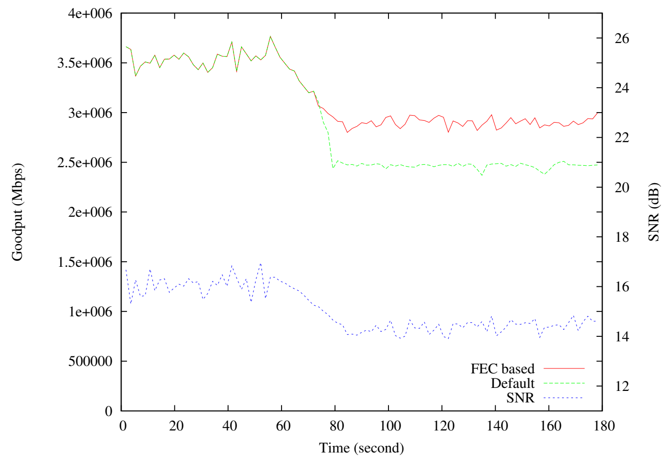

Figure 6. SNR evolution and Default/FEC based UL goodput

brutal modulation rate changes. Indeed, instead of downgrading the transmission rate immediately when signal degrades, we propose the mobile node to keep its current high transmission rate and enforce its flows reliability with FEC redundancy packets at higher layer. The analytical study and the simulation results introduced in this paper demonstrate a significant improvement on the goodput and transmission efficiency offered to WiMax mobile nodes. Our future work will focus on a study about the speed of convergence and auto-stabilisation of the proposed mapping function for various communication contexts.

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