Novel link adaptation for TETRA cellular systems

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SUMMARY
One of the main challenges for future wireless systems is to enhance the effective data throughput by exploiting the allocated bandwidth as much as possible. Among several approaches at different layers, one of the most important is constituted by the so-called link adaptation (LA) techniques. They are characterized by the adaptation of a set of transmission parameters to the channel state in order to improve performance. In this context, this paper is focused on the analysis of a particular class of LA techniques called adaptive modulation and coding, where the modulation and coding rate of transmission can vary according to the channel behavior. In particular, a novel LA algorithm, namely the timed window (TW) method, suitable for time-division Duplex systems is proposed here. The performance of the TW algorithm is evaluated by taking actual user mobility conditions, communication channel behavior, as well as the physical layer effects into account. Finally, it is important to stress that, even if the wireless bearer considered in this study is TETRA (TErrestrial Trunked RA dio), the approach is quite general and it can be of interest for other wireless networks and can be optimized for different channel models (e.g. TU50, HT200, etc.). Copyright © 2008 John Wiley & Sons, Ltd.

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1. INTRODUCTION

In this paper, a performance evaluation system for automatic link adaptation (LA) in time-division Duplex (TDD) cellular systems is introduced. We will focus on TETRA (T ERrestrial Trunked RA dio) technology, which is one of the prominent technologies for the Professional Mobile Radio/Public Access Mobile Radio (PMR/PAMR) market [1] of special interest for applications

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in communication systems for handling emergencies. TETRA specifications are constantly being
evolved by ETSI and new features are being introduced to fulfill the growing and ever demanding
Public Safety Digital Radio (PSDR) requirements. There is no doubt, however, that mobile broad-
band technology could greatly enhance and complement the present and future TETRA networks
and provide the advanced services envisioned in the next generation PSDR systems, such as remote
patient monitoring, 2-way real-time video, 3D positioning and Geographical Information System
(GIS), mobile robots, enhanced telemetry and so forth.

LA is one of the key enhancements proposed for TETRA Release 2 [2, 3]. In literature, the
LA algorithms refer to those that aim to optimize transmission parameters based on the state
of the communication channel [4]. Two well-known LA techniques are power control and adap-
tive modulation/coding (AMC). With power control, the transmitted signal power is adjusted in
order to guarantee a target carrier-to-interference-plus-noise ratio ($C/(I+N_t)$) at the receiving
end. With AMC, the power of the transmitted signal is held constant, but the modulation and
coding schemes are dynamically selected according to the received signal quality (i.e. $C/(I+
N_t)$ ratio). The problem of adapting the modulation and coding schemes to the actual channel
behavior has been widely discussed in literature [5–7]. Moreover, it is important to stress that
AMC techniques stand as an important means for supporting high throughput traffic and, hence,
multimedia services [8, 9] in several applications including emergency and disaster recovery. In
particular, AMC is used in some TDMA systems (e.g. EDGE) as well as in CDMA systems (e.g.
HSDPA), and it has been foreseen that the IEEE 802.16 standard will support broadband wireless
communications [10–12].

This paper focuses on an AMC-based LA algorithm suitable for a cellular TDD system. In
AMC systems, users close to the Base Station (BS) are typically assigned high-order modulation
and high code rates. The modulation-order and/or the code rate decreases as the distance from the
BS increases.

A cross-layer modeling approach is proposed here where the effects of mobility and propagation
on physical and data link layers are analyzed. Similar approaches have been adopted in literature
[13] without taking mobility and channel propagation conditions, physical and data link layers
jointly into account. This paper is organized as follows. Section 2 provides an overview of the
communication system of interest. The simulation model used in deriving the performance of
the proposed cross-layer system is also outlined in this section; moreover, the description of the
simulator modules is completed by focusing on the particular functionalities related to user mobility
and signal propagation, physical layer and finally LA. Numerical results are presented in Section 3,
while concluding remarks are drawn in Section 4. Appendix presents a detailed description of the
mobility and propagation models used in deriving the numerical results presented herein.

2. SYSTEM OVERVIEW

The operation scenario considered is a Mobile Station (MS) traveling within a TETRA cell. The
MS sends data packets to the Base Station (BS) on the uplink and receives acknowledgements on
the downlink. It is assumed that the downlink channel is error-free and the MS stays within the
cell. The uplink channel is subject to path loss and slow fading according to TU50 and HT200
[14]. Travel path of the MS is also modelled such that a user may define test paths as well as
realistic travel routes. This model is designed as a platform to evaluate LA algorithms in order to
maximize data throughput under varying propagation conditions.

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It should be noted that the TETRA Release 2 protocol is significantly different from other cellular systems, mainly because semi-duplex and simplex calls in PMR systems are of higher importance than full duplex (or hooked) calls. The TDD framing is structured in such a way that many slots (up to 68) are sent before ACKs and NACKs are issued. This leads to the need to design a particularly suitable LA algorithm based on ACK/NACK reception, as explained in Section 2.3.

System comprises of three core modules (see Figure 1). User-defined simulation parameters are fed to the relevant system modules for configuration. A detailed description of the capabilities and functionalities of each module is provided below.

The Mobility and Propagation Module (MPM) is devoted to the implementation of the following three functions: link budget analysis, path generation and uplink loss calculations. The link budget analysis is performed to obtain the cell radius based on user-defined parameters such as MS transmission power, BS sensitivity and antenna gains. The path generation is a function of the cell radius as well as a set of user-defined parameters obtained from a map. This map provides the travel path and street length, direction change probabilities and associated standard deviation. The instantaneous attenuation on the uplink includes mean path loss, loss due to slow-fading (i.e. shadowing). The median path loss calculation depends on the MS–BS separation. The shadowing loss is calculated using the second-order statistics with the aid of a Gaussian process. The shadowing model will be elaborated upon later. The output of the MPM is a list of path and shadowing loss samples that are calculated along the user-defined travel paths and environments.

The second module, i.e. the Physical Layer Module (PLM), used in our architecture is mainly devoted to implement the channel encoding/decoding, QAM modulation/demodulation as well as the error injection functions of the communication channel. The propagation loss values provided by the MPM are used for demodulation in the PLM. PLM operates on TETRA timeslot reference where each timeslot is either positively (ACK) or negatively (NACK) acknowledged. The output of the PLM is a timeslot acknowledgement map along the MS travel route.

The Link Adaptation Module (LAM) is the last module in the processing chain. Its main function is generating continuous or bursty packet data traffic. LAM uses the acknowledgement map information provided by the PLM to determine whether a generated packet is assigned an ACK or an NACK. Depending on the retransmission strategy used in the LA, packets with NACK are retransmitted. Depending on the LA algorithm deployed in the LAM, a packet may either be successfully transmitted or failed. The transmission status of each transmitted packet is stored by the LAM for use in performance analysis. The key feature of LAM is to provide a time-efficient way to assess any LA algorithm without having to re-run MPM and PLM. This is achieved by assuming accurate synchronization of timeslots in PLM and LAM.

The performance analysis is performed using metrics such as ‘channel occupancy’, ‘throughput’ and ‘transmission efficiency’. These metrics are functions of propagation environment, LA
parameters and packet traffic types. In the following sections, the system components in Figure 1 will be described in more detail.

2.1. Mobility and propagation module

The link budget analysis in MPM is performed using the transcendental equation $g(R)$:

$$g(R) = p_{MS} + G_{MS} + G_{BS} - S_{BS} - L_{p}(R) - \sigma$$  \hspace{1cm} (1)

which is a function of the MS transmission power ($p_{MS}$), antenna gain of MS ($G_{MS}$) and BS ($G_{BS}$), sensitivity of the BS ($S_{BS}$), propagation loss between the MS and BS ($L_{p}$) and the scaled standard deviation of the shadowing loss ($\sigma$) [15]. Standard deviation scaling is done to achieve 95% cell coverage with log-normally distributed slow fading.

By applying the regula falsi or false position iterative method [16], the root of Equation (1) (i.e. cell radius $R$) is determined so that $g(R)$ converges to zero. In other words, the cell radius for a given set of parameters is the value where the link losses and system gains between the MS and the BS are equal. Cell radius is used for generating the MS travel path as explained in the Appendix. Although our model is flexible enough to generate paths in open area, suburban, urban environments, results in this paper are presented for a circular path at the edge of the cell, representing the most unfavorable conditions, which will be sufficient to provide the reader with a clear insight into the performance of the proposed LA scheme.

2.2. Physical layer module

The PLM uses the time resolution between the sample positions along the travel path and the total path loss, $PL$, at each sample point, to deliver a timeslot acknowledgement map for the simulated travel path.

The PLM supports 4, 16 and 64QAM schemes and includes a 16-state turbo code with code rates ($R_c$) $\frac{1}{2}$ and $\frac{2}{3}$. It also includes the models for TU50, HT200 and static channels. In our study, the PLM was set to deliver acknowledgement maps for three sets of modulation and coding schemes (MCS) in TU50 channel, which are 4QAM with code rate $\frac{1}{2}$ (scheme 1), 16QAM with code rate $\frac{1}{2}$ (scheme 2) and 64QAM with code rate $\frac{2}{3}$ (scheme 3). Scheme 1 provides the highest data reliability while scheme 3 achieves the highest data throughput. Scheme 2, on the other hand, is a compromise between the two.

For a travel scenario, the PLM generates three acknowledgement maps. This is equivalent to a replay of a MS taking the same route in the same environment transmitting data using a different MCS each time. The output of the PLM is a list of ACKs and NACKs provided for each timeslot. This information is then fed to the LAM for testing any LA algorithm for a given route and propagation environment.

2.3. LA module

In order to adapt the MCS to the actual state of the wireless channel, a symbol error rate (SER) measurement is used. We introduce a SER estimation method using the reverse channel where ACKs and NACKs are received. The SER is estimated by using the ACK/NACK statistics obtained over a number of consecutive slots. It is assumed that the ACKs are received with acceptable delay and during the delay period new packets can be sent. The SER method allows estimation of channel behavior without overhead signaling (at least in the case of reliable communications where an
explicit ACK request is expected), while in current cellular systems, feedback of the channel quality indicator (QI) is needed, which consumes more resources than the proposed scheme. In order to provide the reader with some insight into the resources used in cellular systems, GSM system will be briefly explained next.

In GSM, the receiver performs link quality measurements (yielding QI) for switching between 4 codec modes [17]. The QI is calculated both on the uplink and the downlink for suitable codec selection at both ends. The codec selection is also transmitted using inband signaling that consists of frequent signaling and robust signaling based on frame stealing.

Frequent signaling is used for codec mode indications, commands and requests. Such control information is transmitted using part of traffic channel capacity normally used for speech. Synchronization of the receiver and the transmitter is also required for this signaling to work.

Robust signaling is designed to allow codec adaptation without interrupting the speech transmission by using the Robust AMR (Adaptive Multi-Rate) Traffic Synchronized Channel (RATSCCH). The RATSCCH protocol consists of a number of request (REQ) and three acknowledgement (ACK) messages. In a REQ-ACK cycle, precise handling of four counters is required as well as synchronization of the transmitter and the receiver. RATSCCH messages consist of 35 bits that are used for coding codec configuration, switching thresholds, acknowledgement and phase information elements.

Our attention has been focused to an ACK/NACK adaptation method because in the TETRA standard more attention has been put on this type of link adaptation with respect to alternatives based on a direct exploitation of the received signal-to-noise ratio (SNR). Moreover, as TETRA is quite bandwidth limited and, hence, it makes sense to discard such sophisticated schemes and rely on ACKs/NACKs.

After choosing the most suitable MCS based on the channel behavior, the problem of when to switch between the available schemes arises. One suitable technique is based on the number of ACKs and NACKs received over a certain time period. Based on the statistics, a suitable MCS selection can be made. In Figure 2, the proposed time window (TW) ACK/NACK-based algorithm is presented.

By measuring the signal plus interference noise ratio (SINR) at the receiver before starting the communication session, the selection of the best MCS is achieved. In Figure 2, after the SINR measurement, a more suitable MCS is selected. The selection is based on the target frame error rate (FER) that is strictly related to the traffic class and its QoS parameters in terms of the expected FER. After a certain number of consecutive retransmissions, the packet is discarded to avoid wasting bandwidth.

The above scheme applies to the recently transmitted segments as well as to retransmissions because the channel behavior affects both in the same way. If a counter takes into account how many times each segment has been transmitted, whenever it approaches to the last retransmission (e.g. due to delay constraints) the most reliable MCS can be used. Also, irrespective of the maximum number of retransmissions, the last two attempts always use the most reliable MCS.

As depicted, channel behavior is modeled based on the $\rho = \text{ACKs}/(\text{ACKs} + \text{NACKs})$ ratio within a TW. Once thresholds (Th1 and Th2) are suitably defined, it is possible to choose the best MCS for the radio environment of interest. The MCS switching parameters are defined as follows:

- $\rho < \text{Th1} \Rightarrow \text{NACKs dominate}; \text{ choose lower-order MCS.}$
- $\rho > \text{Th2} \Rightarrow \text{ACKs dominate}; \text{ choose higher-order MCS.}$
- $\text{Th1} < \rho < \text{Th2} \Rightarrow \text{Ratio within neutral range}; \text{ use current MCS.}$
Figure 2. TW-LA algorithm.
Table I. MCS selection for the next transmission window.

<table>
<thead>
<tr>
<th>The last $x%$ of the transmission window</th>
<th>$\rho&lt;\text{Th1}$</th>
<th>$\text{Th1}&lt;\rho&lt;\text{Th2}$</th>
<th>$\rho&gt;\text{Th2}$</th>
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<tbody>
<tr>
<td>$\rho&lt;\text{Th1}$</td>
<td>L</td>
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<tr>
<td>$\text{Th1}&lt;\rho&lt;\text{Th2}$</td>
<td>L</td>
<td>S</td>
<td>H</td>
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<tr>
<td>$\rho&gt;\text{Th2}$</td>
<td>S</td>
<td>H</td>
<td>H</td>
</tr>
</tbody>
</table>

It is worth noting that an ACK is requested for each segment, where the segment is part of a packet and an acknowledgement is always transmitted using the most reliable scheme. Moreover, if an ACK has not been received within a time period, the segment has to be retransmitted.

The ratio between ACKs and NACKs within a transmission window provides an acceptable initial estimation about the channel behavior, but in some cases it is not sufficiently accurate. To address this shortfall, it is possible to use the FER measurements in two ways; one using the entire transmission window and the other considering only the last $x\%$ of the window, which can be optimized according to the channel behavior. In order to improve decision accuracy, the information from the entire window and the last part are jointly used in this study. In Table I, a decision scheme is listed, where $L$, $S$ and $H$ stand for switching to the lower MCS, using the same MCS and switching to the higher MCS, respectively.

Performance can be improved if the optimal MCS is selected at the first attempt. This can be achieved by using the scheme adopted in the last successful transmission between the same pair of devices. This statement is true when many segments are sent continuously at low user mobility (i.e. the boundary conditions in terms of fading and shadowing remain close between successive packets).

As a benchmark comparison the ‘Start high go low’ (SHGL) algorithm will be considered. In Figure 3, the algorithm is shown where the flowchart iterations continue until a packet has been completely sent; whenever a new packet has to be sent the algorithm is reinitialized. It is assumed that each packet can be retransmitted for up to six times.

3. NUMERICAL RESULTS

In this section the TW algorithm will be evaluated by considering different threshold values. The results will be compared with those obtained by resorting to the SHGL algorithm. The simulation parameters used in the LA algorithm evaluation phase of the simulations are listed below:

- $\text{MaxAttempts}=6$ (this is the maximum allowed retransmission attempts).
- $\text{Transmission window}=10s.$
- $\text{Slot}=0.014167s$ (the duration of one TETRA timeslot).
- $\text{Frame length}=68$ slots (number of maximum timeslot allocation).
- $\text{Symbols per slot}=31$ (the number of modulation symbols per timeslot).

It should be noted that a truncated Pareto distribution for message length is used, while the message interarrival time is modeled using an exponential distribution. It is worth noting that the
Figure 3. The start high go low algorithm.
The truncated Pareto distribution is

$$
    f(x) = \begin{cases} 
        \frac{x \cdot k^x}{x^{2+1}}, & k \leq x \leq m \\
        \frac{k^x}{m^{2}}, & x > m 
    \end{cases}
$$

where \( x = 1.1 \), \( k = 81.5 \), and \( m = 66666 \). This is a suitable model for web traffic proposed for performance evaluation in 3GPP standard [18].

Each slot can be made up of a different number of segments, where a segment is an atomic group of symbols, in terms of their modulation and coding; for the 50kHz bandwidth used in the simulations, in each 4QAM slot there are 403 bits, excluding the MAC headers. Whenever data arrive at the data link layer, it is divided into segments. In both the uplink and the downlink, one slot at 50kHz bandwidth can be filled with one of the following:

- one segment of 4QAM \( R_c = \frac{1}{2} \)
- up to two segments of 16QAM \( R_c = \frac{1}{2} \)
- up to four segments of 64QAM \( R_c = \frac{2}{3} \)

Moreover, segments with different MCS cannot co-exist in the same timeslot. In the simulated scheme, assumed in deriving numerical results, if a new segment requires a different modulation and coding, it is sent in the next slot and the rest of the current slot is left empty.

In deriving our numerical results, first of all we have considered the performance of the proposed LA algorithm by varying the values of the thresholds (Th1 and Th2). Numerical results have been obtained by considering a circular path on the edge of the cell, because it is the most unfavorable environment resulting in the highest number of transmission errors from the physical layer, and allowing us to appropriately assess the behavior of the considered techniques. The performance of the LA algorithms was measured in terms of:

- **Throughput**: The throughput figure shows the average number of successfully transmitted bits per timeslot.
- **Delay**: This figure represents the average time spent in the queue by each segment; it is calculated for both correctly delivered segments and for discarded segments, by measuring the time that each segment spends in the transmission queue. The delay, in this case, is calculated as the difference between the time a segment is deleted from the transmission queue and the time the message of that segment enters the queue.
- **Transmission efficiency**: This shows the number of quarter time slots (QTS) required to successfully deliver a segment. The QTS is defined as a quarter of a timeslot and corresponds to the amount of time needed to send one 64QAM segment; this means that one 16QAM segment occupies two QTS and one 4QAM segment occupies four QTS, i.e. one slot. The QTS is provided for both the failed slots and failed segments. In this plot, the failed segments are assumed to be delivered successfully using 4QAM in an imaginary 100% reliable seventh transmission.
- **Channel occupancy**: This gives the proportion of time the MS is actually transmitting. It is expressed as the percentage of resource occupied. It gives a measure of occupancy independent of the error status of segments.
- **Discarded packets**: This figure represents the percentage of discarded packets because the maximum number of attempts of retransmission has been reached.
Note that the maximum achievable throughput (in 50kHz bandwidth) is when all the symbols within a slot are transmitted using 64QAM and $R_c$ equal to $\frac{2}{3}$:

$$\text{(bits/slot)}_{\text{max,50kHz}} = N \cdot \log_2 M \cdot R_c = 403 \cdot 6 \cdot \frac{2}{3} = 1612$$

(3)

where $N$ is the number of symbols within a 50kHz slot, $M$ is the modulation order, so that $\log_2(M)$ is the number of bit per symbol and $R_c$ is the coding rate.

The numerical results are focused on the above-described performance factors, for a circular path along the edge of the cell. For the sake of simplicity, the results have been reported only for the Poisson distributed mean interarrival message rate ($\lambda$) of 15msg/s. The performance is derived by varying the Th1 value from 0 to 1 and the Th2 value from Th1 to 1 (as explained in Table I the Th2 is the higher threshold and so make no sense to chose it for values lower than Th1), and by fixing the most recent timing window to the 20% of the transmission window. Figure 4 shows the performance of the proposed method in terms of throughput. It is possible to note in this figure that higher values of Th1 result in lower throughput; this is because when lower Th1 values are selected the system usually chooses lower modulation order. Similarly, by fixing a certain value of Th1, it is possible that higher Th2 values result in lower throughput. Also, there is a range of values for which the throughput is maximized. This is achieved when Th1 is between 0.1 and 0.4 and Th2 is lower than 0.9. Within these values the system has a good compromise between modulation efficiency and error probability, hence, maximizing overall throughput.

Figure 5 shows the performance of the proposed scheme in terms of packet delay. The delay profile for different values of Th1 and Th2 is similar to that obtained for the throughput. The delay is minimized within a certain threshold interval, in particular for Th1 between 0.2 and 0.5 and for Th2 lower than 0.9 we can obtain the best delay performance. Note that when the thresholds are
Figure 5. Performance results in terms of delay.

Figure 6. Performance results in terms of transmission efficiency.

higher the delay becomes very high, mainly due to the fact that lower modulation order schemes are used and hence a longer transmission time is needed for each segment.

In Figure 6, the performance of the proposed scheme is given in terms of transmission efficiency. The average QTS figure gives of the average number of QTS needed to successfully transfer
one segment. This means that if QTS is equal to 1, the segment is successfully delivered at the first attempt with 64QAM, while higher values mean more segments are needed, including more retransmission or lower modulation order. The average QTS values are within 2 and 4 even if an optimal value is achieved within a certain thresholds interval, in particular for Th1 between 0.1 and 0.5 and for Th2 lower than 0.9 we can obtain the best performance in terms of transmission efficiency.

In Figure 7, the performance in terms of discarded segments is shown. It is possible to note that for higher threshold values, the number of discarded segments is lower. This is because if we select lower thresholds, the system is forced to send data with a higher modulation order, implying also higher error probability. This reflects in a higher number of discarded segments (i.e. segments that exceed the maximum number of six attempts).

Finally, we compare the performance of the proposed algorithm, also by considering different values of the recent ACK/NACK timing window (see Table I), with the SHGL performance with respect to a variable message arrival rate \( \lambda \). In particular, we will fix the thresholds to Th1 = 0.4 and Th2 = 0.5 that give one of the best combinations in terms of performance.

It is possible to see that by varying the recent ACK/NACK timing window, it is possible to improve or deteriorate the performance. In particular, for the circular edge path, a smaller window gives better performance. In Figure 8, it is possible to see the performance in terms of throughput, delay, transmission efficiency and channel occupancy for the proposed algorithm with three different timing windows (10, 20 and 50%) compared with the SHGL. It is possible to see that with a suitable threshold and the recent ACK/NACK timing window selection, the system can outperform the SHGL algorithm.

With reference to this, we can point out that even if the improvement introduced by the proposed algorithm depends on the optimization procedure (i.e. optimal selection of the threshold values), this process is quite general and is independent of the actual propagation conditions (e.g. application...
Figure 8. Performance comparison among the proposed LA algorithm with three different more recent ACK/NACK timing windows, with the SHGL in terms of throughput, delay, transmission efficiency and channel occupancy.

environment). As a consequence, the additional complexity in implementing the proposed approach can be considered negligible with respect to that required by the SHGL method.

4. CONCLUSION

A LA technique for TDD cellular system, with a particular focus on TETRA has been presented. The performance of the TW algorithm has been evaluated by resorting to a suitable simulation tool that integrates function of physical and link layers as well as MS mobility. The proposed cross-layer approach allows a more accurate algorithm evaluation with a small increase in computational overhead and reduced transmission overhead compared with cellular systems. Numerical results show that the proposed TW allows for easy adaptation to the transmission environment by varying the thresholds and the transmission window size. The optimal thresholds have been found to be around 0.20–0.40 for the lower threshold and up to 0.90 for the higher threshold.
APPENDIX A: MOBILITY AND CHANNEL MODELS

Path generation is performed using the probabilistic model explained in [19]. The MS starts its travel at a random point at the cell edge. The mobility function of the MS is represented as $f_{\text{mob}}(\varepsilon_i, l)$, where $\varepsilon_i$ is the angular direction change at point $i$ in radians and $l$ is the street length between two possible direction changes. Note that the value of $l$ depends on the geographic characteristic of the travel environment. In urban areas, direction changes can occur frequently due to the high number of crossroads and streets. However, in rural environments this is not the case. Therefore, the value of $l$ in urban areas is smaller than that in rural areas.

The direction change variable, $\varepsilon_i$, has a probability density function (pdf), which is the sum of four Gaussian processes:}
\begin{equation}
\text{pdf}(\varepsilon_i) = p_0 \frac{1}{\sigma_\varepsilon \sqrt{2\pi}} e^{-\varepsilon_i^2 / 2\sigma_\varepsilon^2} + p_{-\pi/2} \frac{1}{\sigma_\varepsilon \sqrt{2\pi}} e^{-(\varepsilon_i+\pi/2)^2 / 2\sigma_\varepsilon^2} + p_{\pi/2} \frac{1}{\sigma_\varepsilon \sqrt{2\pi}} e^{-(\varepsilon_i-\pi/2)^2 / 2\sigma_\varepsilon^2} + p_\pi \frac{1}{\sigma_\varepsilon \sqrt{2\pi}} e^{-(\varepsilon_i-\pi)^2 / 2\sigma_\varepsilon^2} \tag{A1}
\end{equation}

Note that each Gaussian process has a mean of a direction change, $\bar{x}$, of either $\bar{x} = 0$ rad (no change), $\bar{x} = -\pi/2$ rad (right turn), $\bar{x} = \pi/2$ rad (left turn) and $\bar{x} = \pi$ rad (U-turn). In addition, each distribution is scaled by four $p_x$ direction change probabilities where $\sum p_x = 1$. Each $p_x$ is specified according to the travel trends of the people and the vehicles in an environment. For example, if the business centers of a city are located to the left of the largest residential area, the $p_{\pi/2}$ term will be dominant.

The $\sigma_\varepsilon$ term in the above pdf depends on the road network pattern. An irregular road network pattern has a higher standard deviation than a Manhattan grid, where all streets are perpendicular to each other. A 100 km long path in a cell with radius 3000m is given in Figure A1 as an example; notice the path regularity for $\sigma_\varepsilon = 0$. The cell boundary has been calculated by considering a probability of 95% cell coverage, including the effects of path-loss, shadowing and multipath fading.

Once the full travel path is generated, the route is split into equidistant segments, equal to the product of the constant MS speed and the timeslot duration. The coordinates of the MS position at the start of each segment is stored and is used to calculate the MS-BS separation ($d$). Distance $d$ is used to calculate the overall path loss on the uplink ($L_p$), which comprises the median loss, $L_m$, and loss due to slow-fading (i.e. shadowing), $L_s$:
\begin{equation}
L_p = L_m + L_s \tag{A2}
\end{equation}

We have resorted to the modified Hata model for calculating $L_m$ according to the value of the MS-BS separation ($d$) in three main cases [20], which will be described in detail. The choice of the modified Hata model has been chosen because of the TETRA frequencies in use (i.e. 400 and 800 MHz) and the frequencies for which the modified Hata model has been developed (i.e. $150 \text{ MHz} < f \leq 1500 \text{ MHz}$), as well as the propagation environment. Note that $L_m$ is in units of dB and $d$ is in km. In the following, we will consider three cases depending on the value of $d$. The
Figure A1. Illustration of the effects of $\sigma_z$ on path generation. $\sigma_z = 0.125\pi$ (top) versus $\sigma_z = 0$ (bottom).

first case considers $d \leq 0.04$ km, where:

$$L_m(d \leq 0.04) = 32.4 + 20 \cdot \log_{10}(f) + 10 \cdot \log_{10}\left(d^2 + \frac{(H_b - H_m)^2}{10^6}\right)$$  \hspace{1cm} (A3)$$

where $H_b = \min\{h_1, h_2\}$, $H_m = \max\{h_1, h_2\}$, $h_1$ is the transmitter antenna height in meters, $h_2$ is the receiver antenna height in meters and $f$ is the frequency in MHz (400 and 800 MHz for
TETRA). It should be noted that the $1 \leq H_m, H_b \leq 200$ condition needs to be satisfied to ensure the accuracy of this model.

The second case refers to the situation in which $d \geq 0.1$ km; the model varies for urban, suburban and open area environments. Three additional parameters need to be defined to specify path loss for each environment; $a(f, H_m)$, $b(f, H_b)$ and $\alpha$

$$a(f, H_m) = [1.1 \cdot \log_{10}(f) - 0.7] \cdot \min(10, H_m) - (1.56 \cdot \log_{10}(f) - 0.8)$$

$$+ \max \left\{ 0, 20 \cdot \log_{10} \left( \frac{H_m}{10} \right) \right\}$$ (A4)

$$b(f, H_b) = \min \left\{ 0, 20 \cdot \log \left( \frac{H_b}{30} \right) \right\}$$ (A5)

$$\alpha = 1, \quad d \leq 20 \text{ km}$$

$$\alpha = 1 + [0.14 + 0.000187 \cdot f + 0.00107 \cdot H_b] \cdot \left[ \log_{10} \left( \frac{d}{20} \right) \right]^{0.8}, \quad 20 \text{ km} < d \leq 100 \text{ km}$$ (A6)

Having introduced the required parameters, we can now specify $L_m$ for the urban, suburban and open area environments. It should be noted that the path loss is accurate for $150 \text{ MHz} < f \leq 1500 \text{ MHz}$ frequency range.

$$L_{m, \text{urban}}(d \geq 0.1) = 69.6 + 26.2 \log_{10}(f) - 13.82 \log_{10}(\max(30, H_b))$$

$$+ [44.9 - 6.55 \cdot \log_{10}(\max(30, H_b))] \cdot [\log_{10}(d)]^2$$

$$- a(f, H_m) - b(f, H_b)$$

$$L_{m, \text{suburban}}(d \geq 0.1) = L_{m, \text{urban}} - 2 \left\{ \log_{10} \left[ \frac{\min(\max(150, f), 2000)}{28} \right] \right\}^2 - 5.4$$ (A7)

$$L_{m, \text{openarea}}(d \geq 0.1) = L_{m, \text{urban}} - 4.78 \left[ \log_{10}(\min(\max(150, f), 2000)) \right]^2$$

$$+ 18.33 \cdot \log_{10}(\min(\max(150, f), 2000)) - 40.94$$ (A8)

Finally, the third case considers the environment when $0.04 \text{ km} < d < 0.1 \text{ km}$; in this case the $L_m$ expression becomes:

$$L_m(0.04 < d < 0.1) = L_m(d = 0.04) + \frac{\log_{10}(d) - \log_{10}(0.04)}{\log_{10}(0.1) - \log_{10}(0.04)} \cdot [L_m(d = 0.1) - L_m(d = 0.04)]$$ (A10)

Note that the $L_m$ expressions in (A10) are calculated using one of (A7), (A8), (A9) depending on the type of environment we are interested in.
For our purposes, we have assumed MS antenna height equal to 1.5 m, BS antenna height 20 m and transmission frequency 400 MHz, so that $L_m$ can be written as a function of $d$ as some of the system parameters in our study are fixed:

$$L_m(d \leq 0.04 \text{km}) = 10 \cdot \log_{10}(d^2 + 3.42 \cdot 10^{-4}) + 204.44$$  \hspace{1cm} (A11)

$$L_m(d \geq 0.1 \text{ km}) = 35.225 \cdot \log_{10}(d) + \Delta$$  \hspace{1cm} (A12)

$$L_m(0.04 \text{ km} < d < 0.1 \text{ km}) = 18.252 \cdot \log_{10}(d) + 82.839$$  \hspace{1cm} (A13)

Note that, $\Delta$ in (A12) is the environment-dependent correction factor, which is initialized to 131.04, 122.97 and 104.43 dB for urban, suburban and open area environments, respectively.

The log-normal shadowing loss is calculated via a statistical method, which uses a correlation between two sample locations separated by $d_r$, along the travel path of the MS [21]. Given the loss at point 1 (denoted $L_{s1}$) and the normalized autocorrelation function $f(d_r)$, the mean of the loss at point 2 can be calculated as:

$$\mu(L_{s2}) = f(d_r) \cdot L_{s1}$$  \hspace{1cm} (A14)

Similarly, the variance of the shadowing loss at point 2 can be calculated using $f(d_r)$ and the propagation dependent variance $\sigma^2(e)$:

$$\sigma^2(L_{s2}) = [1 - f^2(d_r)] \cdot \sigma^2(e)$$  \hspace{1cm} (A15)

Once the mean and the variance are determined, the shadowing loss at point 2 can be calculated:

$$f(d_r) = \exp\left(\frac{|d_r|}{d_{corr}} \cdot \ln(2)\right)$$  \hspace{1cm} (A16)

Note that function $f()$ is dependent on $d_r$, as well as the de-correlation distance, $d_{corr}$. The $d_{corr}$ is the minimum separation between two sample points where the shadowing loss values become statistically independent. In our model, $d_{corr}$ is equal to 20 m while $d_r$ is equal to 0.78 m, which correlates the shadowing loss among 25 neighboring samples. The decorrelation distance ($d_{corr}$) is typically 20 meters in a vehicular test environment [21].

The $\sigma(e)$ parameter is determined according to $d$ and a range of values have been listed in [20]. In our model, $\sigma(e)$ is equal to 9 dB, which is suitable for all outdoor simulation scenarios where $d > 600$ m.

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