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REAL-TIME CHANNEL EMULATION FOR MOBILE COMMUNICATION TEST BEDS

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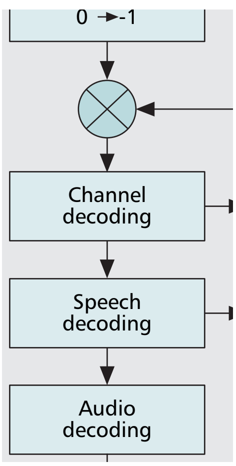

This article describes a methodology to model in real-time the channel quality variations for mobile communication systems. This method was used to implement a GSM hardware test bed employed to demonstrate the improved perceived voice quality obtained with link adaptation.

ABSTRACT

This article describes a methodology to model in real-time the channel quality variations for mobile communication systems. This method was used to implement a GSM hardware test bed employed to demonstrate the improved perceived voice quality obtained with link adaptation. Real-time operation is achieved by use of a large error pattern database, derived for time varying channel models. The modular design of the demonstrator and the reusability of the database enable a relatively straightforward extension to conduct novel multi-channel link level investigations. The proposed methodology can be adapted for a range of radio interfaces.

INTRODUCTION

As the complexity and capabilities of mobile communication systems increase, the use of simulation tools to evaluate the performance of new techniques or algorithms is becoming increasingly popular. Although suitable for providing an initial performance estimation, a crucial step to demonstrate feasibility is the development of hardware prototypes. A key challenge in the development of such prototypes is the emulation of real-time channel quality conditions. Such emulation is particularly important when analyzing the performance of adaptive radio resource management (RRM) techniques, such as link adaptation (LA), that dynamically manage the resources and parameters of a given communications session according to the system state and experienced channel conditions.

In this context, this article describes a novel and efficient methodology to emulate in real-time the radio channel quality variations in mobile communication systems. The proposed methodology models the radio transmission effects using a pre-recorded softbits database extracted from the receiver's equalizer. The softbits metric closely follows real bit error rate (BER) variations, demonstrating its suitability as a channel quality estimate. Consequently, the softbits database was used in a hardware emulator to demonstrate in real-time the effectiveness of LA under dynamically-varying-channel-quality conditions. The modularity of the demonstrator and the reusability of the softbits database enabled their evolution into a high-speed link-level emulation tool used to conduct novel investigations into multi-channel link-level performance.

REAL-TIME CHANNEL EMULATION

Channel quality estimation has been an intensive area of research over the last years. This research is driven by two important requirements: channel emulation and channel estimates for link-level adaptive RRM techniques.

Channel emulation investigations aim to obtain channel models that accurately represent realistic channel variations. The availability of such models is of paramount importance for developing test beds used to validate new air interface technologies. To achieve this objective, previous work developed channel emulation tools by means of real measured data [1] or hardware implementations [2].

Traditionally, system-level RRM studies assess the channel quality conditions using carrier to interference ratio (CIR) estimates [3]. Several investigations reported methods to achieve accurate CIR estimates in multipath fading channels [4]. Simpler proposals to obtain channel quality estimates also were reported in the literature. For example, [5] proposes to base adaptation decisions on the block error rate (BLER). The work reported in [6] bases the downlink channel-quality estimation on the transmitting power of the associated dedicated physical channel (DPCH) in high-speed downlink packet-access (HSDPA) systems. However, it is important to note that when deriving novel metrics, their relation to the actual link quality, measured in terms of BER, must be established [7].

In this context, this work proposes and implements a novel software channel modeling methodology that can be used for both channel emulation and channel quality estimation for adaptive RRM techniques. The proposed

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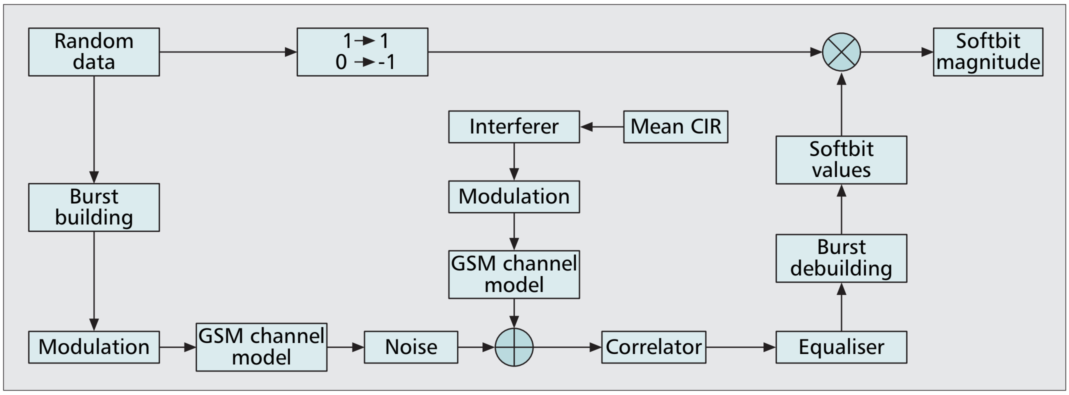


■ **Figure 1.** *Method of extracting the softbits database.*

The Viterbi algorithm maximizes the probability of the transmitted sequence, given the received samples. This is achieved by minimizing the Euclidean distance, or metric difference, between the received signal and all possible transmitted reference signals.

methodology, closely following BER variations and requiring a much lower computational cost compared to CIR estimates, extracts channel quality metrics directly from the transmission chain. In particular, the approach achieves real-time channel emulation by modeling the radio transmission effects using a large-scale error pattern database obtained from previous bit level simulations. The described approach is based on the global system for mobile (GSM) communications system but can, in principle, be adapted for a range of radio interfaces.

Softbits Database Extraction

The proposed channel emulation approach models the received bit-stream quality using a softbits database. This database was produced, as illustrated in Fig. 1, through bit-level simulations using the package now called CoCentric System Studio (formerly called COSSAP). The bit-level simulator is composed of three main parts: transmitter, radio transmission, and receiver. The transmitter's design follows the GSM/GPRS (general packet radio service) specifications and includes: formatting and building of a GSM normal burst and the Gaussian minimum shift keying (GMSK) modulator. The radio transmission includes the multipath radio channel, implemented as described in the GSM05.05 standard, and the effects of thermal noise and co-channel interference. The interference is modeled as a single, strong, random, and continuous GSM-modulated interfering signal following the guidelines in GSM05.05. The interfering signal is uncorrelated with the transmitting signal, and the same interfering pattern is used for all CIR levels. The design of the receiver part is only specified in the GSM specifications by means of performance requirements. In this work, the main blocks of the receiver are a correlator and an equalizer. The correlator uses the GSM training sequence to estimate the channel impulse response, which is then used by the Viterbi equalizer to recover the transmitted signal.

The Viterbi algorithm maximizes the probability of the transmitted sequence, given the received samples. This is achieved by minimizing the Euclidean distance, or metric difference, between the received signal and all possible transmitted reference signals. The algorithm is based on a recursion over the accumulated Euclidean distance for each of the 16 equalizer states. An equalizer state is represented by the history of the previously transmitted five bits. For each of the 16 new possible states, the Viterbi algorithm selects the most likely transition to the new state by comparing the accumulated path metrics of the two predecessor states plus the branch metrics. The most likely pre-state is stored in the survivor memory. The difference between the metrics of the two accumulated paths is stored for the soft decision. To allow for a quantized soft output, the variation of the energy from burst to burst must be compensated. To do that, the metric differences are normalized by a noise estimate. This noise estimate is extracted from the training sequence, as the average Euclidean distance between the received symbols and the reference symbols for the training sequence bits. Finally, the soft decision value produced for each data bit on the received burst is quantized with 10 bits. These softbits indicate the degree of confidence placed in the equalizer decision for the corresponding data bit. Positive values of a softbit indicate that the equalizer considers the received data bit to be equal to one, while negative values suggest the received data bit is equal to zero. The larger the amplitude of the softbit, the more reliable is the equalizer decision. The soft output average (SOA) channel metric is then defined as the average of the modulus of the softbits for each data bit of a GSM burst.

A softbit magnitude is defined as the softbit value multiplied by 1 if the transmitted data bit is binary one or by –1 if the transmitted data bit is binary zero. A softbit magnitude is then positive if the equalizer's estimation was correct and negative if it was incorrect. The softbit magnitudes, representing the radio transmission effects, are then stored so that they can be used in any process requiring the channel emulation. This is possible since the softbit magnitudes and the transmitted data bits are uncorrelated for the considered modulation scheme. The softbit magnitude is also statistically independent of whether any bit in a burst is a binary one or zero. As a result, the softbits magnitude database can be used to model the radio transmission of any data sequence.

A database of prerecorded softbits was generated to use in the hardware demonstrator. Softbit magnitudes were produced for single- and multi-slot transmissions, different mean CIR values, mobile speeds, and radio channels (typical urban and rural environments).

Real-Time Channel Emulation

The real-time implementation of a full transmission chain using the softbits database is illustrated in Fig. 2. To model the effects of the radio transmission, the softbit value of each received data bit is generated by mapping the transmitted data bits equal to a binary one to 1, and the ones equal to zero to –1, and then by multiplying the output of this mapping by the corresponding softbit magnitude. The softbit magnitude is extracted, from pre-recorded sequences, according to the scenario and experienced CIR. When simulating the physical layer, the channel coding output is first interleaved and then the propagation effects are added. The output of this sum is then de-interleaved before being passed to the channel decoding process. However, de-interleaving the softbit magnitudes, representing the propagation effects, and adding them to the channel coding output is equivalent. The last solution was adopted for the sake of simplicity.

Validation of the Softbits Proposal

To assess the softbits suitability as channel quality metrics, [8] investigated their relationship with the BER. The aim of this investigation was to demonstrate that the softbits closely follow real BER variations and exhibit a similar dynamic behavior. To conduct this investigation, the SOA values were first mapped onto BER estimates, and then their prediction error was compared against that obtained when the real BER sequence was known. The prediction error, defined as the difference between the real BER and the predicted BER — in this case, the BER observed in the previous time-division multiple access (TDMA) frame — not only provides information on how closely the softbits follow BER variations but also on their suitability as a triggering criteria for LA algorithms. In fact, LA algorithms are prediction schemes because they forecast channel quality conditions and the consequent best transport mode, based on previous channel estimates.

Figure 3 compares the BER prediction error obtained using the SOA metric against the one that could be achieved if the real BER was known.[1] As can be observed, the SOA prediction error is very similar to the one obtained from the real BER sequence itself. This indicates that the proposed softbit metric can be considered as a valid channel estimate, and that it also can be used to forecast the expected BER with the same degree of accuracy as could be achieved if the real BER was known.

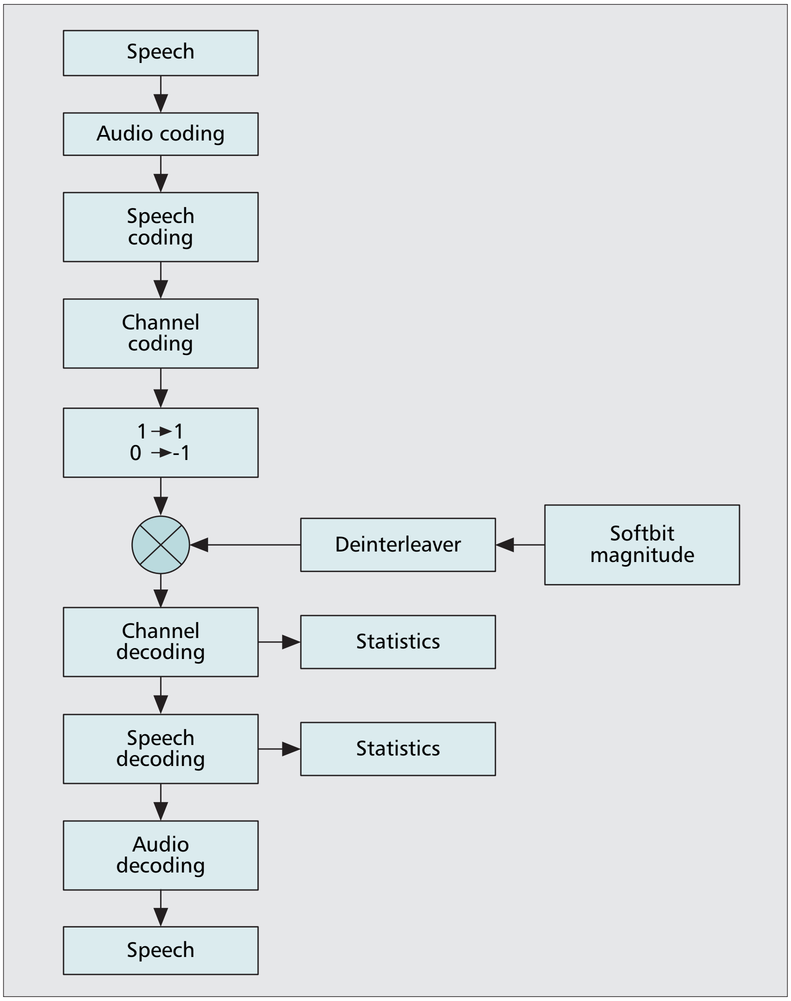
Hardware Demonstrator

To demonstrate the usability and potential of the real-time channel emulation approach being proposed, the authors implemented a hardware emulator used to demonstrate the effectiveness of LA to improve GSM speech quality under dynamically varying channel quality conditions. The demonstrator was developed in the Cairngorm project, a joint venture involving Motorola, Scottish Enterprise, and the University of Strathclyde. The implemented prototype was shown in a number of industrial events including the Embedded Systems Show, the UK's largest system engineering event.

Link Adaptation

LA has been identified as a key RRM technique implemented in evolved second generation (2G) systems, such as adaptive multi-rate (AMR), GPRS, or enhanced data rates for global evolution (EDGE), and in evolved third generation (3G) systems such as HSDPA. The basis of LA is to assess the short-term radio channel conditions and then use a transport mode (modulation and/or coding scheme) from a set of possible modes that is optimized for these conditions according to a predefined criteria. The different transport modes must provide different resilience to propagation errors under unfavorable radio conditions.

The hardware demonstrator implements a gross rate LA algorithm where the adaptation is performed between the standard GSM half-rate (HR) scheme and the extended half-rate (EHR) scheme [9]. The EHR format, proposed to illustrate the potential benefits of LA, uses the half rate speech codec output with extended error protection within a full rate channel. By increasing the error protection, the EHR scheme enables an increase in the robustness of the

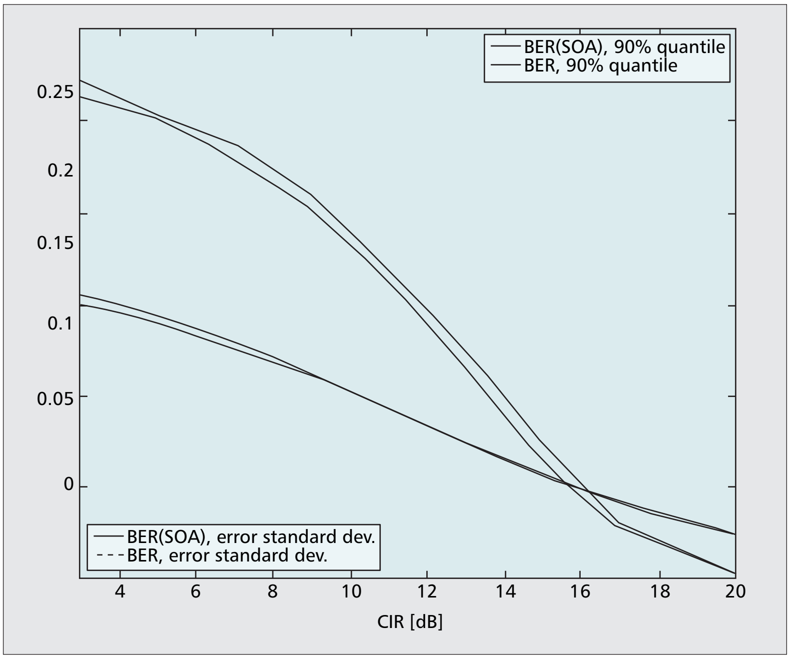


■ **Figure 2.** *Modeling a full transmission chain using the softbits database.*

[1] *The BER was obtained from extensive link-level simulations.*

speech channel under bad channel conditions. Such robustness can be seen in Fig. 4, which shows under varying CIR conditions, the performance of the standard GSM codecs — full-rate (FR), enhanced full-rate (EFR), and HR — and the EHR proposal. The HR codec exhibits a lower performance since it uses a traffic channel/half-rate speech (TCH/HS) channel, compared to the other schemes that employ a traffic channel/full-rate speech (TCH/FS) channel.[2] On the other hand, twice as many users may be supported by use of the HR codec. As a result, the implemented gross rate LA algorithm offers a trade-off between error protection and capacity.

It is interesting to note that the combination of the EHR, FR, and EFR codecs is very similar to the AMR-FR mode of the AMR codec. The modular design of the demonstrator and the possibility to extend the real-time channel emulation methodology to other systems, highlight the possibility to easily integrate AMR into the emulator and to further evolve the demonstrator to emulate other systems, as it will be demonstrated later with GPRS.



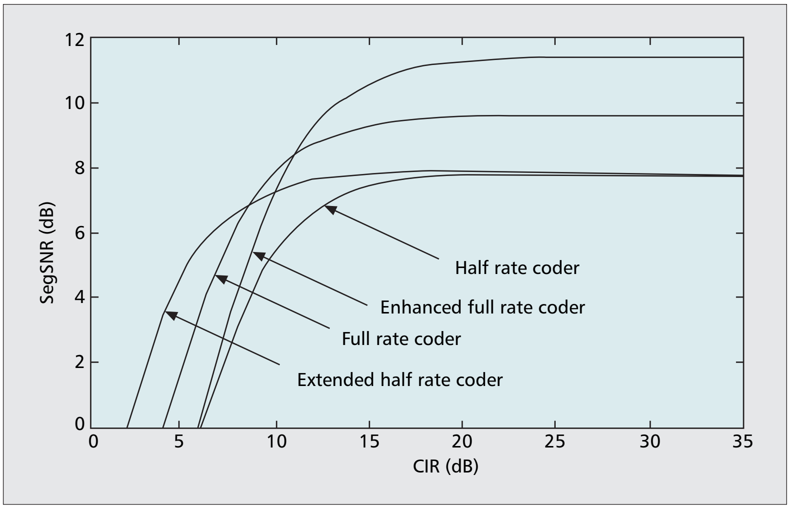
■ **Figure 3.** *BER prediction error.*

Structure and Components of the Demonstrator

The demonstrator is composed of three main elements: a PC and two Motorola DSP56307 evaluation module (EVM) boards. The PC includes the graphical user interface (GUI) and the GSM simulator. The GSM simulator models the radio channel and extracts, according to the experienced channel quality conditions (represented by the CIR), the softbits modeling the quality of the received bit stream. The softbits are transmitted to the slave EVM, which includes the channel coding and decoding, and carries out the LA process. The master EVM implements the GSM speech codecs and the audio interface. Both DSP boards work stand-alone, and the system works in real-time.

Following the GSM specifications, the master EVM produces 20 ms speech frames.[3] This EVM implements in parallel the three GSM standard speech codecs (FR, HR, and EFR) that were provided by Motorola as library modules. This parallel implementation enables a seamless switch between the codecs without any loss in speech quality as the selected speech codec is changed.

After receiving the speech-coded frames from the master EVM, the slave EVM performs the channel coding process. Four channel coding schemes are implemented in parallel: FR, HR, EFR, and EHR. In parallel, the slave EVM receives from the PC the softbits modeling the channel quality variations. The received softbits are de-interleaved, following either the FR or HR de-interleaving scheme, before being multiplied to the output of the channel codecs. The resulting data sequences are then passed to the channel decoders before being passed to the master EVM for performing the speech and audio decoding.



■ **Figure 4.** *Performance of the GSM codecs. Segmented SNR (SegSNR) is defined as the arithmetic mean of the SNR of voice packets where SNR is the ratio of the input vocal signal variance to reconstruction error variance. It is well correlated with mean opinion score (MOS) [10].*

The slave EVM also implements the described gross-rate LA algorithm. The algorithm bases its mode selection on the SOA channel metric; the work reported in [8] demonstrated its suitability as a channel estimate for LA algorithms. Considering the trade-off between capacity and error correction present with the considered codecs, an ideal selection algorithm is the one that selects the less protected mode (also offering the higher capacity) for which the speech frame can be transmitted without significant errors. As a result, the implemented switching thresholds were

[2] *While the TCH/FS channel uses the assigned physical channel in each TDMA frame, the TCH/HS channel uses only one time slot in alternate TDMA frames, which permits allocating two users to the same physical channel.*

[3] *The timing introduced by the audio coding process, and imposed by the GSM standard, is used to synchronize all the elements of the demonstrator, hence the term master EVM.*

selected so that the frame erasure rate (FER) is close to that obtained using EHR during the entire call, while the usage of EHR is kept low [9]. The LA algorithm uses a filter to produce the metric prediction for the next TDMA frame. Although the channel quality is estimated at each frame, a metric prediction is only produced once every eight TDMA frames in the hardware prototype. The triggering algorithm is based on a hysteresis window to reduce the likelihood of fast consecutive changes when the channel quality varies around the switching thresholds.

The real-time capability of the emulator is made possible by modeling the effects of the GSM transmission chain using the softbits database. During a simulation run, the PC extracts the corresponding softbits that are sent to the slave EVM, according to the experienced CIR and the modeled scenario.[4] The PC was equipped with a high-speed RS232 card to transmit the softbits to the slave EVM. The slave EVM uses the serial communication interface (SCI) port with a transmission rate of 307kb/s for this link. The master and slave EVM exchange speech frames using the enhanced synchronous serial interface (ESSI) interface working at a speed of 4.096 Mb/s. The real-time nature of the emulator and the timing imposed by the GSM required a careful definition and implementation of a synchronization and communication protocol between the different elements of the demonstrator [11].

Real-Time Performance

To obtain an initial indication of the expected perceived voice quality improvements that could be achieved with the implemented LA algorithm, a series of offline mean opinion score (MOS) tests were run. A MOS test is a method used to obtain a subjective assessment of the speech quality in a given system. For that purpose, the opinions of a group of listeners subjected to speech laboratory tests are collected during the experiment. The tests consider a five-point opinion scale: excellent, good, fair, poor, and bad. In the conducted experiments, the application of LA improved the voice quality by more than one point for medium speeds, passing from a perceived quality between poor and fair when not considering LA, to a perceived quality of close to good with LA. The LA voice quality improvement was less noticeable at low speeds. This is due to the fact that at slow speeds, multipath fades affect a significant number of successive data bursts. As a result, the interleaving process is not capable of randomizing (or spreading) channel errors, which significantly reduces the efficiency of the coding schemes, irrespective of the error protection. On the other hand, high speeds contribute to spreading propagation errors, producing shorter error lengths. In this case, robust coding schemes (such as EHR) are capable of correcting such errors, which results in a superior voice quality.

Figure 5 depicts an example of the improvements that can be obtained using LA in the real-time hardware emulator. Counting from the top, the first sub-plot shows the CIR variations in dB, and the second one indicates the BER at the input of the channel decoder. Considering that the implemented LA algorithm switches between the HR and EHR codecs, the third sub-plot displays the selected codec. Finally, the fourth and fifth sub-plots illustrate, respectively, the measured bad frame indicator (BFI) using only the HR codec and using LA. Importantly, as can be observed from Fig. 5, the use of LA can alleviate *black spot* situations. In the example considered, the link quality is considerably degraded, between the 10th and 20th second, as the CIR sharply decreases and remains at low levels. This sharp and sudden link quality decrease is detected by LA, automatically switching to the more robust coding scheme (EHR). Only after the quality improves during a given amount of time, will the LA algorithm switch back to the less robust coding scheme (HR). Considering just the interval between the 10 and 20 s markers, the FER is reduced from 3.3 percent when only HR was considered to 0.4 percent with the use of LA. This reduction produced a significant improvement in terms of perceived voice quality during the real-time evaluation.

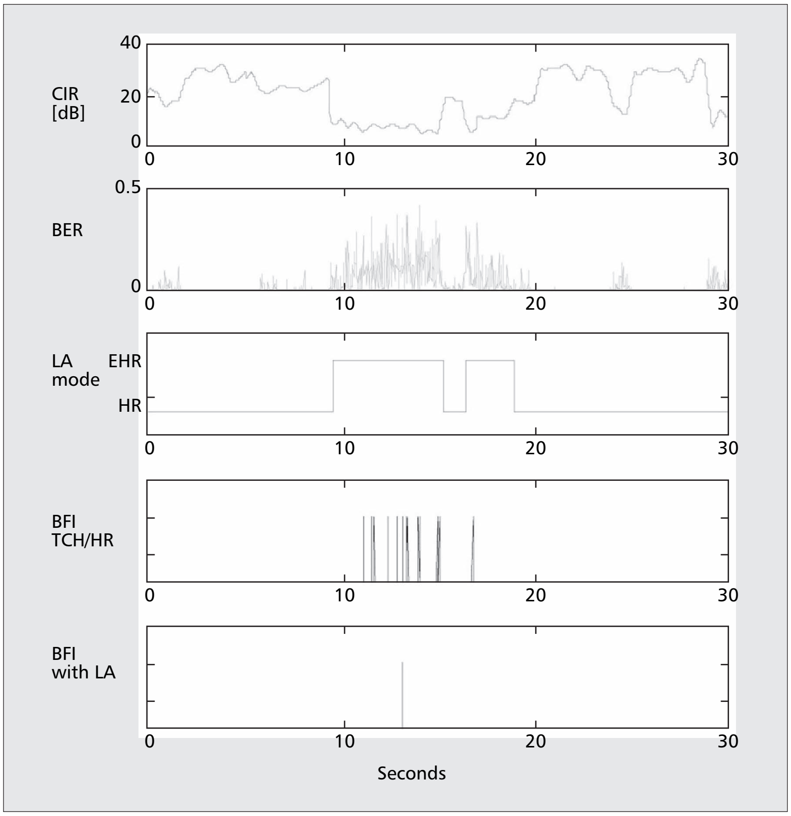


Figure 5. *Real-time LA performance.*

GPRS Emulation

The demonstrator's modularity and the reusability of the softbits database permitted the development of an enhanced software version of the emulator used to analyze the GPRS link-level

[4] A series of CIR sequences, previously extracted as a user moves within a closed circuit in an urban environment, is available. These sequences were obtained for different scenarios, varying the base station (BS) locations and user speeds (pedestrian and vehicular). Alternatively, the CIR sequences also can be produced manually through the GUI.

performance. Such extension has required the substitution of the GSM channel coding and interleaving schemes by their corresponding GPRS counterparts. The GPRS software emulator reuses the softbits database modeling the channel transmission errors because GSM and GPRS use the same physical radio frequency (RF) layer. A further extension to emulate the EDGE GMSK modes also was performed in [12]. It is important to emphasize that the development of softbits databases to model radio transmissions effects is not limited to GMSK-based systems since the described methodology also could be employed with different modulation schemes.

HIGH-SPEED LINK LEVEL SIMULATIONS

The implemented GPRS link-level emulation tool was used to produce different link-to-system level interfaces required for a complete evaluation of a mobile system performance.[5] The obtained link-level results [13] were in accordance with indications provided in the GSM05.05 standard, further validating the softbit proposal to model the radio transmission effects.

Employing the softbits database to model radio transmission effects resulted in a significant reduction of the simulation time, without sacrificing the accuracy of the radio link quality representation. In fact, the production of the softbits database[6] required more than 15 days of COSSAP simulations using a Sun ULTRA Sparc 1 workstation. That is approximately the time that will be required to analyze the GPRS link-level performance if the transmission chain and channel coding were simulated together using tools such as COSSAP. By decoupling the simulation of the coding schemes from the transmission chain, after the softbits database was produced, the same study took only eight hours using the same equipment, resulting in a significant reduction of 98 percent in simulation time.

MULTI-SLOT LINK-LEVEL ANALYSIS

Multi-channel (or multi-slot for TDMA) operation has become an interesting option in mobile radio systems to increase transmission data rates and improve user perceived quality of service (QoS). While most efforts have been devoted toward system-level and RRM aspects of multi-slot operation, no efforts have been dedicated to its link-level performance. In fact, current multi-slot system-level investigations employ link-level results obtained from single slot transmissions.

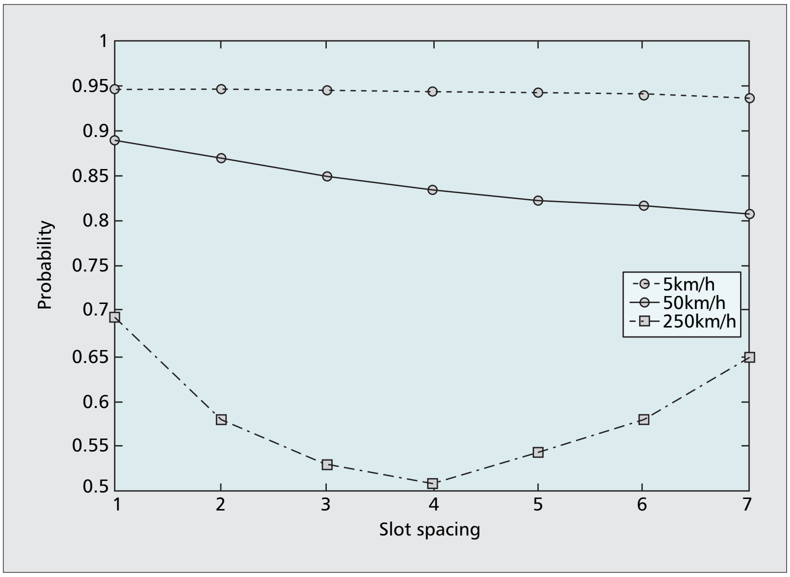


Figure 6. *Multislot link level performance.*

This modeling approach fails to represent the time properties of the link quality in channels allocated simultaneously to a single user by assuming that the transmission of data blocks in channels of the same frame exhibits a totally uncorrelated link-level performance. Since such properties can be highly relevant to the operation of RRM techniques, the softbits channel emulation methodology also was used to conduct novel multi-slot link-level investigations [14]. For that purpose, new softbits databases considering different interfering multi-slot scenarios[7] were produced following the procedure described in Fig. 1. For details on the multi-slot link level investigations, see [14]. Figure 6 demonstrates that the link-level performance in channels simultaneously allocated to a single user can be correlated and that the degree of correlation depends on a number of parameters, such as user speed and spacing between channels assigned to the same user (parameter referred as slot spacing). Figure 6 plots the correlation probability as a function of the slot spacing and the user speed.[8] The correlation probability parameter defines the probability that a data block transmitted using an arbitrary slot Y in four consecutive TDMA frames is received, after channel decoding, with the same state (error/no error) as a data block transmitted using another slot X of the same four consecutive TDMA frames. An interesting observation is that, depending on the user speed, the particular slots

[5] *To interface both levels, the link-level analysis is generally used as a source of information for the system level. In particular, the link-level performance is represented by a simplified model consisting of a set of look-up tables mapping the CIR to a given link quality parameter, such as the block error rate.*

[6] *The softbits database includes, for each mean CIR considered, 147,000 softbit metric values representing the channel quality conditions experienced during the transmission of 147,000 radio link control (RLC) blocks. As a result, the softbits database is able to take into account the effects of multipath fading and noise for the same mean CIR.*

[7] *In fact, in multi-slot transmissions, the interference in each assigned slot can come from the same interference source (also a multi-slot transmitter) or from different independent sources (e.g., single-slot transmitters).*

[8] *A spacing of zero slots corresponds to the case in which two data blocks are transmitted in contiguous slots of four consecutive frames. Taking into account that a GPRS frame consists of eight time slots, the maximum slot spacing is six slots. This figure corresponds to an average CIR of 8 dB and the GPRS coding scheme CS3.*

The modularity of the demonstrator and the reusability of the softbits database enable their extension to emulate other systems. In fact, the proposed channel emulation methodology also was used to efficiently conduct computationally-expensive link-level investigations.

selected for a multi-slot transmission can have an impact on the instantaneous multi-slot link-level performance. The investigations conducted also demonstrated the dependence of the multi-slot link level correlation on other parameters such as the interference scenario, the CIR, or the employed coding scheme.

CONCLUSION

This article presented a methodology to emulate channel quality conditions using a database of previously extracted softbit sequences. Such a database was crucial to enable the development of a real-time hardware prototype initially designed to demonstrate the effectiveness of LA. The modularity of the demonstrator and the reusability of the softbits database enable their extension to emulate other systems. In fact, the proposed channel emulation methodology also was used to efficiently conduct computationally-expensive link-level investigations.

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REFERENCES

[1] J. Kolu, T. Jamsa, and A. Hulkkonen, "Real Time Simulation of Measured Radio Channels," *Proc. IEEE VTC*, Oct. 2003, pp. 183–87.

[2] D. Noguet et al., "A Hardware Testbed for UMTS/TDD Joint Detection Baseband Receivers," *Proc. IEEE Int'l. Symp. Spread Spectrum Techniques and Apps.*, Sept. 2004, pp. 972–76.

[3] K. K. Leung *et al.*, "Link Adaptation and Power Control for Streaming Services in EGPRS Wireless Networks," *IEEE JSAC*, Oct. 2001, pp. 2029–39.

[4] A. Sampath and D. R. Jeske, "Analysis of Signal-to-Interference Ratio Estimation Methods for Wireless Communication Systems," *Proc. IEEE ICC*, June 2001, pp 2499–2503.

[5] P. J. Ameigeiras-Gutierrez *et al.*, "Performance of Link Adaptation in GPRS Networks," *Proc. IEEE VTC*, Sept. 2000, pp. 492–99.

[6] K. Miyoshi *et al.*, "Link Adaptation Method for High Speed Downlink Packet Access for WCDMA," *Proc. 4th WPMC Int'l. Symp.*, Sept. 2001, pp. 455–60.

[7] P. Gunreben *et al.*, "On Link Quality Estimation for 3G Wireless Communication Networks," *Proc. IEEE VTC*, Sept. 2000, pp. 530–35.

[8] J. Pons and J. Dunlop, "In-Service Link Quality Estimation for Link Adaptation Algorithms Applied to GSM," *Proc. IEEE Int'l. Conf. Universal Pers. Commun.*, Oct. 1998, pp. 1169–74.

[9] J. Pons and J. Dunlop, "Bit Error Rate Based Link Adaptation for GSM," *Proc. IEEE Int'l. Symp. Pers., Indoor and Mobile Radio Commun.*, Sept. 1998, pp. 1530–34.

[10] H. Shi, P. K. M. Ho, and V. Cuperman, "Combined Speech and Channel Coding for Mobile Radio Communications," *IEEE Trans. Vehic. Tech.*, vol. 43, no. 4, Nov. 1994, pp. 1078–87.

[11] J. Dunlop *et al.*, "Real-Time GSM Link Adaptation Hardware Demonstrator," *Proc. IEEE VTC*, May 2000, pp. 590–94.

[12] M. Mzyece, J. Dunlop, and J. Irvine, "Optimization of TCP Performance over EGPRS in Incremental Redundancy Mode," *Proc. IEEE VTC*, Oct. 2003, pp. 2574–78.

[13] J. Gozalvez and J. Dunlop, "Link Level Modeling Techniques for Analyzing the Configuration of Link Adaptation Algorithms in Mobile Radio Networks," *Proc. Euro. Wireless*, Feb. 2004, pp. 325–30.

[14] J. Gozalvez and J. Dunlop, "On the Effect of Correlation in Multislot Link Layer Analysis for GPRS," *Proc. IEEE VTC*, Sept. 2000, pp. 444–50.

BIOGRAPHIES

JAVIER GOZALVEZ (j.gozalvez@ieee.org) received an electronics engineering degree from ENSEIRB (Bordeaux, France), a D.E.A. in electronics from Université de Bordeaux I, and a Ph.D. in mobile communications from the University of Strathclyde, Glasgow, United Kingdom. Since October 2002, he is a lecturer at the Signal Theory and Communications Division of the University Miguel Hernández, Elche, Spain, where he received the 2005 Young Researcher Award. He currently serves as mobile radio senior editor of IEEE Vehicular Technology and as associate editor of IEEE Communication Letters. He also served on the organizing and/or technical committees of various international conferences including IEEE Vehicular Technology Conferences and International Symposia on Wireless Communications Systems and as a member of the IASTED Technical Committee on Telecommunications. He has researched various aspects of adaptive RRM techniques. His current research interests include RRM, heterogeneous systems, wireless sensor networking, and application of wireless communications to the vehicular environment. While working on his Ph.D., he received a Best Student Paper Award at WPMC in 2001.

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