Performance prediction of DAB modulation and transmission using Matlab modeling

Lukas M. Gaetzi and Malcolm O. J. Hawksford

Abstract — *A Simulink-Matlab simulation model is described that enables an accurate performance prediction of complete DAB (digital audio broadcasting) transmission channels. Embedded compiled C-code subroutines include modulation protocols, error correction and MPEG layer-2 perceptual audio coding. Rapid assessment of critical design related factors could be performed that include channel interference, multi-path reflection and a range of modulation-parameters. Software is PC compatible with both DAB system and transmission channel configurable using a bespoke graphical user interface, which facilitates changing on the fly modulation and transmission-path related parameters. Overall audio quality can also be assessed*[1].

Index Terms — DAB, Matlab, Modulation, Simulation.

I. Introduction to DAB

DIGITAL audio broadcasting (DAB) was developed in the early 1990's by the European consortium *Eureka 147* [1], mainly to replace the widely used analogue frequency modulation (FM) broadcasting system. The VHF band is a scarce resource in many parts of the world, so there was a need for a spectrally more efficient modulation method than FM. In DAB, this is achieved by multiplexing several programmes into a so-called ensemble with a bandwidth of 1.536 MHz, where the number of programmes per ensemble is flexible and depends on individual programme bandwidth requirements. Further, conventional analogue techniques do not provide satisfactory performance in a mobile environment, because they are highly affected by multi-path propagation and thus fading. In DAB, orthogonal frequency division multiplex (OFDM) has been chosen to overcome the effects of multi-path propagation, enabling the system to operate in so-called single-frequency networks (SFN).

The modeling and processing of a DAB system is well suited for Simulink and Matlab. Less common blocks can be implemented in Simulink by programming appropriate 'C'-code that can be compiled and dynamically linked to the model by means of *S-functions*. Earlier work focuses more on one particular component such as the fading radio channel [2], whereas the study in hand tries to embrace the whole system. The simulation of communication systems is a broad activity that combines elements from the simulation side with more traditional communication disciplines.

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A. DAB transmission system

The overall DAB transmission system can be broken down into a number of sub-blocks (Fig. 1). The audio signal is MPEG layer-2 encoded and then scrambled. Forward error correction is applied to the scrambled bit-stream by employing punctured convolutional codes with code-rates

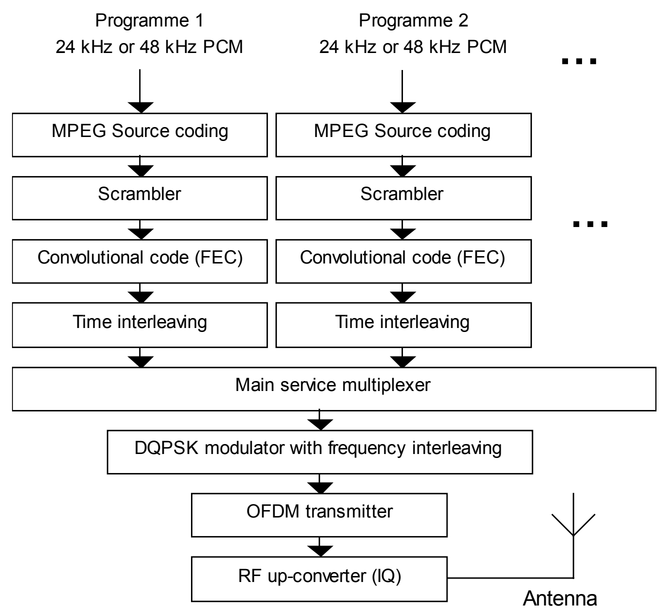

Fig. 1. DAB functional diagram of transmitter

ranging 0.25 – 0.89. The bit-stream is sent through a time-interleaver before it is multiplexed with the other programmes to form an ensemble. The ensemble bit-stream is fragmented into individual OFDM symbols, which are obtained by differential QPSK modulation of the sub-carriers and basically an inverse Fourier transform (IFFT) operation within the OFDM transmitter. In the receiver the corresponding inverse operations have to be carried out.

The information bit-stream is divided into a number of lower rate bit-streams in OFDM, which are individually modulated onto orthogonal sub-carriers. To achieve orthogonality sub-carriers are spaced in frequency by the inverse of the symbol duration, theoretically resulting in zero inter-carrier interference. Although the sinc(f) responses mutually overlap, they go through zero at center frequencies of all other sub-carriers, giving a spectrum efficiency of up to 2 (bit/s)/Hz for QPSK modulation of each sub-carrier [3]. Orthogonal sub-carriers can be realized with the IFFT algorithm, which can be readily integrated into hardware. Each sub-carrier is modulated with differential QPSK, which

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maps the incoming bits to complex symbols *G(k)* for each sub-carrier *k*. The last complex output values of the IFFT are copied to the front of the symbol to add the guard interval (cyclic prefix). It must be noticed that the complex symbols are frequency interleaved before being fed into the IFFT. Corresponding inverse functions are applied at the receiver. The clock of the receiver has to be synchronized exactly to the incoming signal, whereas the guard interval between the symbols is discarded. Then the FFT is applied, the individual carriers are DQPSK demodulated and the original bit-stream is regenerated.

The relatively high ensemble bandwidth of about 1.5 MHz gives good frequency diversity, since frequencies are not affected in the same way by fading. Adjacent bits within the MPEG bit-stream are made statistically independent with respect to bit errors by employing frequency and time interleaving, leading to good performance of the convolutional decoder (Viterbi).

B. DAB transmission modes

Technically, the DAB transmission system can be used in all VHF and UHF broadcasting frequency bands between 30 and 3000 MHz and four specific modes for typical applications have been defined (Table I).

TABLE I
THE FOUR DAB TRANSMISSION MODES

Mode	Total symbol duration	Main application
I	1246 µs	Terrestrial DAB, large coverage areas, VHF (Band III)
II	312 µs	Terrestrial DAB, small to medium coverage areas, UHF (L-Band)
III	156 µs	Satellite DAB, no long echoes, UHF (L-Band)
IV	623 µs	For Canada, between modes I and II

The total symbol duration consists of the principal symbol period and a guard interval, which prevents the echo of the previous symbol from interfering with the current symbol. By doing so, inter-symbol interference (ISI) is reduced to almost zero as long as the echoes from the various transmitters and propagation paths do not substantially exceed the guard interval. The maximum permissible propagation path difference Δ_d in meters can be calculated from the guard interval T_{guard} and the propagation speed v_c, as in (1), below.

$$\Delta_d = T_{guard} \cdot v_c \quad \left(v_c = 3 \cdot 10^8 ms^{-1}\right) \tag{1}$$

All modes have an ensemble bandwidth of exactly 1.536 MHz, but since the symbol duration and therefore the carrier spacing (inverse of the useful symbol duration) vary, the number of carriers that can be accommodated within the ensemble bandwidth differs from one mode to another.

If the receiver physically moves within the reception area, Doppler spread increases and temporal coherence of the channel is reduced. In addition the signal spectrum is Doppler shifted. If the received OFDM sub-carriers are shifted with respect to the reference frequency in the receiver, inter-carrier interference is increased. This can be understood by frequency shifting the sub-carriers shown in Fig. 2. The closer the sub-carriers are spaced, the more severe becomes inter-carrier interference, consequently symbol duration is a compromise: i.e. if symbol duration is too short then delay spread of the channel causes inter-symbol interference, while if symbol duration is too long then sub-carriers become closely spaced in frequency enabling already small Doppler shifts to produce high inter-carrier interference.

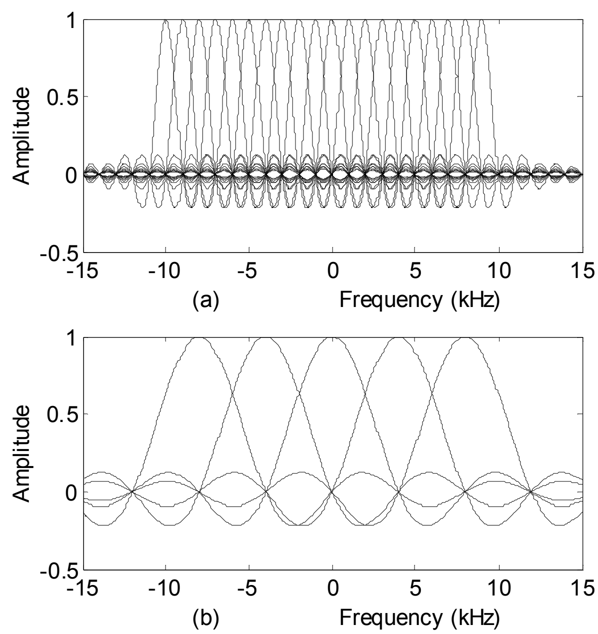

Fig. 2. Orthogonal sub-carrier spacing for DAB transmission modes I (a) and II (b)

II. MODELING

The communication channel impairs the OFDM symbols by attenuating the signal and by adding noise. Furthermore, multiple reflections can occur within the channel such that numerous copies of the signal arrive at the receiver. It is possible to comprehensively model such fading environments including Doppler shift and Doppler spread in case of a mobile receiver. An intuitive graphical user interface has been developed to enter and edit individual topologies of interest by arranging transmitters, reflectors and gap-filler repeaters around the receiver (Fig. 3). Fig. 4 shows the top-level Simulink model. The model is implemented in a hierarchical way and individual processing blocks can be expanded if required. No recursive loops are required within the model, which avoids the use of differential equation solvers and therefore minimizes the amount of processing required. Based on the band pass sampling theorem, it is not necessary to sample the system at twice the RF carrier frequency. It is more convenient to

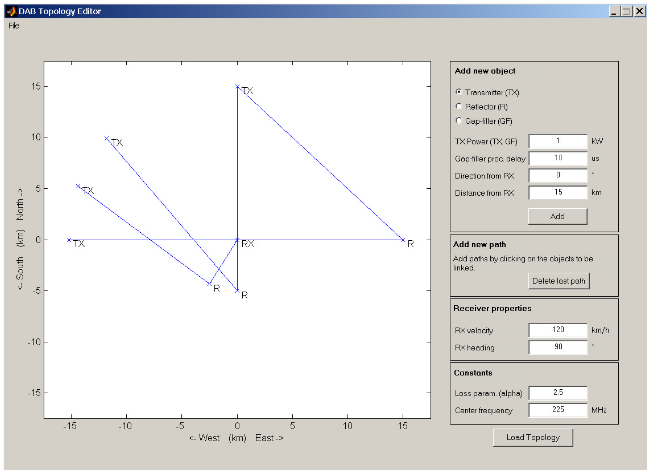

Fig. 3. Topology editor window as programmed in Matlab. Coordinates of objects and key parameters can be entered. Receiver movement can be entered as velocity (km/h) and direction (degrees).

work with equivalent complex baseband representations of the individual blocks instead [4].

A. *Simulink model timing*

The model uses frame-based processing, which means that whole frames of data are passed from one block to another rather than single samples. Frame-based processing is more efficient in terms of processing overhead because subroutines have to be invoked only once per frame. Another advantage of frame-based processing is its ability to handle multi-rate systems. The frame sizes are not constant throughout the DAB model, which means that at a given frame-rate (frames per second) the number of samples per frame, which are transferred between blocks, can vary. This is reasonable because for instance a convolutional encoder adds redundancy to the source signal and so increases the number of bits per second.

Due to complexity of the DAB system it was however impractical to work with only one frame-rate throughout the whole model. In fact two frame-rates are used, divided by the *Block partitioner* blocks that model the main service multiplexing as shown in Fig. 1. The first part of the model uses the MPEG frame-rate (every 24 ms), whereas the second part calculates one simulation frame per OFDM symbol (every 1.25 ms in transmission mode I). Simulink supports multiple frame-rates.

B. *Modeling of blocks*

Fig. 5 shows typical examples of sub-block models. The convolutional coder is shown in (a), which consists solely of standard Simulink blocks. The numbers in square brackets show the frame-sizes for DAB transmission mode I and it can be seen that they vary, leading to different bit-rates within the sub-block. The convolutional decoder in Fig. 5 (b) performs the inverse operations and employs a 4-bit soft decision Viterbi decoder. Fig. 5 (c) shows the OFDM transmitter, which maps 1536 complex DQPSK symbols to the corresponding sub-carriers, zero-pads to 2048 sub-carriers and then performs an IFFT to obtain a time-domain representation of the symbol. It must be noticed that both the real and imaginary parts of the obtained complex baseband signal have to be transmitted separately. Finally, the guard interval is added and normalization takes place. A typical multi-path Rice channel is shown in Fig. 5 (d). In a typical urban environment, reflection, scattering and refraction occur with objects that are located in vicinity of the propagation path between transmitter and receiver. Objects that are large relative to the wavelength of the carrier cause reflection whereas relatively small objects cause scattering. This results in different waves propagating along different paths creating an interference pattern in space, which in practice is not stationary due to fluctuations in the medium

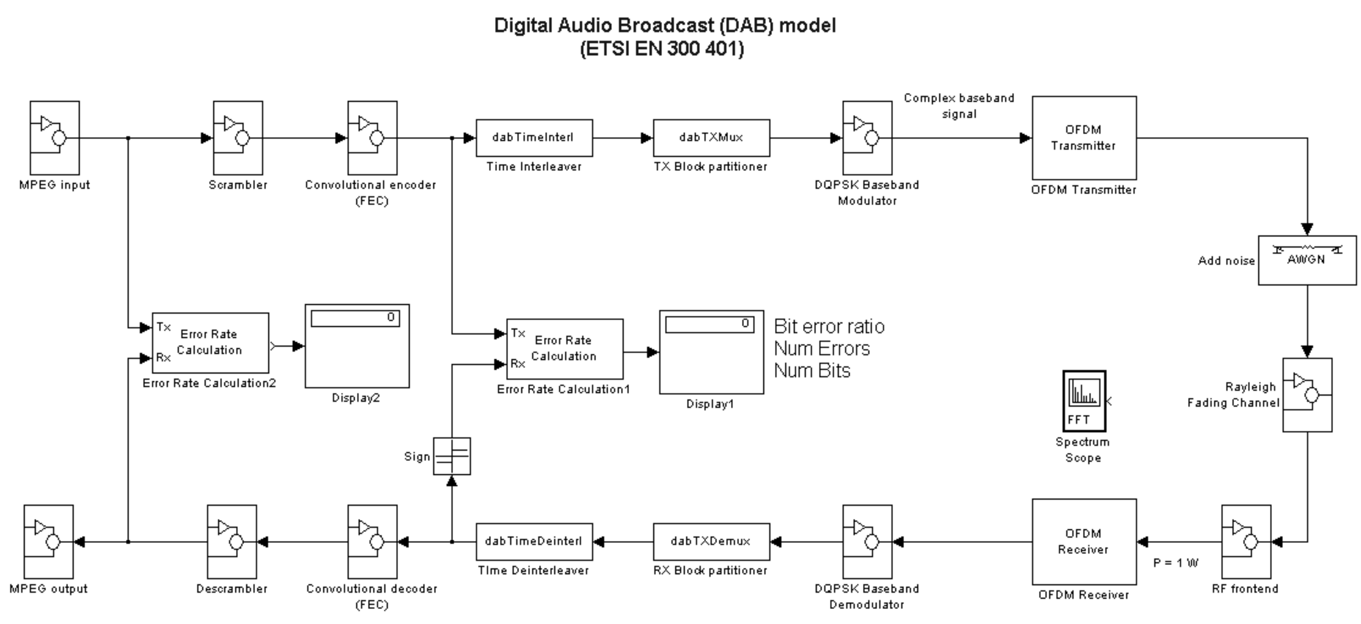

Fig. 4. Top-level Simulink model of the DAB transmitter (above), noise and multi-path channels (right) and the receiver (below).

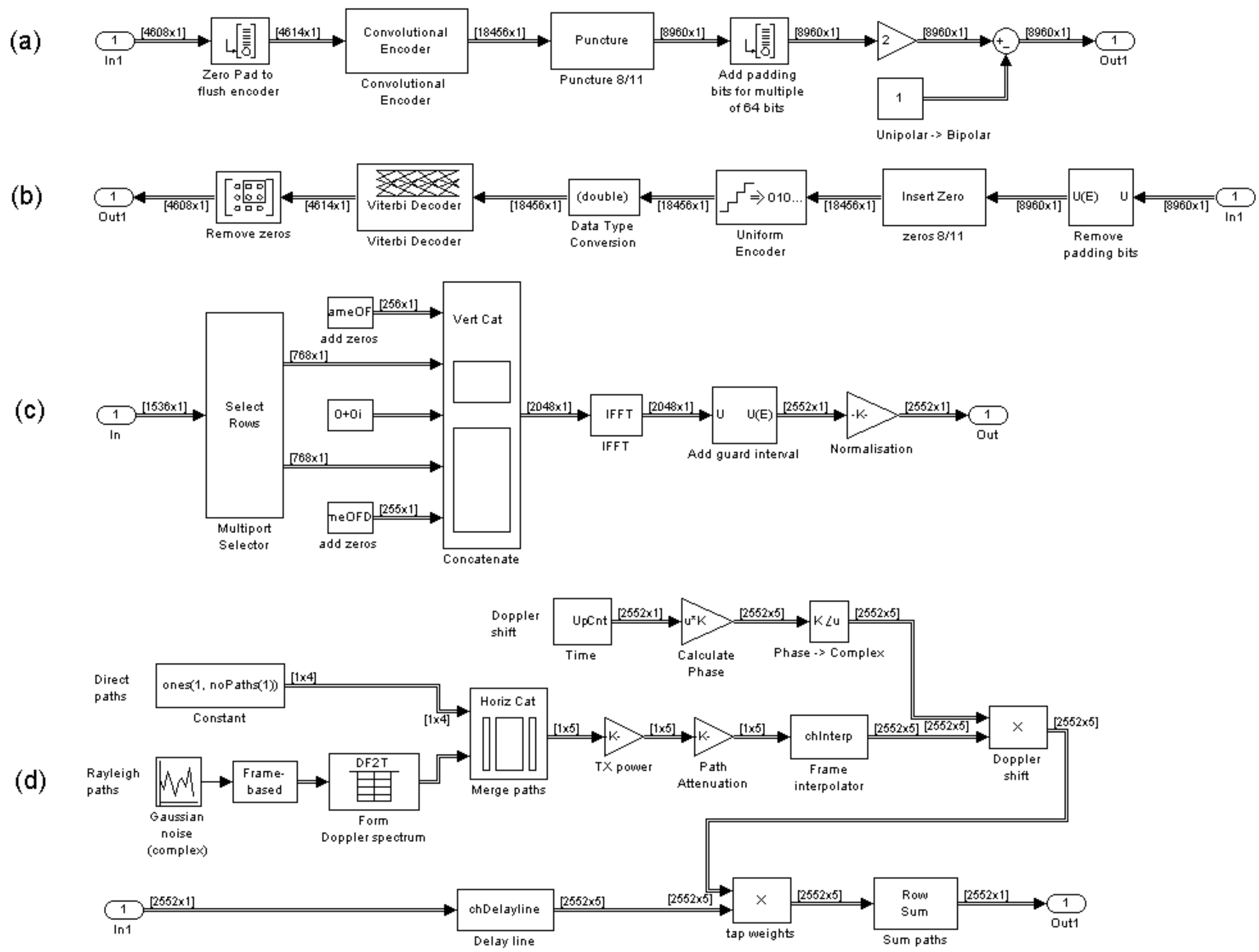

Fig. 5. Selected Simulink sub-blocks. Convolutional encoder (a) and decoder (b), OFDM transmitter (c) and Rice multi-path channel (d).

and because scatterers such as trees or vehicles may move. If a sinusoidal carrier is sent through a channel with time-variant amplitude and phase response, it is spread out in frequency. Delayed replications of the transmitted signal are created by the *Delay line* block, which had to be implemented as Simulink *S-function* in 'C'. Each copy of the signal is weighted by a complex time-variant coefficient as outlined in [5]. Finally, the *Sum paths* block superimposes all individual paths.

C. Model limitations and errors

Simulation models never fully correspond to the real world. There is always a trade-off between the accuracy of the model and the time required to develop it and to run simulations. Very detailed modeling may be justified if a hardware implementation is targeted and expected ranges of achievable parameters are known, e.g. slew-rates or clock jitter. The DAB model in hand differs from the real world at several points as follows:

- In the model the receiver is always perfectly synchronized to the transmitter. In reality clock drift and jitter occur which both degrade orthogonality and therefore increase inter-carrier interference.
- In reality receiver noise is not perfectly Gaussian, but a mix of 1/f noise, Gaussian and impulsive noise, depending on the exact hardware implementation.
- The RF front-end contains non-ideal filters that have not been modeled.
- Image rejection of the RF front-end is assumed to be infinite, which is not the case in reality resulting in additional in-band noise.
- Numerical precision of the digital circuits is usually around 10 bit in reality, whereas the model performs calculations in 64 bit floating point precision (*double*).
- Special propagation issues are not modeled such as Fresnel zones and path profiling or earth curvature.

III. Results

Whereas the previous section discussed the modeling, this section focuses on performance prediction by applying test bit-streams to the model.

A. Definition of performance criteria

In the interest of usefulness of the results, performance criteria have to be defined prior to running simulations. The optimization criterion for digital transmission of audio is not primarily the bit error rate, but to minimize perception of audio signal distortions in the case of bit errors. The annoyance of bit errors depends on their location within the MPEG frame. By means of a file selection dialog box, processed MPEG files could be played and listened to with headphones, such that audio quality could be analyzed for different puncturing schemes (protection levels). It was observed that the bit errors start to get significantly perceptible at BER higher than 10^{-4} in case of unequal error protection (onset of impairment).

B. Channel interference

The transmitted signal was exposed to wideband and narrowband interference and the bit error ratio was measured versus signal-to-noise ratio per bit. Fig. 6 shows the performance of different protection levels, representing different FEC code rates. It can be seen that levels 1 to 4 are equally spaced by about 1.1 dB whereas level 5 provides much less coding gain.

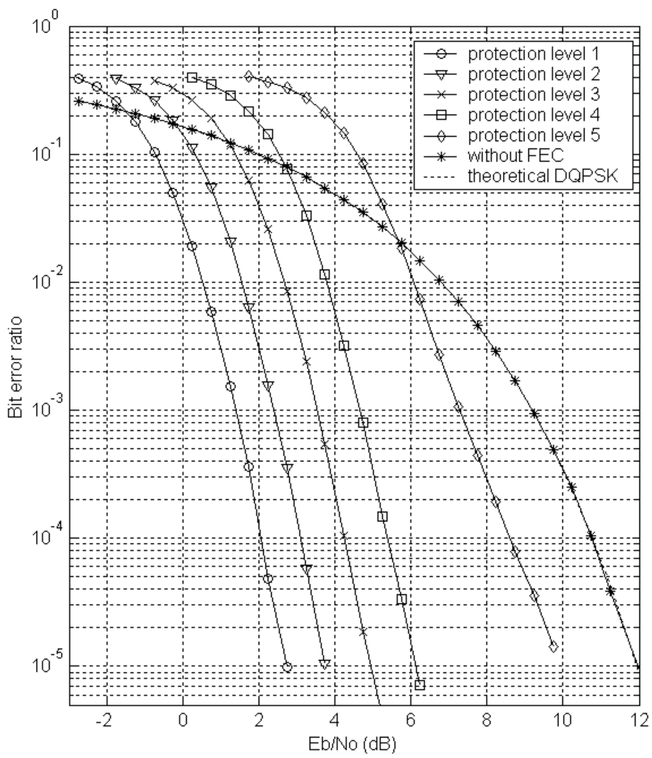

Fig. 6. Bit error ratio vs. Eb/No in case of wideband Gaussian noise and for different error correction schemes available in DAB standard.

If the interference bandwidth is decreased and if the interferer power is equal to the signal power, a number of sub-carriers of the DAB OFDM signal are corrupted. Fig. 7 shows the BER as a function of the noise bandwidth, in which the sub-carrier information has been corrupted. The affected sub-carrier data has been replaced with random binary data for this experiment resulting in a sub-carrier BER of 0.5. It can be seen that for a broadcast with protection level 3 up to 70 kHz can be lost without significant deterioration, which corresponds to 4.5% of the overall bandwidth.

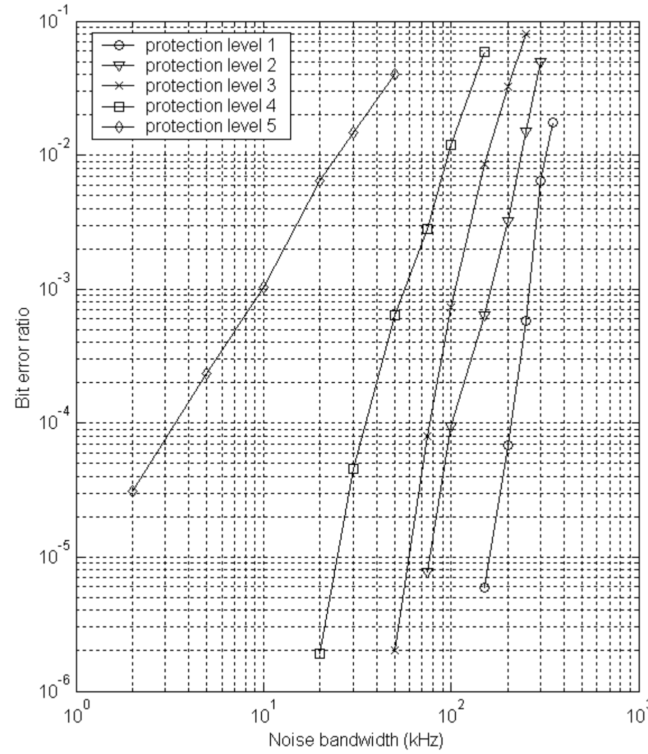

Fig. 7. Bit error ratio vs. Eb/No in case of narrowband interference and for different error correction schemes available in DAB standard.

Error distribution in a fading channel is illustrated in Fig. 8 (a), in which the exact location of bit errors has been analyzed during a simulation run. The abscissa shows the sub-carrier number representing the frequency at which the errors occurred. It can be seen that the bit errors are highly correlated and occur in bursts. If the bit errors are plotted after performing time and frequency interleaving, then they are evenly distributed throughout the MPEG frame as shown in Fig. 8 (b). The convolutional decoder works well with this error distribution, depending on the protection level applied.

Fig. 9 shows BER with and without frequency and time interleaving at protection level 3. It can be seen that

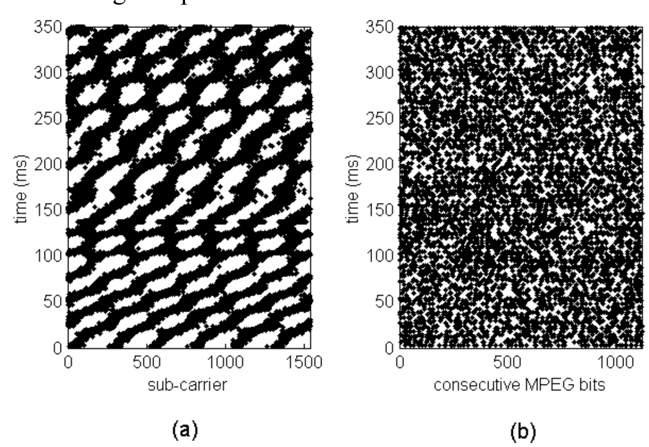

Fig. 8. Bit errors due to deep fades (a) are evenly distributed by interleavers (b).

interleaving is essential for the forward error correction to work properly.

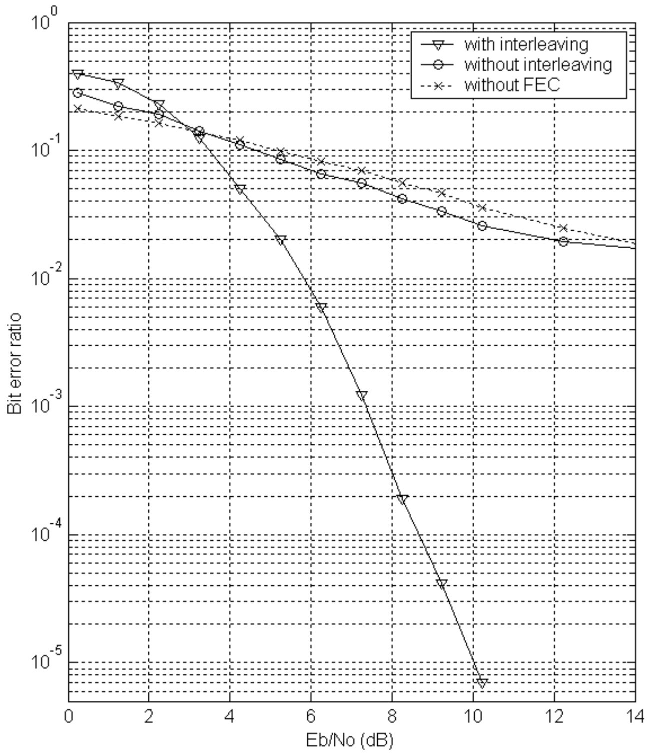

Fig. 9. Bit error performance with and without interleaving in a fading channel.

C. Case studies

Two case studies have been performed in order to obtain results relevant to more realistic scenarios. Both a dense urban topology and rural topology have been simulated. The urban topology represents a typical local-area broadcasting system in the center of a large city and consists of 17 reflected signal paths with no direct path because of possible tall building obstruction (Rayleigh channel). The receiver is moving at a speed of 70 km/h and operates in the L-Band (mode II). The rural topology represents a typical large area coverage single-frequency network in the VHF band, which is used to receive DAB programmes between cities on motorways for instance. It consists of three direct signals (10 kW each) and one reflected path, e.g. from a mountain (Rice channel). Simulation results are plotted in Fig. 10. The three cases *urban*, *rural* and *AWGN* (pure Gaussian noise taken from Fig. 6) were compared for protection level 3. It can be seen that multi-path reception degrades pure AWGN system performance by around 2 dB in case of the rural and by up to 8 dB in case of the urban case. To achieve an acceptable error performance of 10^{-4}, the signal-to-noise ratio in the urban case has to be 6 dB (factor 4 in power) higher than in the rural case. By changing the protection level to 2 in the urban case, error performance is substantially improved. However, the penalty is a decrease in throughput of useful bits by around 20%, caused by the additional redundancy transmitted over the channel to achieve higher coding gain.

For an audio sequence length of 8 s, the computation times for the simulation are 62 s (urban topology) and 41 s (rural topology) respectively on a 2 GHz notebook PC.

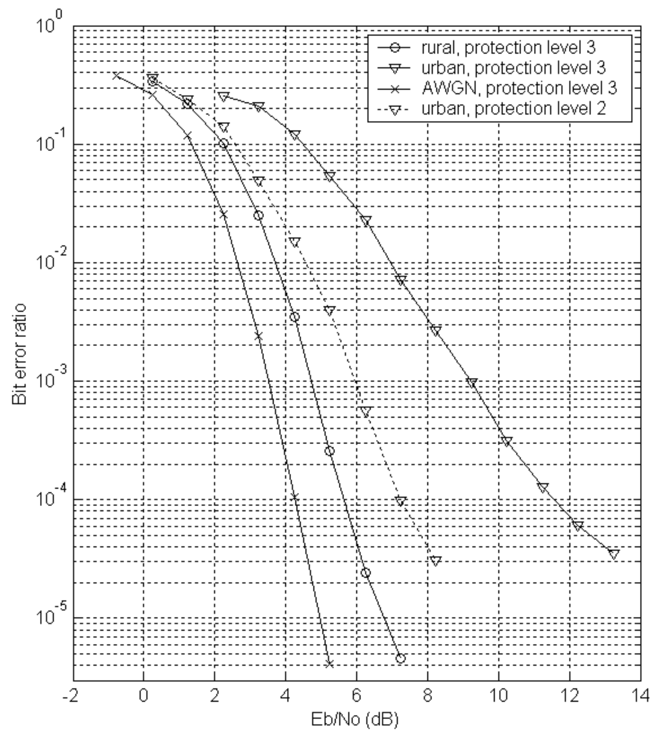

Fig. 10. Bit error ratio versus Eb/No for different channels.

IV. Conclusion

A software simulation model was developed which covers the major aspects of a real DAB transmission system. The performance of DAB modulation schemes can be predicted under specific artificial test conditions in Matlab/Simulink. The graphical user interface enables the user to adjust parameters rapidly and to obtain a quantitative feel as to how transmission quality is affected if these parameters are adjusted. Finding the relation between bit error ratio and subjective audio quality at an early stage leads to efficient performance estimation since extensive listening tests are avoided. This holds under a variety of impairments due to the fact that time and frequency interleavers spread errors out in frequency and time. Simulation in the complex baseband domain is well suited to predict the performance of modulation schemes including different kinds of channels. Frame-based processing should be used to model multi-rate systems such as DAB.

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