

Special Issue on MC-SS

Validation of a HF spread spectrum multi-carrier technology through real-link measurements

Héctor Santana-Sosa^{1*}, Santiago Zazo-Bello², Iván A. Pérez-Álvarez¹, Ivana Raos²,
Eduardo Mendieta-Otero¹ and Javier López-Pérez¹

¹Departamento de Señales y Comunicaciones, Universidad de Las Palmas de Gran Canaria, Campus de Tafira s/n, Las Palmas de Gran Canaria, Spain
²Escuela Técnica Superior Ingenieros de Telecomunicación, Universidad Politécnica de Madrid, Madrid, Spain

SUMMARY

The HF band data communication link has been traditionally desired by many of the large range transmission systems although it is associated to unfavourable performances as low transmission rate, large delay and low confidence in terms of link establishment and maintenance. Although transmission rates may be high enough to transmit digital voice, delay, usually over several seconds, has been the main handicap to let the systems provide interactive digital voice links. Indeed, there is no unclassified equipment with this capability. The main achievement of this proposal is that we are able to guarantee digital voice transmission with low latency, around 135 ms (modem + codec), providing a full interactive digital voice link. Performances of two new 2460 bps HF modems are presented versus the 39-tone 2400 bps MIL-STD-188-110A modem, working over an ITU-R moderate channel. Furthermore, these results are corroborated by real tests carried out in a 1800 Km link in the 18 MHz band. Copyright © 2006 AEIT.

1. INTRODUCTION

1.1. The HF channel

In the HF band (3–30 MHz), long distance communications are feasible, thanks to the use of the ionosphere as passive reflector. However, the atmospheric nature of this reflector makes the systems face a very hard communication environment. Multipath effects are always present and have to be considered in depth, besides the very fast time-varying channel characteristics.

The main parameters used to measure channel behaviour are *frequency coherence* (Δf_c) and *time coherence* (Δt_c) [1]. The first one (Δf_c) gives information about how narrow must one modulated carrier be in order to consider the channel flat. Values in HF channel are usually below 1 KHz. The second parameter (Δt_c) establishes the time separation between two pulses with different attenuations.

Its inverse is known as *doppler spread* (f_d) and in HF, it is usually from 0.1 to 2 Hz. This multipath environment makes efficiencies over 0.5 bit/Hz very difficult to achieve. The ITU-R has defined several parameter combinations in order to establish a base of representative HF channel conditions [2]. Figure 1 shows the channel baseband time-frequency response estimation of a 'Mid-Latitudes Disturbed Conditions' HF link as a graphical example of the hard condition a HF modem has to cope with. Although this channel scenario sets $\Delta t_c = 2$ ms and $\Delta f_d = 1$ Hz these two values could be increased up to 7 ms and 10 Hz in severe channel conditions. Usually, the bandwidth of HF transmissions are below 3 KHz [3] and the best performances in data communications are obtained by the combination of powerful codes and very long interleavers, which introduce an important delay in the communication and make interactive digital voice link impossible.

* Correspondence to: Héctor Santana-Sosa, Departamento de Señales y Comunicaciones, Universidad de Las Palmas de Gran Canaria Campus de Tafira s/n, 35017, Las Palmas de Gran Canaria, Spain. E-mail: hector@gic.dsc.ulpgc.es

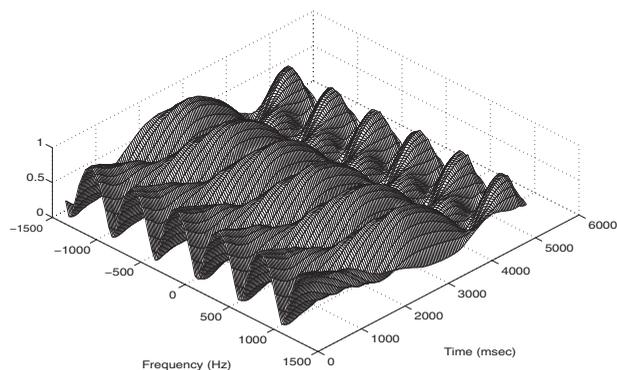


Figure 1. 'Mid-latitudes disturbed conditions' HF channel.

1.2. State of the art

Single carrier schemes have been typically used for facing the hard conditions of short-wave radio communications. Indeed, most of the systems designed for data transmission in the HF band use single carrier techniques beside powerful coding [3]. Optimum coding performance needs the use of long interleaving matrices in order to cope with the long burst errors introduced by the HF channel. Interactive digital voice communications are not feasible with this kind of systems due to interleaving delays that usually rises above several seconds. Most of present data modem applications are based on the standard MIL-STD-188-110A [4] as a military data transmission system, or in the standard STANAG 4285 [5].

The modem described in MIL-STD-188-110A standard specifies data rates range between 75 and 2400 bps. Transmission process uses an 8-states convolutional encoder with 8PSK modulation scheme with variable interleavers. This standard also describes, in two appendices, two multi-carrier schemes: one of them is a non-orthogonal 16 tones DPSK modem where channel estimation is not required, but also spectral efficiency is reduced. The other operating mode is an orthogonal (OFDM) 39-tones including a (14,10) Reed-Solomon encoder with variable interleaving depth which is selected by the user. This modem has to be considered as one of the first HF-band military systems published in the open literature and thus, it is one reference for this kind of systems although it does not provide support for an interactive digital voice link due to the aforementioned interleavers delay.

1.3. Multi-carrier modulations

One of the main problems to consider when a HF modem is being designed is related to the long impulse response length of the channel. The first strategy to deal with this

problem is to reduce transmission rate reducing multipath distortion to a small symbol fraction. Required data transmission rate might be obtained by means of Multi-Carrier techniques, and in particular with *Orthogonal Frequency Division Multiplex* (OFDM), thus giving maximum spectral efficiency.

The performances of OFDM schemes in HF channels have been deeply analysed by C. Cook [6], also published by E.E. Johnson [3], considering a large set of parameters as interleaving length, robustness against impulsive noise, behaviour in front of co-channel interference, relationship between average and peak power, equipment specifications, synchronisation issues and spatial diversity techniques applicability. Cook's studies stated that OFDM-based modems are more efficient than single carrier systems when long interleaving and powerful coding were reduced. This conclusion was our initial motivation for using multi-carrier techniques in HF communications in order to fulfill our main goal: design, implementation and test of a HF physical layer capable to provide interactive HF voice communication with lower latencies and better performance than the already mentioned standards.

2. DATA MODEM DESIGN

The modems presented in this paper follow the idea initially explained in Reference [7]. That paper shown, in a basic simulation scenario, the suitability of multi-carrier spread spectrum techniques in HF communications where heavy channel coding and long interleavings should be avoided.

Further researches with complete channel simulators demonstrated that SS-MC-MA techniques provide a robust mechanism to avoid the effects of deep-nulls introduced by the channel (Figure 1) without the need for powerful coding and long interleaving. SS-MC-MA provides frequency diversity keeping the bit rate constant, however it suffers from self-spreaded symbols interference. Results, shown in References [8] and [9] proved the viability of HF interactive digital voice links with low signal-to-noise ratios by means of in-band spreading, reduced channel coding and no interleaving. The absence of time diversity is supplied by SS-MC-MA, where the serial symbols are transmitted over the whole band so the information affected by deep-nulls can be restored by the unaffected carriers. Those papers also pointed out that plain OFDM does not outperform spread-spectrum techniques for the affordable interleaving depths and channel coding; the low data rate achieved and the low latency constrain make impossible

the use of powerful channel coding and longer interleavers, respectively.

HF channel effect generates an orthogonality loss between in-band self-spread symbols, thus providing Multiple Access Interference (MAI). Although channel estimation mechanism demonstrated to be accurate enough through the carried out tests (Figure 1 is taken from the actual channel estimation process), high levels of interferences and noise from human or natural sources introduce estimation errors that reduce the ideal performances of the Multi-User Detection (MUD) process.

Since the presentation of References [8,9], two different decisions have been taken in order to cope better with MAI. On one hand, two new different MUD techniques have been applied to the design. On the other hand, the number of self-spread transmitted symbols have been reduced, so MAI has been also reduced and the aforementioned channel coding has been completely eliminated. It is important to remark the different self spread symbols represent different ‘virtual’ users.

Two schematic block diagrams are presented in Figures 2 and 3. They represent transmitter and receiver, respectively and as it can be noticed, channel coding has been removed. An S&C scheme [10] has been selected for time/frequency synchronisation [9]. The two new modems presented in this paper follow the same transmission scheme but they apply two different MUD techniques that are described in the sequel.

2.1. Two-stages MUD implementation

In order to improve previous performances by introducing more sophisticated algorithms with also higher computa-

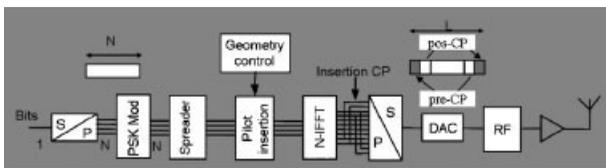


Figure 2. Block diagram of the transmitter.

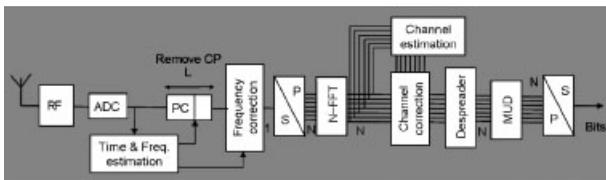


Figure 3. Block diagram of the receiver.

tional requirements but maintaining the low delay constraint, we have applied global minimum mean square error (GMMSE) detectors [11]. If simplified simulation scenarios are applied, this technique clearly outperforms those proposed in our previous contributions. However, noise power should be estimated and also large matrices have to be inverted.

The incoming signal at receiver side can be represented in the frequency domain (assuming perfect time and frequency synchronisation) as follows

$$y = HCx + n \tag{1}$$

where C represents the spreading matrix, H is the channel matrix as a diagonal matrix whose elements are the corresponding frequency response at carrier frequency. Vectors x and n represent respectively the transmitted symbols and the additive white Gaussian noise (AWGN). The two stages implementation is shown in Figure 4. The first stage implements a *Parallel Interference Cancellation (PIC)* scheme where $Q()$ is the standard symbol detector by minimum distance criteria and M represents the interference regeneration process

$$M = (C^H G^{(1)} HC - \text{diag}(C^H G^{(1)} HC)) \tag{2}$$

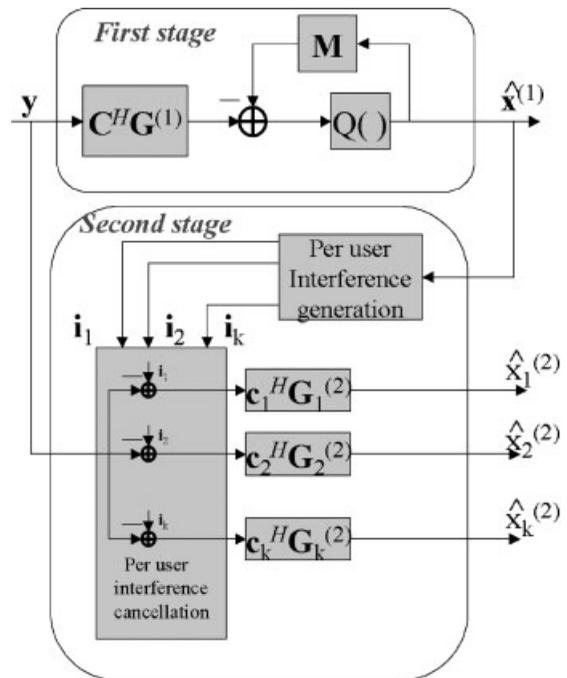


Figure 4. Interference cancellation process.

The $\text{diag}()$ operator means a diagonal matrix whose entries are the corresponding elements of the processed matrix and $\mathbf{G}^{(1)}$ is a matrix calculated following the GMMSE criteria

$$\mathbf{G}^{(1)} = \mathbf{H}^H (\mathbf{H}\mathbf{C}\mathbf{C}^H \mathbf{H}^H + \sigma^2 \mathbf{I})^{-1} \quad (3)$$

where superscript (1) refers to the first stage, σ^2 is the noise power and \mathbf{I} stands for the identity matrix.

The use of GMMSE with PIC techniques should reduce MAI thus providing a nearly one-user set of scenarios if proper interference is cancelled. Such a case, per user MMSE [12] will improve the performance of first stage, so it is implemented in a second stage (Figure 4). Interference generator simply provides the corresponding interference affecting to a particular user

$$i_k = \mathbf{H}\mathbf{C}\mathbf{I}_k \hat{x}^{(1)} \quad (4)$$

\mathbf{I}_k is a diagonal matrix identity-like but with a zero in position (k,k). Eventually, a set of single user (SU) equalisers is provided following MMSE criteria

$$\mathbf{G}_k^{(2)} = \mathbf{H}^H (\mathbf{H}\mathbf{c}_k \mathbf{c}_k^H \mathbf{H}^H + \sigma^2 \mathbf{I})^{-1} \quad (5)$$

where \mathbf{c}_k is the corresponding user code.

2.2. One-stage MUD implementation

In previous versions of the development, PIC with simple *Equal Gain Combining* techniques were used with quite satisfactory performance. This scheme is very simple because no matrix inversion neither noise power estimation are required so the computational complexity of the implementation is decreased. For this case, only the first stage of Figure 4 is applied and \mathbf{G} is calculated as follows

$$\mathbf{G} = \mathbf{H}^H |\mathbf{H}|^{-1} \quad (6)$$

3. IMPLEMENTATION AND HF LINK ISSUES

The whole system has been implemented in ANSI C language with *Basic Linear Algebra Subprograms* (BLAS) [13] libraries for the algorithms that make use of heavy computational operations with vectors and matrices. Nowadays the system runs in real-time over legacy *Intel x86* (PC) platforms running *Linux* operating systems: both configurations running together on the same *Pentium IV@2.4GHz* need only 10 percent of CPU. Thanks to BLAS libraries availability for several DSP platforms, it should be feasible to port the whole system to an embedded platform.

The system has been designed [9] to be spectrally compatible with existent *High Frequency Data Link* [14] equipment. HFDDL uses vocal-bandwidths so it has been possible to make use of legacy sound cards plugged to the PC.

The real-link has been established between Canary Islands and Madrid in the 18 MHz band. For the transmissions, we have used an amateur HF radio (*Kenwood TS-130SE*) and an all-mode receiver (*AOR AR-5000*). *Kenwood* filter bandwidth transmitter is not HFDDL compliant so, for test purposes, several carriers were not used in order to reduce transmission bandwidth. The test will be remade and expanded in a very near future using a couple of recently acquired professional Rohde & Schwarz XK2100L transceivers.

Figures 5 and 6 show the scheme of the stations used in the real link tests. The distance between these stations is 1800 Km and at least one ionospheric bounce is needed to establish the communication.

The following table shows the main parameters of the *Full specification* of the designed modems and the *Test Configuration* parameters used during the real-link tests. It is important to remark that the test parameters were selected keeping the spectral efficiency constant in order to be able to extract reliable conclusions:

	Full specification	Test configuration
Total carriers	73	55
Pilot carriers	13	10
Data symbols per OFDM symbol	41	30
Sample rate (sps)	9600	9600
FFT length	256	256
Modulation	QPSK	QPSK
Bit rate (bps)	2460	1800
Bandwidth (Hz)	2737.5	2062.5
Samples per OFDM symbol	320	320

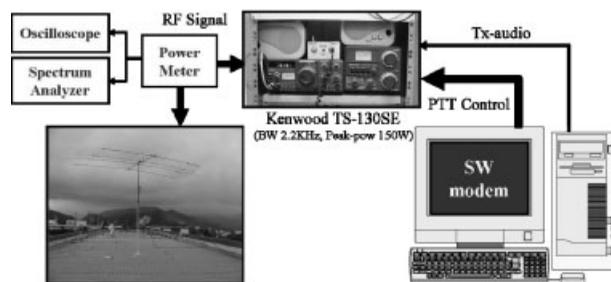


Figure 5. Tx station. Las palmas de gran canaria.

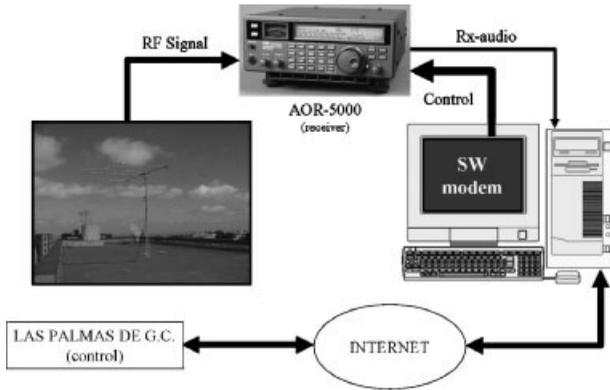


Figure 6. Rx station. Madrid.

4. RESULTS ANALYSIS

Simulated and 5 h real-link performances for the two new modems are shown in Figure 7, and they are compared to MIL-STD 39 tones modem [3,4,6] efficiency. Simulated results were obtained with a Moderate Channel Watterson HF Channel (2 rays, time spread 1 ms, Doppler spread 0.5 Hz) [2]. As it was indicated in the *Full specification* parameters table, the new modem operate at 2460 bps and the 39-tone MIL-STD-188-110A operates at 2400 bps. It can be observed that the two new modems overcome the MIL-STD performance by several decibels. It is very important to recall that these results are obtained without any kind of interleaving matrix, so the data delay

is kept extremely low, 90 ms if only the modem is considered, and 135 ms if vocoder is included. Considering that the 2400 bps MELP vocoder [15] will operate satisfactorily with error probabilities below 10^{-2} , it is concluded that, for a moderate ITU-R channel, modem operability for SNR in 4.8 KHz above 10 dB is guaranteed.

It is noticed that the simpler EGC outperforms the more complex GMMSE. This result is due to the fact that noise estimation is carried out in the whole bandwidth during communication training stage, and there is no tracking of this parameter as it is presumed in Equation (3). The error of this estimation along with the error estimate of the channel are responsible for this unexpected result related to the present implementation of the modem. This aspect is currently under deeper analysis.

In Figure 7, performance in the real link is also presented. This link was established between Canary Islands and Madrid in the 18 MHz band. It can be noticed the parallelism between the simulated results and the real link performances. The difference between simulated and real-link behaviours have its explanation in the difference between the real channel and the moderate ITU-R one (apparently, during the test, the real channel was better than the moderate ITU-R model). Nowadays, the modem does not classify the channels, so deeper studies and test are considered in order to obtain actual channel conditions.

Figure 8 presents the spectrum of a real signal 3 s length, which was received during the test. It can be noticed the effect of channel multipath with a null around 1700 Hz and the presence of a interference just in the middle of transmission band.

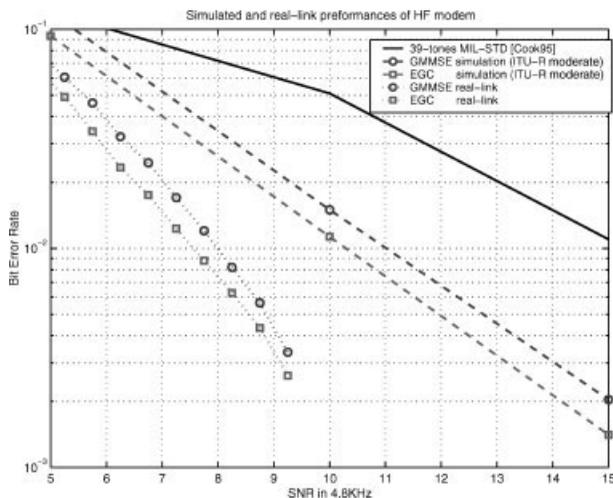


Figure 7. Performance of the two new HF modems vs.39-tone MIL-STD-188-110A.

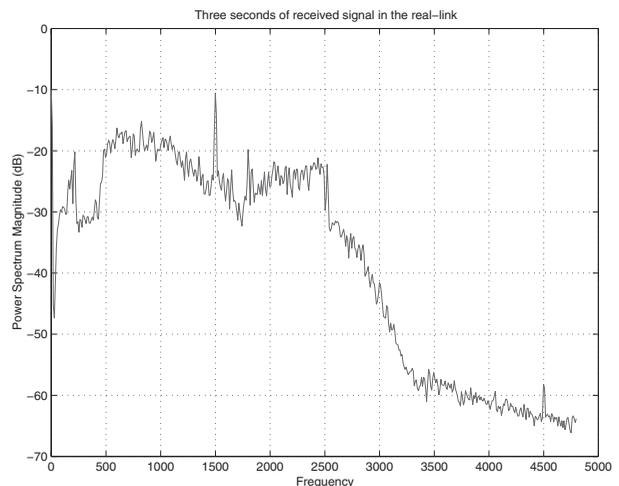


Figure 8. Spectrum of a signal received in the real-link test.

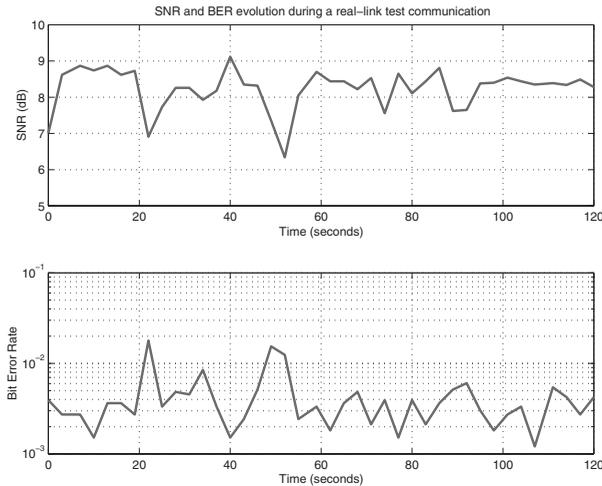


Figure 9. SNR and BER evolution in a 2 min communication.

Finally, Figure 9 presents SNR and BER evolution during a 2 min digital voice communication. Only during short time periods, BER is higher than 10^{-2} . That means that nearly all the communication is free of error from the vocoder point of view, since for lower BER, vocoder behaviour is the same than an error-free communication. It is possible to download the received audio samples from the HFDVL Project web site [16].

Clearly under actual conditions where transmission took place, it would have been possible to establish an interactive digital voice communication with SNR above 8 dB.

5. CONCLUSIONS AND FUTURE RESEARCH LINES

Present work has shown the real feasibility of interactive digital voice transmission over the HF channel. This accomplishes a significant achievement in comparison with conventional analogue transmission with a significant subjective improvement in terms of quality and intelligibility. Key issues are the following: the use of OFDM modulation and the use of MC-CDMA techniques inspired by mobile applications increase system robustness in front of deep spectral nulls and homogenise performance over different subcarriers.

Furthermore, once the transmission scheme has demonstrated to be efficient, complementary analysis, relaxing delay constraint for data transmission [8], is a promising

path to achieve new goals. This operation mode will increase data rate, by the increase of the constellation size and the use of combined interleaving and channel coding strategies. Updated information of this work and audio samples of the link can be downloaded from the HFDVL Project web site [16].

ACKNOWLEDGEMENTS

The work presented in this contribution was supported by the Spanish National R&D Project TEC2004-06915-C03-01/02, AENA (Spanish Airports and Air Navigation) Project 240/033/0051 and by the Canary Islands Local Government R&D Project IDT-LP-04/017. It has been partially supported by the Spanish National R&D Projects TEC2005-08377-C03 and TEC2005-07010-C02.

REFERENCES

1. Proakis JG. *Digital Communications* (4th edn). McGraw-Hill: NY, USA, 2000.
2. ITU-R Recommendation F.1487. *Testing of HF Modems with Bandwidths of up to about 12 kHz Using Ionospheric Channel Simulators*. International Telecommunications Union (ITU), 2000.
3. Johnson EE, et al. *Advanced High Frequency Radio Communications*. Artech House: MS, USA, 1997.
4. Department of the Army, Information Systems Engineering Command, *MIL-STD-188-110A: Interoperability and Performance Standards for Data Modems*. PA. Naval Publications and Forms Center, Attn. NPODS, Philadelphia, Sep 1991.
5. STANAG-4285. *Characteristics of 1200/2400/3600 Bits per Second Single Tone Modulators/Demodulators for HF Radio Links*. NATO Standardization Agreements, 1990.
6. Cook SC. *Advances in High Speed HF Radio Modem Design*. Proceedings of Nordic Shortwave Conference (HF95), Faro (Sweden), 1995.
7. Zazo S, Bader F, Páez-Borrillo JM. Spread Spectrum OFDM Modem for HF Voice-Band Link over Fading Channels. *European Transaction on Telecommunications* 2000; **11**(6): 593–598.
8. Raos I, Del Cacho A, Pérez-Álvarez I, Zazo S, Mendieta-Otero E, Santana-Sosa H, Páez-Borrillo JM. *Performance of an OFDM-CDMA HF Modem*. Ninth International Conference on HF Radio Systems and Techniques. Bath, U.K. June 2003.
9. Pérez-Álvarez I, Raos I, Zazo S, Mendieta-Otero E, Santana-Sosa H, Páez-Borrillo JM. *Interactive Digital Voice Over HF*. Ninth International Conference on HF Radio Systems and Techniques. Bath, U.K. June 2003.
10. Schmidl TM, Cox DC. Robust Frequency and Timing Synchronization for OFDM. *IEEE Transactions on Communications* 1997; **45**(12): 1613–1621.
11. Daudais JY, Héland JF, Citerne J. An improved linear MMSE detection technique for multicarrier CDMA systems: comparison and combination with interference cancellation schemes. *European Transaction on Telecommunications* 2000; **11**(6): 547–554.
12. Fazel K, Kaiser S. *Multi-Carrier and Spread Spectrum Systems*. John Wiley and Sons: Chichester, UK, 2003.
13. Blackford LS, Demmel J, Dongarra J, Duff I, Hammarling S, Henry G, Heroux M, Kaufman L, Lumsdaine A, Petitet A, Pozo R, Remington K, Whaley RC. An updated set of basic linear algebra Subpro-

- grams (BLAS). *ACM Transactions on Mathematical Software* 2002; 28(2): 135–151.
14. SARPs for HF DL. *Validation by AMCP-Memo/8*. Montreal, Canada, September 1998.
 15. Draft to Federal Information Processing Standards Publication. *Analog to Digital Conversion of Voice by 2400 bit/second Mixed Excitation Linear Prediction (MELP)* May 1998.
 16. High Frequency Data + Voice Link Project. <http://www.hfdvl.eu>.

AUTHORS' BIOGRAPHIES

Héctor Santana-Sosa received his B.Sc. and M.Sc. from the Universidad de Las Palmas de Gran Canaria (ULPGC) in 1999 and 2003, respectively. He has participated in several R&D projects including one European AST project. Nowadays, he works as a Research Assistant for the Department of Signals and Communications in the ULPGC. He is also working towards his Ph.D. in the fields of ionospheric communications and multi-carrier techniques.

Santiago Zazo-Bello is Telecom Engineer and Dr. Engineer by the Universidad Politécnica de Madrid (UPM) in 1990 and 1995, respectively. From 1991 to 1994, he was with the Universidad de Valladolid and from 1995 to 1997 with the Universidad Alfonso X El Sabio at Madrid. In 1998, he joined UPM as Associate Professor in Signal Theory and Communications. His main research activities are in the field of Signal Processing with applications to Audio, Communications and Radar. Since 1990, he has participated in 12 projects with the Spanish industry and 1 European project (ESPRIT) and 3 IST projects. He is also co-author of 9 international papers and more than 60 conference papers.

Iván A. Pérez-Álvarez is Telecom engineer and Doctor Engineer by the Universidad Politécnica de Madrid (UPM, Spain) in 1990 and 2000, respectively. From September 1989 to October 1990, he was with Europea de Comunicaciones S.A. (Madrid, Spain) as a project engineer. From December 1991 to January 1997, he was with Telefonica Sistemas S.A. as project engineer and project manager since 1995. In both companies, he was involved with the development of data communications systems, radio communications systems, digital signal processing algorithms and signal classification systems for the Spanish Department of Defense, between others. He joined the Universidad de Las Palmas de Gran Canaria (ULPGC, Spain) as faculty staff in April 1998. He presently belongs to the staff of Departamento de Señales y Comunicaciones (DSC) as an Associate Professor in the Escuela Técnica Superior de Ingenieros de Telecomunicación (ETSIT). His main research activities are in the area of signal processing with special emphasis in the fields of radio communications.

Ivana Raos has received her Dipl. Ing. and Ph.D. Telecommunications degrees in 1999 and 2006 from Belgrade University, Serbia and Universidad Politécnica de Madrid, Spain, respectively. Her research interests span a wide range of applications of signal processing in communication systems such as multiple antennas systems, multicarrier systems, multiuser interference problem, coordination in cellular systems. Since 1999, she has participated in 2 European IST projects and she has co-authored 2 international papers and 15 international conference papers.

Eduardo Mendieta-Otero is Telecom Technical engineer in Electronic Equipment and in Radiocommunication by the Universidad de Las Palmas de Gran Canaria (ULPGC, Spain) in 1992 and 1994, respectively, and is Telecom engineer by the Universidad de Las Palmas de Gran Canaria (ULPGC, Spain) in 2001. He joined ULPGC as faculty staff in January 1994. He presently belongs to the staff of Departamento de Señales y Comunicaciones (DSC) as an Associate Professor in the Escuela Universitaria de Ingeniería Técnica de Telecomunicación (EUITT). His main research activities are in the area of signal processing with special emphasis in the fields of radio communications. He is working towards his Ph.D. Degree in the fields of ionospheric communications and DSP Real Time Signal Processing.

Javier López-Pérez received his M.S. Degree in Telecommunication Engineering from the Universidad de Las Palmas de Gran Canaria, Spain, in 2004. He works as a Research Assistant for the Department of Signals and Communications in the Universidad de Las Palmas de Gran Canaria, where he is working towards his Ph.D. Degree in the fields of MC-CDMA, Software Defined Radio and DSP Real time Signal Processing.