Real-Time Implementation of a COFDM Modem for Data Transmission over HF Channels with the TMS320C31 DSP

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Real-Time Implementation of a COFDM Modem for Data Transmission over HF Channels with the TMS320C31 DSP

Abstract

This application report describes a transmission system with Orthogonal Frequency Division Multiplexing (OFDM) and Quadrature Phase Shift Keying (QPSK) over noisy multipath fading channels. When an encoder and a coherent receiver based on the maximum-likelihood estimation is added, we obtain the modulator-demodulator (COFDM MODEM) presented in this paper. The required performance was obtained thanks to specific research into channel estimation for the coherent demodulator and on frequency-shift correction.

A digital implementation of the MODEM is built around a Texas Instruments (TI™) TMS320C31 Digital Signal Processor (DSP) and efficient special-purpose hardware for the encoder-decoder. Simulation results for this modem are given.

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Introduction

Communication over the ionospheric channel is typically available by multipath propagation, which leads to a spread in arrival times. The channel, with varying characteristics, cannot be considered as stationary. Moreover, the signal transmitted to the receiver will also be subject to atmospheric noise and more importantly, to interference from other HF transmissions. Thus, a Doppler shift, a frequency spread, and shadow fading can be associated with each path. In severe cases, the channel produces rapid random amplitude and phase variations in the signal received when the data stream is transmitted over the ionospheric channel. Given the harsh ionospheric environment and the scarcity of the available spectrum, a data transmission system was chosen for an ionospheric channel with a nominal bit rate of 2400 bits per second and a standardized bandwidth of 2,3 kHz.

The recent advances of Digital Signal Processing (DSP) and Very Large Scale Integration (VLSI) circuit technologies have provided a parallel approach. The initial problems of Orthogonal Frequency Division Multiplexing (OFDM), such as passive complex computations and high-speed memories, no longer exist. Moreover, the use of a discrete Fourier transforms eliminates the banks of sub-carrier oscillators and coherent demodulators required by parallel data transmission systems.

In our MODEM, OFDM, and especially COFDM is exploited to minimize degradation caused by the ionospheric channel. This modem includes a frequency-offset estimation and a channel-transfer-function estimation, and allows coherent demodulation with decoder using a maximum-likelihood criterion.

In *Basic Principles of the Modem*, an overview of the COFDM system and its development is presented.

In *Development Tools*, we describe tools that are used to test algorithms. The test environment is also presented. In addition, simulation results of the MODEM under ionospheric channel conditions are shown.

Basic Principles of the Modem

An efficient use of bandwidth can be obtained with an OFDM[1] process because the spectra of the individual sub-channels are allowed to overlap (the more sub-carriers used, the flatter the spectra of the signal). A simple relation associates the symbol time *Tu,* including the information to be transformed, and the space between sub-channels ΔF:

$$Tu=1/\Delta F$$

Tu must be greater than the scattering of the impulse-response of the channel.

The number of sub-carriers is in the range of one thousand to accommodate the data rate and the value of *Tu.*

Figure 1 and Figure 2 illustrate the process of the FFT-based COFDM system.

Channel Coding and Modulation

To improve transmission performances, channel coding and pseudo-random interleaving are used. Thus, transmission errors on each sub-channel can be considered as being independent. The incoming serial data is first coded and interleaved. The transmission system described herein allows two types of encoder to be used:

- a convolutional encoder with constraint length 6 and a rate R=l/2 or R=314 (generator polynomia 133-171 in octal);
- a turbo-encoder[2,3] with a rate R=1/2;

The modem implemention has a nominal bit rate of 2400 bit/s with R=3/4 and of 1600 bit/s with R=112.

The constellation of each sub-carrier corresponds to a QPSK modulation. There is a great need to work with a constellation of the constant amplitude signals because the fading caused by the ionospheric channel can be great. The encoded and interleaved data is converted from serial to parallel and grouped into two bits each to form a complex number (modulation signal in the frequency domain). The inverse discrete Fourier transform applied to the complex numbers generates samples of the signal in the time domain. When the number of sub-carriers is a power of two, the complexity of the operation can be greatly reduced by using the efficient Fast Fourier Transform (FFT) algorithm implemented on the DSP.

Figure 1. Transmitter Design

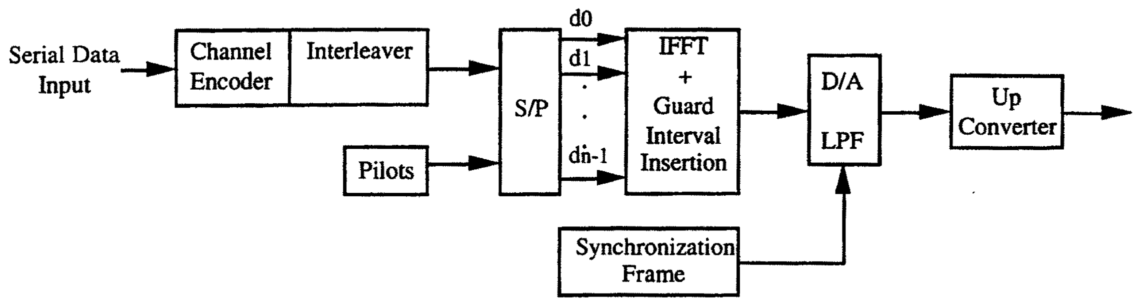
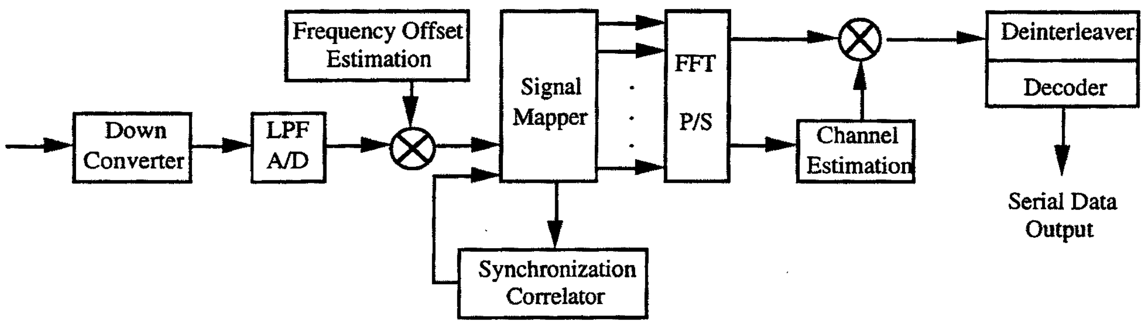


Figure 2. Receiver Design



Channel Characteristics

Coherent detection, by definition, requires a phase reference. However, gain correction is also needed in a COFDM system in a fading environment to take into account intra-symbol, intra-carrier interference. The use of pilots is an efficient method for estimating channel characteristics.[4] Pilot-based correction provides an amplitude and phase reference which can be used to counteract the unwanted effects of multipath propagation. X modulated sub-carriers among N are used as pilots. At the receiver, the transfer function of the transmission channel at sub-carrier frequencies can he obtained by FFT-based interpolation. In the MODEM described, the pilots are sine waves. The number of useful pilots depends on the time spread of the channel, *Tc*:

X>NTc / Tu

Note that using pilots reduces the bit rate.

Figure 3 shows the module of two channel transfer functions: one at the beginning of the transmission (solid line) and the other five ms later (dashed line). This channel is made up of two paths that have relative magnitudes: 1 and 0.5 (fading of 6 dB). The relative frequency shift between both paths is 0.5Hz.

Figure 4 presents the phase evolution with the same channel.

(Note that the sub-carrier number is represented on the abscissa axis).

Figure 3. Channel Characteristics: Transfer Function of the Channel (in dB)

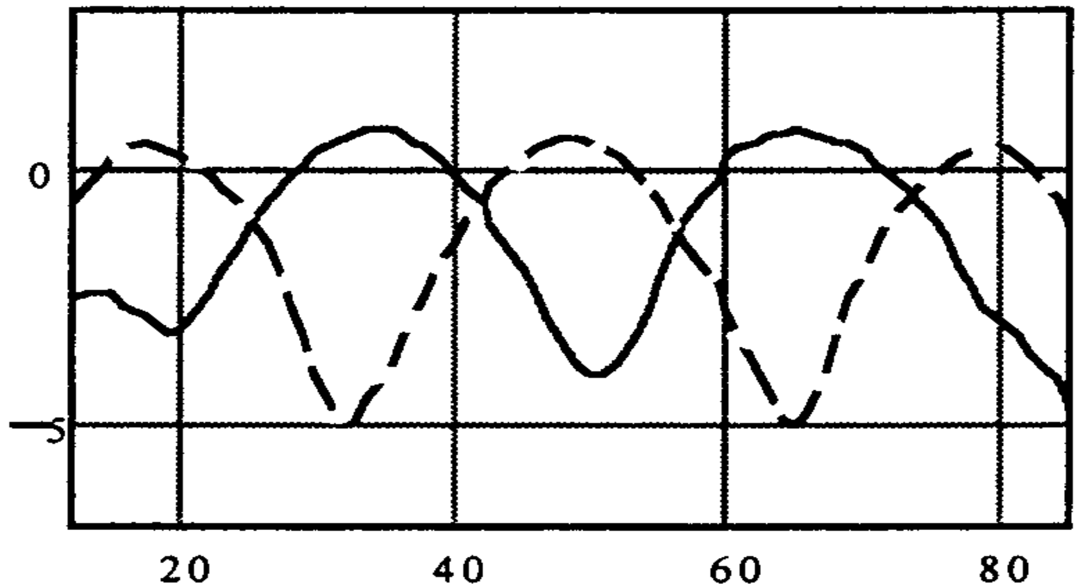
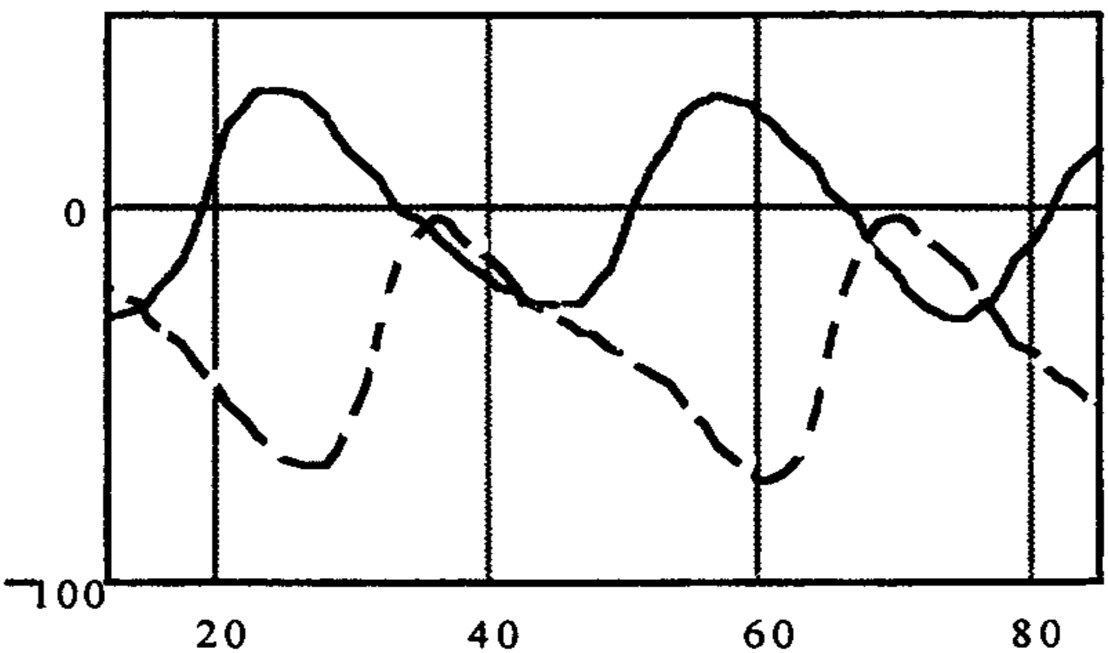


Figure 4. Channel Characteristics: Dephasage Introduced by the Channel (in)



Guard Interval

As described in Weinstein et al. and Ebert et al.[5,6], Intersymbols inter/intra carrier(s) interference can be eliminated by the simple solution of inserting a guard interval between symbols. Each COFDM symbol is preceded by an extension of the signal. The total symbol duration is $Ts = Tu + Tg$ where Tu is the useful symbol duration and Tg is the guard interval. The length of the guard interval is greater than or equal to the length of the time spread of the channel.

Receiver

The receiver implements the inverse functions of the transmitter. The FFT algorithm makes the demodulation as an array of N matched filters to the signal sent on each sub-carrier. The data decoding is implemented using a maximum-likelihood estimate algorithm. A Viterbi decoder or a turbo-decoder is associated with the DSP. Soft decision decoding allows more importance to be given to the sub-carriers, which are not faded out.

Figure 5 presents the constellation of the received signal for the channel described above. The same constellation, corrected by the transfer function, is shown in Figure 6.

Figure 5. Illustration of Decision Criterion: Constellation of the Different Sub-Channels of the Received Signal

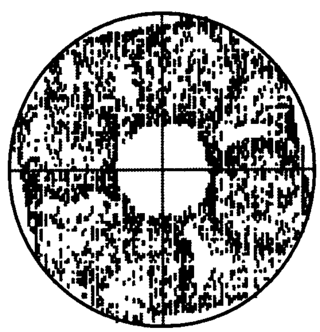
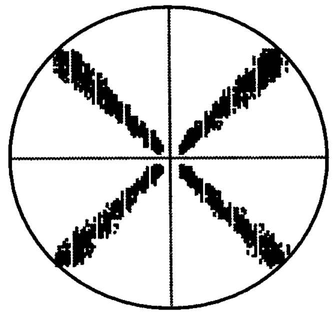


Figure 6. Illustration of Decision Criterion: Constellation Provided on Decoding



Synchronization[7]

Reference symbols (fixed data packets) are inserted for the synchronization frame and pilots (fixed carriers) are used for the frequency correction.

Synchronization Frame

The channel cannot considered as stationary, thus several symbols are grouped together to make up a frame. To be able to define exactly the beginning of useful frames a synchronization frame is inserted between each of them. The synchronization frame is made up of two n-long complementary series[8] S1(i) and S2(i) with i=1...n. Their respective autocorrelative series are defined by:

$$Cj = \sum_{i=1}^{n-j} S1(i)S1(i+j)$$

And

$$Dj = \sum_{i=1}^{n-j} S2(i)S2(i+j)$$

With

$$Cj + Dj = 0 \qquad \text{if } j \neq 0$$

$$C0 + D0 = 2n$$

This auto-correlative property of the complementary series delivers a robust synchronization frame at the receiver.[9] The channel must be considered as stationary for the whole synchronization frame. Moreover, the time interval Δt between two synchronization frames must enable group-delay evolutions to be estimated. The former must depend on the maximum frequency shift Fmax in the channel:

$\Delta t < 1/F\text{max}$

Frequency Correction

An OFDM system is affected by carrier frequency errors. Frequency shifts generate two harmful effects: one is the reduction in signal magnitude at the output of the filters matched to each of the carriers and the second is the cancellation of orthogonality between the sub-channels leading to a degradation in system performance. This degradation increases rapidly with frequency shift and with the number of subcarriers.

The process described by MOOSE[10] makes it possible to estimate the frequency shift. In the system shown, the frequency estimation algorithm uses the pilots of the channel estimation.[11] The maximum limits of the algorithm are +/- 1/2 the inter-carrier spacing. To bring the offset within the limits, a strategy for initial acquisition must be developed. A reference symbol is included at the beginning of the transmission leading to an initial-offset estimation.[12]

Development Tools

The application has been evaluated with an OROS.AU32 board. This board is a PC plug-in card; it carries a 32-bit floating-point TMS320C31 DSP.

OROS-AU32 Board Features

- Analog Interface:
  - Two independent analog inputs and outputs
  - Variable D/A and A/D sampling rate and filtering
  - Amplifier
  - Sample frequency: 48 kHz maximum
- Processor:
  - Texas Instruments TMS320C31
  - Clock frequency: 33 MHz
  - 60 *ns* instruction cycle time
  - 33 MFLOPS (million floating-point operations per second)
- RAM:
  - 1 Mo on the board
- Input, Output:
  - Four TTL digital inputs and two TTL digital outputs
- One Serial Port
- PC/DSP Communications:

Information is transferred from the PC to the board through a 32-bit bi-directional bus: the PC ISA bus is connected to the TMS320C31 32-bit data bus/24-bit address bus.

Algorithms

Modulator programming is used to develop the program code. The algorithms are developed in C language and functions collected from specific libraries such as AU32.LIB or SPOX are included. AU32.LIB is the AU32 board associated library. The SPOX API (Application Programming Interface) includes many of the standard library functions that are not supplied with C compilers for DSP development. The SPOX API also supports over 100 standard math functions that can be used as building blocks for algorithms used in advanced DSP applications. Included among these are: vector functions, matrix functions, filter functions.…

Algorithm Evaluation

Figure 7 and Figure 8 show the transmitter and receiver sections of the MODEM. The PC host is the master and initiates commands to the DSP. The PC sends/receives data to/from the DSP. All the functions are executed by the DSP except the turbo coding or decoding and the Viterbi decoding. The serial port provides a means to transfer data between the DSP and the encoder or decoder.

Figure 7. MODEM Transmitter Section

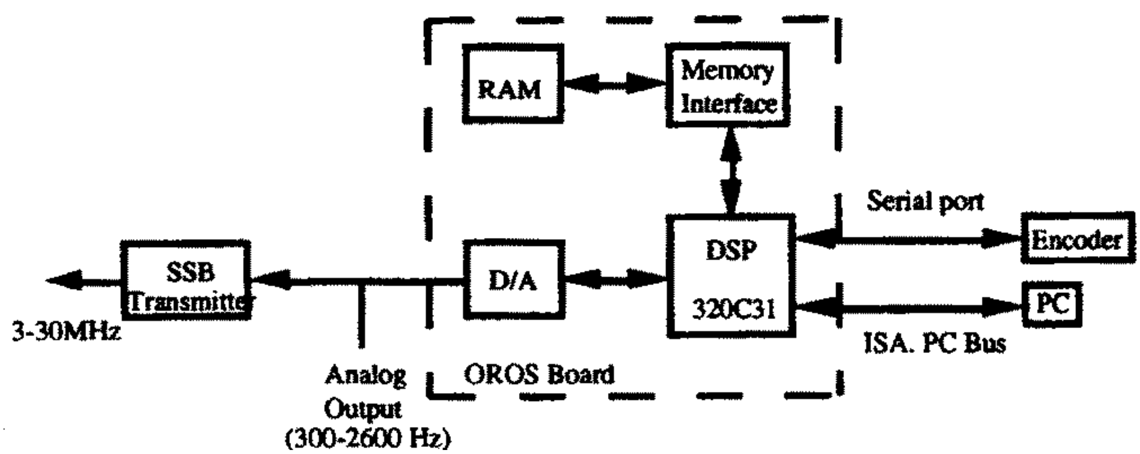
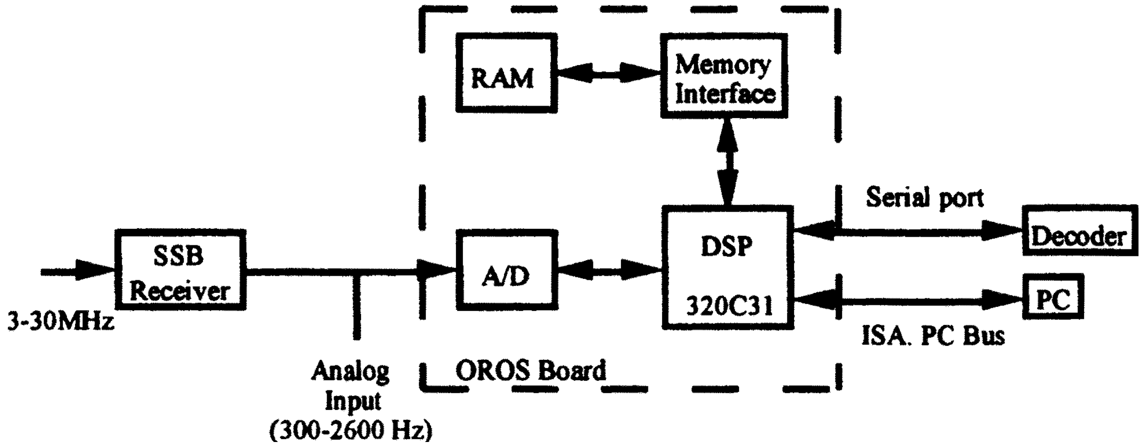


Figure 8. MODEM Receiver Section



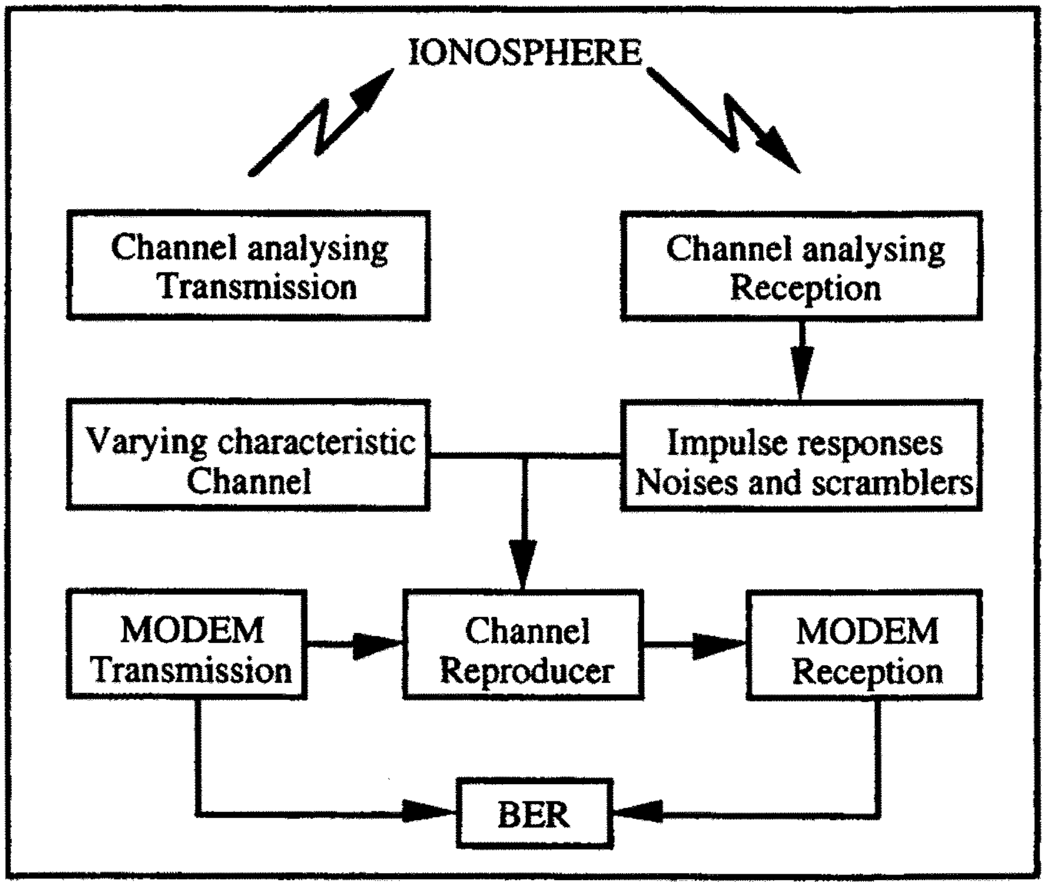
To test the MODEM transmitter, random data is sent from the DSP through the ISA PC bus. Handled data files are sent back to the PC. The DSP output data is converted from digital to analog and the analog output is then used to test the transmission. The data is subsequently up-converted and sent through the ionospheric channel.

The same principle is used to test the MODEM receiver. Data files are sent from the PC to the DSP, processed, and returned to the PC.

Transmission Evaluation

In order to analyse the real performance of the transmitter, the CNET (Centre National d'Etudes des Télécommunications - Lannion France) has realized a HF channel analyzer.[12] This system allows the channel impulse response and different propagation parameters to be measured simultaneously. It can also generate channels with varying characteristics, such as: the number of paths, the offset frequency, and the fading depth. Two methods are used to evaluate the transmission over HF channels. See Figure 9.[10]

Figure 9. Transmission Evaluation over HF Channels



The first method uses varying characteristic channels, the second one uses impulse responses measured on experimental channels. In both methods, a channel reproducer is used to simulate the whole transmission.

Evaluation Criterion

The criterion used in this application is the Binary Error Ratio (BER). It is evaluated with the Monte Carlo method. The number of tests N=400 000 is chosen in order to estimate the error probability around 10^{-3} with a relative precision of 10^{-1}. It is calculated as follows:

$$BER = \frac{1}{N}\sum e_i$$

With

e_i =1: if and error appears on the i*th* bit

e_i =0: if the i*th* bit is decoded correctly

Program Features

The transmitting program requires about 180 Kbytes of program memory, including the DSP/PC interface instructions. It uses from 20% to 30% of the available calculation time. The receiving program requires about 220 Kbytes of program memory, including the DSP/PC interface instructions. It uses from 60% to 70% of the available calculation time.

The sample frequency is equal to 8kHz. The choice of the sample frequency (1/ T*e*) takes into account transmission characteristics, such as: source rate (2400 bit/s), bandwidth (2,3 kHz). Moreover, it has to respect the orthogonality condition:

$$Te = \frac{Tu}{M}$$

Where M is a power of two

Data files for DSP/PC exchanges are stored in an on-board RAM. Different capacities are occupied:

- 8640 x 1 bit
- 1520 x 4 bits
- 28800 x 16 bits

Moreover, 2000 x 32 bits RAM is used for data handling.

Simulations

The following results represent simulations for two channels: one with a single path (example 1*)* and the other with two paths whose relative magnitudes are equal to 1 and 0.5 (example 2). The delay time between both paths is 1 ms. Each channel can be affected with Additive White Gaussian Noise (AWON) and frequency offset.

Frequency Correction

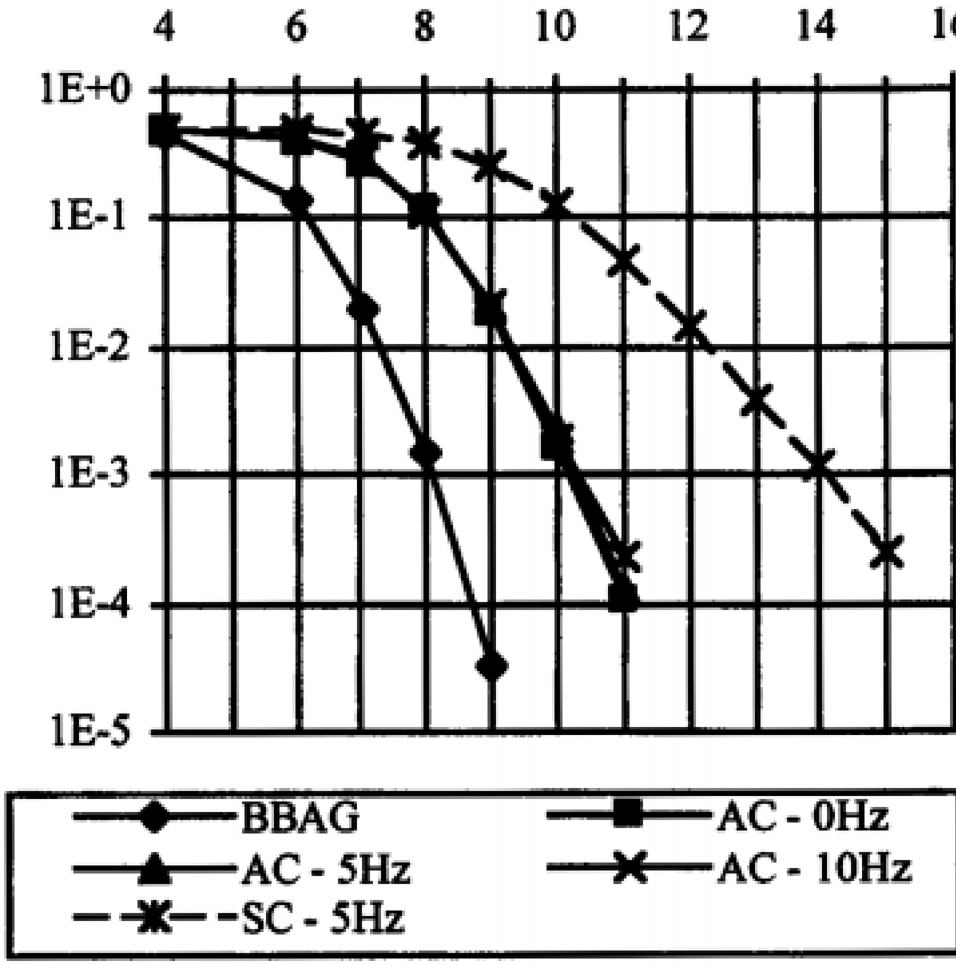
First, the initial offset frequency-acquisition is not included in the system. We examine the convergence time of the frequency-estimation algorithm with several offsets (0, 4 and 8 Hz) of received signal. In all cases, it is smaller than five symbols. The search for the quadratic estimation error shows that it is a function of the Signal-to-Noise Ratio (SNR) at the receiver input whatever the frequency offset. The estimation error becomes independent of the shift after its initialization. The acquisition algorithm must allow the convergence time of the tracking algorithm to be reduced.

Binary Error Ratio

The BER is shown in Figure 10 as a function of SNR for example 2. The BER is independent of the frequency offset when the offset correction is included. In example 1, graphs obtained with frequency offsets are superposed with the AWGN curve. A BER of 10^{-3} is obtained when SNR=8 dB in example 1 and SNR= 10.3 dB in example 2. Selective frequency fadings lead to this difference between the SNRs.

The SNR is quickly improved when the frequency correction is included (4.1 dB for a BER=10^{-3} in example 2)

Figure 10. BER=f(SNR in dB) for a Channel with 2 Paths



BBAG: channel with 1 path and AWGN
AC: with frequency correction
SC: without frequency correction

Selective Fading

Let A1 and A2 be the magnitudes of 2 paths (A1>A2) so the fading depth is:

$FD = -20 \log_{10}[1 - A2/A1]$

Table 1. BER = l0^{-3} Example

FD (db)	SNR (db)
3	8.8
6	10.3
12	13.5
infinite	>18

Conclusions

The program is divided into several modules. This modularity increases the calculation time but it is very useful because it allows easy modifications to the MODEM's characteristics. Moreover, the algorithms are not optimized so improvements can be made. However, some constraints such as the sample frequency, the source rate, or the bandwidth have to be taken into consideration. This MODEM, designed for data transmission, can be adapted to transmit digital speech (2400 bit/sec is a normalized rate for digital speech). Channel coding specific to the speech transmissions must be made in order for this system to be used.

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