# Experimental Performance Results of an Indoor Wireless Extension of IS-136 Based on π/8 D8PSK, Coded Modulation, and Antenna Diversity

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**Abstract - This paper reports the performance results of an experimental study of a digital communications system designed as an extension of the IS-136 North American Digital Cellular System. The proposed system has been designed to achieve twice the effective bit rate (up to 16 kb/s end user data rates) for transmission of high quality coded speech in slow fading indoor wireless environments. This study has covered numerous aspects of the system design including the evaluation of the real time voice communications quality. To create as complete a prototype as practical and to provide means for testing over real and simulated RF channels, the implementation included building a 900 MHz ISM band transceiver.**

**We describe the physical layer design, real time experimental platform implementation and measured performance results of the physical layer on the forward link. Modifications to the IS-136 system include replacing the π/4-QPSK modulation with π/8-D8PSK modulation, changing the channel coding and modulation approach, as well as adding preselection diversity in the handset receiver. Most of the IS-136 slot and frame structure is preserved.**

**The effects of channel and receiver impairments are documented and compared against theoretical and simulation results. Results are reported in the form of frame error rates relevant for speech communications.**

**We show that a frame error rate of about 1 percent is achievable at a 20 dB signal to noise ratio in a 1 Hz Rayleigh fading environment when preselection diversity is active with a 16 kb/s end user data rate.**

## I. INTRODUCTION

The current IS-136 North American Digital Cellular System provides wide-area high mobility wireless access with 8 kb/s end user data rates. Advances in speech coding technology allow the system to provide good speech quality at these low data rates. Recently, we have been investigating extensions to the IS-136 standard[1] that would address the special needs of low mobility users, particularly in an indoor wireless office environment.

Indoor wireless systems face a different set of constraints than the traditional macrocellular systems do. These include:

- user expectations of higher speech quality, comparable to what they get from their wired systems
- expectations for more robust communications
- the additional difficulty of dealing with a slow fading channel, where traditional coding and interleaving techniques may not be as effective.

At the same time, cost, size and power consumption constraints are every bit as important as they are in the wide area cellular system.

We began this effort with a goal of understanding the practicality of mechanisms that would allow one handset/one number ubiquitous communications, both in outdoor macrocellular environments as well as for indoor wireless office system and home environments. Others have studied what it would take to deploy interoperable indoor and outdoor systems[2], but we felt that robustness and speech quality issues with tight cost, size and power constraints had not been fully addressed.

## II. SYSTEM ARCHITECTURE

### *A. Frame Structure*

IS-136 uses a 6 slot, 40 ms frame structure[3] as shown in Figure 1. During each frame, a given user is allocated two evenly spaced downlink slots and two evenly spaced uplink slots. Each slot contains 162 symbols, 130 of which are for end user data, the remainder being for synchronization, signalling and control. Since IS-136 uses π/4-QPSK, this means that every 20 ms, 260 channel bits, or 13 kb/s, are available for user data. Of course, channel coding reduces this number considerably, so the IS-136 speech coder has about 8 kb/s available.

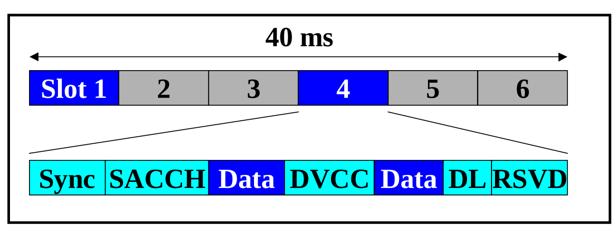


**Figure 1 - IS-136 downlink frame structure**

In the Enhanced IS-136 system, we leave the overhead symbols in the original π/4-DQPSK format. The 130 data symbols are transmitted as π/8-D8PSK, providing 390 channel bits per slot, or an uncoded capacity of 19.5 kb/s. Our initial studies used a 16 kb/s Low Delay-Code Excited Linear Prediction speech coder (G.728), which permitted equal error protection on all the speech coder bits. Experimental results have shown that good performance can be achieved with a rate 5/6 code, providing the 16 kb/s end user data rate.

### *B. General system choices*

Size, power, and cost constraints at the handset are considered to be the most limiting characteristics of the system design. By choosing π/8-D8PSK over π/4-QPSK, and increasing the overall coding rate, we spend several dB of link margin. Some of the loss can be made up with more effective codes[4]. To make up the difference, we investigated various antenna diversity options. For the handset, a low complexity approach that was found to be usable was preselection diversity[5]. For the uplink, where size, power and cost constraints are eased a bit, equal gain combining was investigated. The following sections of the paper present more detail on these choices.

### *C. Hardware Architecture*

The Enhanced IS-136 system was prototyped using commercially available DSP boards based on the TI TMS320C40 DSP[6], a 900 MHz RF transceiver, as well as custom interface circuitry. For ease of implementation and flexibility, major system components were partitioned onto separate DSPs. Functions allocated to different DSPs are:

- Speech coding
- Transmit channel coding, modulation, and filtering
- Receive filtering, synchronization and demodulation
- Receive channel decoding
- Speech decoding

In addition, system control and monitoring (e.g., constellation display and user interface to control system parameters) were placed in a separate DSP with a lesser real-time burden.

This modularity allowed us to easily investigate the performance of different speech coding algorithms and channel coding techniques under the same RF impairments. In particular, we investigated a Low Delay-Code Excited Linear Prediction coder (G.728 at 16 kb/s) as well as the Algebraic Code Excited Linear Prediction coder (US-1 or GSM-EFR at 12.2 kb/s)[7].

Custom interface circuitry provided most of the functions that would be needed in a product implementation of the Enhanced IS-136 system, along with some extra functions needed to demonstrate and measure the performance of our implementation. These functions included:

- Audio and baseband I/Q signal antialiasing and reconstruction filtering
- Transmitter modulation, up conversion, RF filtering, and power amplification
- Receiver RF filtering, amplification, down conversion and demodulation to I/Q signals
- Antenna control, for diversity experiments
- Clock generation and control

Figure 2 below illustrates a block diagram of the downlink portion of our experimental system.

The downlink design dominated the investigation. After the downlink was fully operational, implementing the uplink was relatively straightforward.

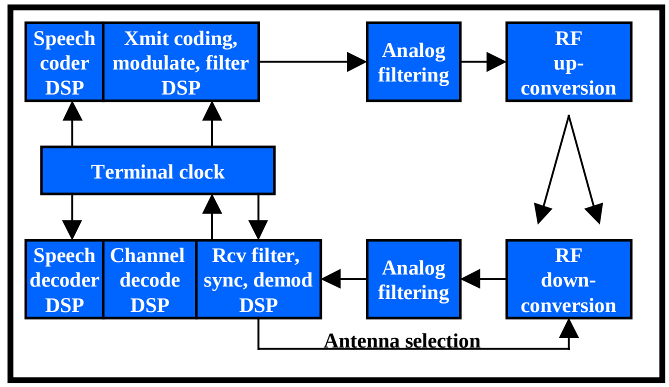


**Figure 2 - Downlink System**

### *D. Software Architecture*

Having made the choice of a signal-processing platform, we looked into the development tools that were available for use on that platform. Experience on a prior project showed that we could expect very good performance from a C compiler optimized for the DSP's instruction set. This had the side benefit that it was much easier to write and maintain the code and the learning curve was steeper to learn how to fit the C code to the optimizer's quirks rather than to get caught up in the details of DSP assembler level code. For a few functions that occupied substantial amounts of the DSP's real time, we looked into writing these segments in assembler, but found that there was little to be gained over carefully written C[8].

The speech coder and the modem receiver are typically the more complex and real-time intensive of the system functions. Since the system used a sample rate of 97.2 kHz (24.3 kbaud with T/4 sampling), the process to get samples in and out of the DSPs in real-time drove the software design.

Figure 3 shows the control flow of the modem receiver. Since the TMS320C40 provides a high-speed communications port that has an 8 word FIFO, we organized our processing in 8 sample chunks. With 32 bits per word, this also allowed 16 bit I and Q samples to be read simultaneously from the A/D converter. Every 82 μsec, the receiver is interrupted to read and buffer 8 samples. DC restoration, filtering, sync word detection, clock control, handset antenna calculation and selection, demultiplexing of frames, differential detection, and interfacing with the channel decoder are handled as background processes that are completed within 20 ms.

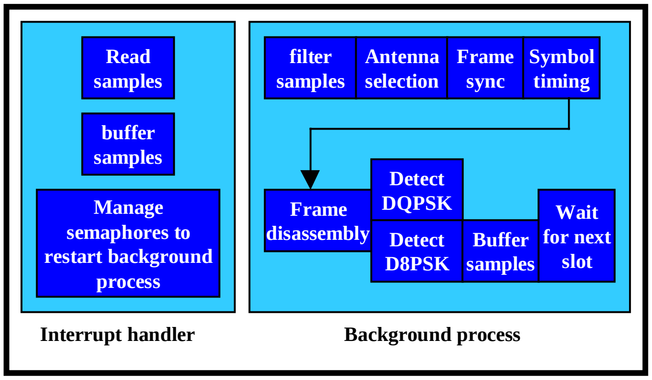


**Figure 3 - Receiver Software Structure**

The uplink base receiver was similar in structure to the downlink portable receiver. However, since the portable used preselection diversity, only one set of processes was needed. In addition, the portable station was able to adjust its clock phase and frequency to track the base clock. The base receiver does not have the flexibility to adjust its clock, since the base clock serves as the master clock for the system. Thus, a different approach was needed at the base with different signal processing requirements. We chose to design 4 square root raised cosine channel filters with 4 different phases, interpolating the T/4 sampling to 16 clock phases. The base receiver correlates the received sync signal in each channel against the possible phases to select the best receive filter (weighted by the signal level in each channel) for the two channels. The two receive channels are filtered and differentially detected separately. Equal gain combining (for two-branch diversity) promises performance that is quite close to optimum combining, as our experiments showed.

## III. ALGORITHMS

In this paper, we focus on the details of two of the more important modem functions: coding and synchronization.

### *A. Channel Coding*

Since the system design was to be optimized for an indoor, slow fading environment, interleaving and other forms of time diversity would not be effective. Due to the goal of high end user bit rate (16 kb/s) and the constraint of 130 data symbols per 20 ms slot, a high rate code was needed. Of the several possible codes studied [4], we chose the following code: 320 data bits per frame are broken into a half frame's worth of data - 160 bits. A 10 bit CRC is applied to each half frame, resulting in 2*(160+10) = 340 bits. These 340 bits are further broken down to 80 bits that will be coded and 260 bits that will not. The 260 uncoded bits are used to select the quadrant of the 8-phase constellation by treating them as the two MSBs of each symbol. The remaining 80 bits are padded with 6 memory bits used by the coding algorithm and result in 129 bits with a rate 2/3 code. One extra 0 is added to the 129 bits to give 130 bits, used as the LSB of each constellation symbol.

Through simulations and over-the-air tests, this code has performed as expected, providing 4-5 dB gain on AWGN channels and with slow fading channels where simple diversity was used. It is to be expected that most errors will be in the LSB of a detected symbol, since this is the nearest neighbor. A rate 2/3 code is effective for controlling these errors. Less frequently, errors that effect the MSBs will occur. These errors cannot be corrected, but will generally be detected by the CRC. Besides, if the channel conditions are severe enough to perturb the MSBs, it is quite possible that the channel decoder working on the LSBs will have already declared the frame to be undecodable. Figure 4 illustrates the channel coding structure.

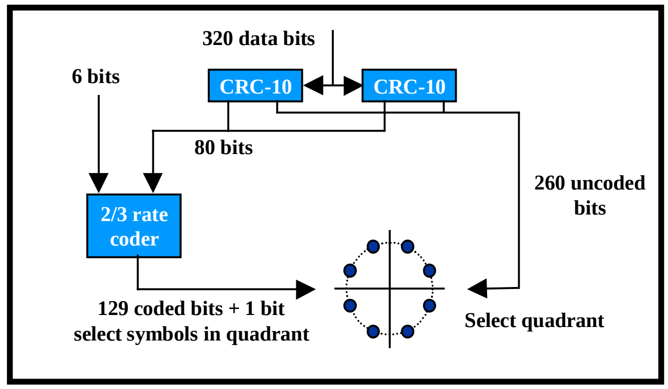


**Figure 4 - Channel Coding**

### *B. Synchronization*

Timing control is done in two stages - acquisition and tracking mode. During acquisition, the received signal is correlated against a known sync pattern for approximately 6 half frames. Each half-frame correlation should ideally have a single prominent peak. The resulting correlation functions are accumulated and summed. This composite correlation function is searched for a peak, most likely corresponding to the correct sync position. Rapid timing adjustments are made to align the receiver with the incoming signal. In our experimental prototype, this phase adjustment is accomplished by adding or deleting pulses from a divide chain driven by a high-speed clock.

After a preset time in acquisition mode, the receiver automatically switches to tracking mode. Here, the correlation is calculated over a small window, plus or minus a few symbols of the expected sync position. By comparing early and late samples, and filtering them by an integral plus proportional control loop, accurate timing is maintained in the presence of fading and momentary loss of signal. During the first few moments of tracking mode, the clock is probably at the right phase, but may be slightly off frequency, requiring a wide loop bandwidth to allow timing tracking. If the loop bandwidth were to remain broad, the receiver would be sensitive to noise and couldn't easily maintain timing during loss of signal. To counter this, we designed a time varying loop filter with a gradually decaying bandwidth. After entering tracking mode, the loop bandwidth decays to a minimum steady state tracking bandwidth, used until the receiver needs to reacquire timing. In this way, the receiver timing is able to “flywheel” through loss of signal, when the old estimate of timing frequency and phase is the best available information. By monitoring frame error rate, loss of sync can be detected, allowing reacquisition.

## IV. EXPERIMENTAL SETUP

Over-the-air demonstrations are the most effective way to show the performance of a wireless communications system and have been used extensively for this project. However, it is difficult to quantify results for this arrangement or to compare different systems, since precise SNRs, fading rates, and other channel conditions cannot be reliably replicated. For this reason, all of the performance measurements were conducted over simulated RF channels using a TAS-4500 RF impairment simulator. As shown in Figure 5, the fader provided two independently fading channels that could be used to alter the 900 MHz RF signals that the modem transmitter generated. For the downlink, the receiver selected between these channels.

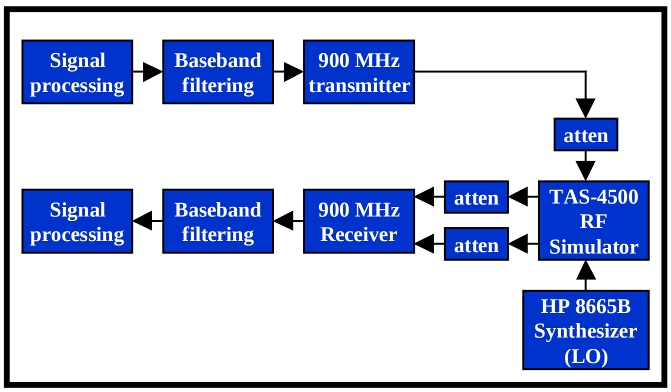


**Figure 5 - Experimental Hardware Configuration**

## V. PERFORMANCE RESULTS

Figure 6 shows the performance of the 16 kb/s channel coder under the best conditions - an AWGN channel. From the theoretical results[9] for differentially coherent D8PSK, it can be seen that our experimental implementation is within 1 dB of ideal. Under these benign conditions, the coder is providing 5 dB gain. (Note, for all of the performance curves, P(E) is the frame error rate (FER) while SNR is the average symbol power divided by average noise power.)

Figure 7 summarizes the system performance with a variety of test conditions on a 1 Hz fading channel: with and without the channel coding, and with and without preselection diversity. For most speech coding systems, a FER of 1% or better is required. As the figure shows, in this fading environment, without coding or diversity it was possible to attain this FER, but only at extremely high SNRs, much higher than would be practical to routinely expect in a deployed system. Compared to the 5 dB the coder provides on an AWGN channel, when we introduce 1 Hz fading the coding gain is less than half of what we previously obtained. Comparing the performance obtained without diversity and without coding to the performance obtained with diversity but without coding, it can be seen that at the 1% FER threshold, the system performance is improved by 10 dB. Finally, adding coding to the system with diversity, the system performance improves by an additional 4 dB, approximating the effectiveness of coding in AWGN.

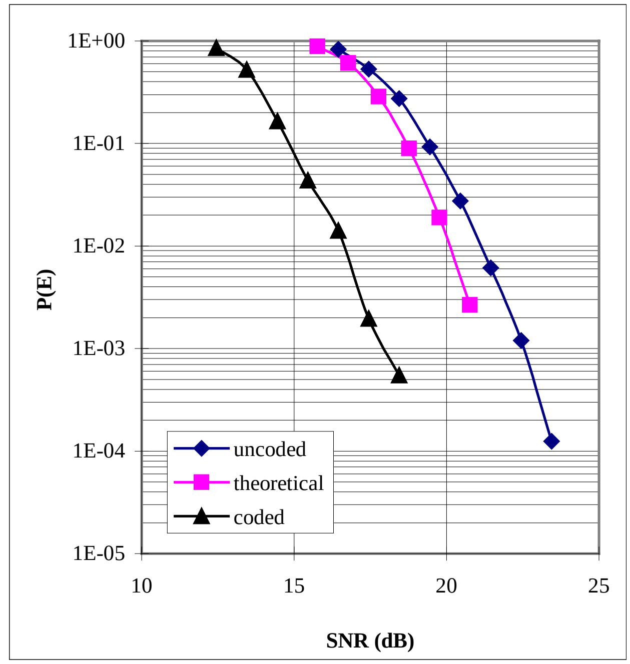


**Figure 6 - Performance on an AWGN channel**

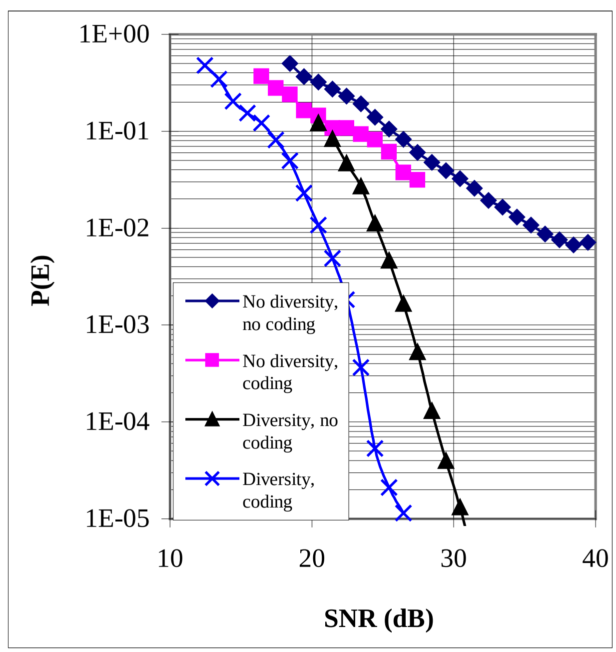


**Figure 7 - Performance with 1 Hz Rayleigh fading**

It is worth noting the relative performance of the system with and without diversity. Our field-strength based experiments and previous theoretical studies have shown that substantial gains are possible with antenna preselection diversity[5]. What is notable is how diversity and coding work together. From the figures above, it can be seen that in the absence of diversity, the coding gain that is obtained on an AWGN channel is mostly lost on a slow fading channel. Diversity, by itself, has some usefulness in improving system performance. In combination, however, the experimental results show that diversity allows nearly the full coding gain to be preserved - essentially, the diversity tends to make the channel look more like an AWGN channel to the coder.

The results presented above have focused on the 16 kb/s downlink performance in the presence of 1 Hz fading. At higher fading rates, there is a greater chance that the signal quality through the 6.66 ms slot may have changed from that measured during the preselection diversity power calculations. In addition, even if the antenna selected remains the better antenna for the duration of the slot, at higher fading rates, the probability of a fade occurrence during the slot is increased. Together, these effects degrade the system performance somewhat at higher fading rates, as shown in Figure 8.

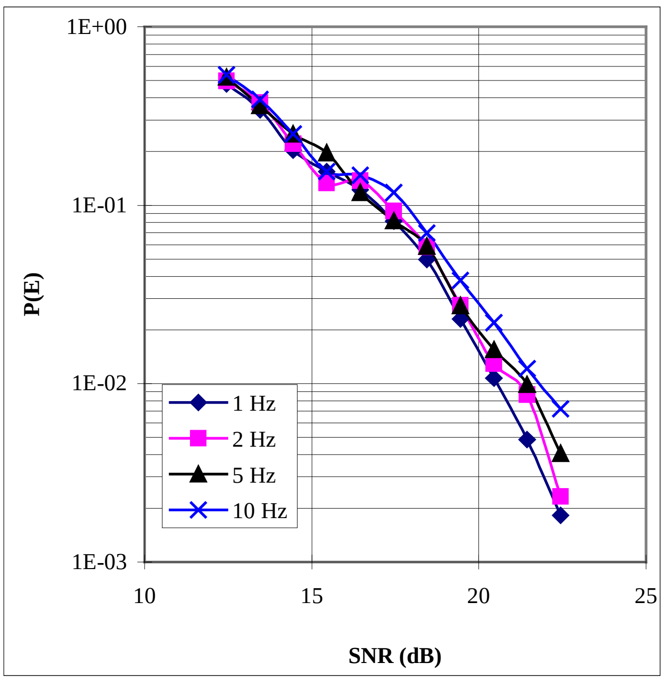


**Figure 8 - Performance at higher fading rates**

Since we focused our attention on the downlink for this portion of the study, it was necessary to pay careful attention to the constraints of the handset. In particular, since the handset is likely to be small, relative to the wavelength and since it will be challenging designing a second efficient antenna, we were interested in how antenna correlation and relative signal strengths influenced overall system performance. The system performance reported above reflects the results with uncorrelated fading and equal signal strengths at the two antennas. We tested performance with one of the antennas operating with -3, -6, and -9 dB gain, relative to the other antenna. For these conditions, we found that the degradation in SNR at the 1% FER were about half the effective antenna gain. That is with a -3 dB gain antenna, overall system performance was degraded about 1.5 dB. For the case where the correlation between fading on the two antennas was set to .5, we found about 1.5 dB degradation in system performance, just as theory predicts[10]. With both correlated fading as well as loss in one of the antennas, we found that the two impairments were additive. That is, with antenna correlation of .5 and 3 dB antenna loss, the overall degradation in performance was about 3 dB. These results are illustrated in Figure 9.

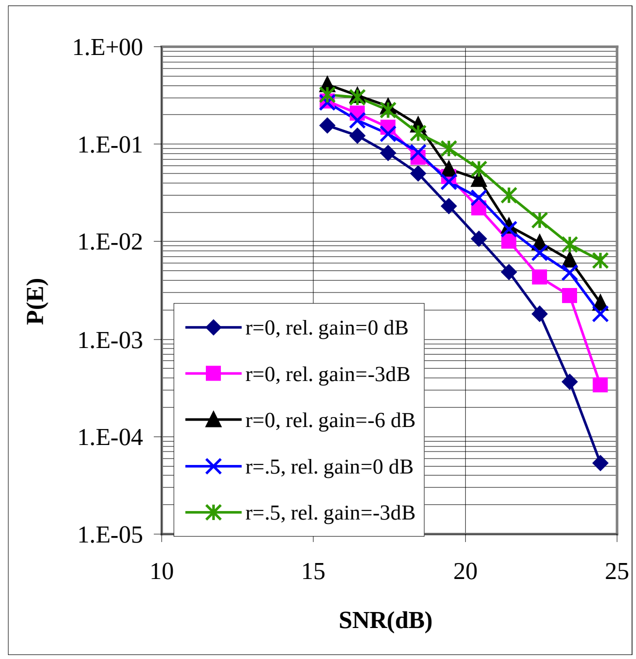


**Figure 9 - Performance with correlation and unequal antenna gains**

## VI. CONCLUSIONS

We have experimentally shown that, with the proper application of channel coding and antenna diversity, it is possible to obtain good performance at 16 kb/s through simple modifications to the existing IS-136 standard. While our investigations focused on the indoor environment, we have shown that other approaches to speech and channel coding can provide significant improvements in speech quality over a wider range of channels, potentially indoors and out.

## ACKNOWLEDGMENTS

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