Research/Projects Summary:

Amogh.R
Wireless Communication Engineer
MIMO OFDM group
ESQUBE Communication Solutions Inc
Bangalore,India

Details of the work done in the company:
Developed, tested and debugged MATLAB and C++ code for the *MIMO OFDM based 802.11n Transmitter* with important features like Space Time Block Coding, Cyclic Delay Diversity, Spatial Spreading, Simple Transmit Beamforming.

Undergraduate Program Projects:

1.) *Turbo Multiuser Detection in Coded Uplink CDMA Systems:*

Guide: Asst Professor C R Natraj, ECE Dept, SJCE, Mysore, India &
Dr Husheng Li, Qualcomm Inc, San Diego, USA.
Duration: 02/2006 - 05/2006
Team Size: 1

Multiuser detection schemes are required for increasing capacity in CDMA systems. Iterative multiuser detection based schemes provide near optimum performance at a reduced complexity. LDPC and Turbo codes due to their powerful error correction capability will become an essential part of a high performance communication system. Use of iterative multiuser detection in LDPC/Turbo coded CDMA environment provides very high capacity systems. Combining the two techniques in an iterative manner results in a joint iterative multiuser detection and decoding receiver.

Our work involved the study of turbo multiuser detection in two environments/systems.

- Turbo multiuser detection in convolutional coded synchronous uplink CDMA system
- Turbo multiuser detection in turbo coded asynchronous uplink CDMA system

Synchronous system considers the AWGN channel for each user. Asynchronous system considers the practical frequency selective multipath channel for each user.

A novel idea in the studied receiver system is a reduced complexity SISO (Soft Input Soft Output) detector which uses a decision statistic using the outputs of the maximum ratio combiners for all the users, similar to the RAKE receiver and performs soft-interference cancellation and instantaneous minimum mean-square
error (MMSE) filtering. SISO detector consists of a bank of parallel RAKE receivers, one for each user followed by SIC stage and a Linear MMSE filtering stage. SISO detector is followed by a bank of parallel extrinsic LLR turbo decoders. There are two types of iterations in the receiver, where the outer iteration exchanges soft information between the multiuser detector and the extrinsic LLR decoders and the inner iterations compute the a posteriori LLR
of each code bit within the extrinsic LLR decoders. The extrinsic LLR turbo decoder consists of constituent MAP decoders whose number is determined by the number of constituent encoders in the transmitter turbo encoder. The constituent MAP decoder uses a modified version of the standard BCJR algorithm. The non linear MAI suppression technique used by the SISO detector gives near optimum performance with computational complexity reduced from that of a optimum multiuser detector. At each iteration, extrinsic information is extracted from detection and decoding stages and is then used as a priori information in the next iteration. Although the iterative receiver system studied can be used for convolutionally coded or block-coded systems that use soft-decision channel decoding, the extrinsic information delivered by extrinsic LLR turbo decoders are more reliable than the one of convolutional or block decoders, because of the parallel or serial concatenated structure of turbo codes.

The two systems were implemented and performance analysis under different conditions was done. The two systems were tested with varying number of users, varying number of iterations (outer and inner), different frame sizes and in strong-weak user environment.

Experience:

1. The initial idea of this project was to study joint iterative equalization and multiuser detection in dispersive DS-CDMA channels based on a reduced state BCJR algorithm. It is based on applying both the state-reducing techniques for the single user trellises and partitioning the soft feedback from channel decoders into unreliable and reliable sets using a confidence metric. The states of the single user trellises are constructed with the unreliable set and the interference from the reliable set is cancelled. Due to lack of literature and subsequent superficial knowledge regarding the theory of reliable and unreliable set of soft feedback, it was decided that iterative MUD for uplink CDMA in multipath channels will be studied.

2. The system was tested in strong-weak user environment under both symmetric and asymmetric conditions. In both cases it was observed that the weak user(s) performed better than or equal to strong user in terms of BER. It was not possible to explain/study this phenomenon analytically and remains a simulation observation.

3. When testing the system for its robustness by varying the number of outer and inner iterations and recording the corresponding BER, it was observed that BER improvement reached a saturation level for some 5 outer and inner iterations each. Any increase in the number of iterations above 5 resulted in no further BER improvement. This simulation observation not predicted by the theory of iterative exchange of soft information can be explained/studied by Stopping rules concept.

4. Under testing conditions, the processing gain of the system was increased in fixed steps hoping for a better BER performance. But with increased PG, the system BER performance did not improve and remained at the level corresponding to initial PG.

5. Although the theory of studied Turbo multiuser detector states that any number of CDMA users can be supported on the channel. The simulation showed that the maximum number of users for acceptable BER performance is 5. If the number of users was increased beyond 5, the BER performance degraded. This was due to errors in the soft estimates of the user’s code bits used by the SIC stage to cancel interference and thus due to improper interference cancellation, BER worsened.

2.) *Subspace Based Blind Linear Multiuser Detector:*

Duration: 04/2005
Team Size: 1

Subspace based blind linear multiuser detector for downlink CDMA offers advantages over blind linear multiuser detectors based on direct matrix inversion (DMI).The blind detectors are based on estimating the signal subspace spanned by the user signature waveforms.

Estimation of user signature waveforms spanning subspace leads to blind implementation of both the linear decorrelating detector and the linear MMSE detector. Each of these blind detectors are expressed in terms of the estimated signal subspace parameters. Both the linear decorrelating detector and the linear MMSE detector can be estimated from the received signal with the prior knowledge of only the signature waveform and timing of the user of interest.
Our work involved only batch processing systems in an AWGN channel environment, although adaptive blind linear multiuser detectors are discussed in the literature.

Batch processing systems were implemented and performance analysis was done by varying the number of users in both the cases.

Note: There was no explicit guide for this work but occasional guidance was obtained from Dr Husheng Li, Qualcomm Inc.

3.) *Spoken Language Recognition:*

Guide: Dr Ashok Rao, CEDT, IISc, Bangalore, India.
Duration: 02/2005 – 03/2005 Summer Project
Team Size: 1

Hidden Markov Model is a statistical model in which the state sequence is hidden & output sequence is observable. Each language to be recognized is statistically modeled by a Hidden Markov Model. HMM is a widely used statistical model used in speech recognition, Language recognition, bioinformatics, weather forecasting & ISI Channel modeling in wireless communication etc.

The feature vectors corresponding to input speech necessary for Language Recognition are derived from LPC Front end analysis. The LPC order and the cepstrum order used for the front end analysis are 10 and 12 respectively. The HMM's used for language modeling were ergodic. The output of the HMM's used were continuous valued vectors with multivariate Gaussian mixtures densities. In our implementation, each language is modeled by a 5 state HMM & each state having 5 Gaussian mixture components. Segmental K-Means algorithm training was used to train the HMM's for specific languages. From a random initial HMM for each language, the optimum HMM is obtained using Segmental K-means training. For a given test utterance, the language corresponding to the HMM which produces the maximum Viterbi likelihood of observation is hypothesized as the spoken language. Improvements to the current system can be done by employing Baum Welch/Forward Backward algorithm for HMM generation and training.

Although this project is related to the field of "Pattern Recognition", training and testing algorithms for HMM patterns studied & implemented will be useful since HMM 's are recently used in Narrowband Interference suppression in CDMA, Nonlinear equalization of ISI channels.

The system was implemented at MATLAB level.

Experience:

1. Initially started with a 5 state HMM but after sufficient training of the HMM's it was learned that only 1 state is sufficient for the HMM's for language modeling.

2. Training was done with Segmental K-Means algorithm. Baum Welch algorithm was tried for training with the hope of building better performance HMM's but the entire algorithm was prohibitively computationally expensive and slow. Techniques to reduce the computational time and cost were tried but they were not useful. So training using Baum Welch algorithm for HMM's had to be given up. If more time was available for the project may be what was happening with the Baum Welch algorithm could have been figured out.

3. For language recognition, Forward likelihood and Backward likelihood were used as decision metrics. But these two did not perform well to an acceptable level when fed by a Viterbi state sequence. But Viterbi likelihood did a good job when fed by a Viterbi state sequence. So for language recognition, Viterbi likelihood was used as a decision metric.

4.) *Multi Tone Code Division Multiple Access:*

Guide: Asst Prof S Gopala Krishna, ECE Dept, SJCE, Mysore, India
Duration: 04/2005 – 05/2005
Team Size: 1

OFDM is a multi carrier modulation technique used for high data rate communication due to its resistance against frequency selective fading ISI channels. This scheme is included in standards like IEEE 802.11x, Hiperlan, ADSL etc. The main aim of combining OFDM with CDMA is to provide high data rate services in multiple access systems under hostile wireless environments. The primary benefit of combining OFDM with DS spreading is that it is possible to prevent the obliteration of certain subcarriers by deep frequency fades. This is achieved by spreading each subcarrier's signal with the aid of a spreading code and thereby increasing the error resilience, since in case of corrupting a few chips of a spreading code the chances are that the subcarrier signal still may be recovered.

Our work involved the study of a Multi Tone CDMA system operating in AWGN channel. OFDM part used QAM modulation and for spreading, short spreading codes were used. The number of subcarriers used was 32.Generic MT-CDMA transmitter and receiver simulation blocks operating in AWGN enviornment were implemented in Simulink using C. AWGN channel is too simple a condition to be operating in ,so practical hostile wireless channels need to be considered.

5.) *Anti Jam Communication System:*

Guide: Asst Prof S Gopala Krishna, ECE Dept, SJCE, Mysore, India
Duration: 11/2004 – 12/2004
Team Size: 1

Spread spectrum modulation is used for anti jam communication .The two types of spread spectrum modulation used are

- Direct Sequence
- Frequency Hopping

Our work involved the study and implementation of generic forms of anti jam communication systems using the two methods. The two systems were tested against the following types of signals under AWGN condition

  - Single tone jamming signal
  - Multi tone jamming signal
  - Pulse jamming signal

The following simulation blocks were developed in Simulink using C.
DS-SS transmitter
DS-SS receiver
FH-SS transmitter
FH-SS receiver

6.) *Residual Excited Vocoder:*

Duration: 11/2004, 15 days
Team Size: 1

Waveform coders and vocoders are the two forms of speech coders widely used. The analysis-synthesis approach based vocoders are used in low bit rate speech coding. Linear prediction based vocoders are the most widely used class of vocoders. Residual excited linear prediction vocoder is one among the different types of LP based vocoders and also the simplest.

The speech encoder part does the following

- Short term linear prediction(STP) analysis
- SIFT based pitch estimation
- generates LP residual

The speech decoder uses the parameters from STP analysis, pitch period, U/V decision and generates the LP speech, which is residue corrected to yield the approximate speech at the receiver. Pitch estimation of the speech frame is done using Simple Inverse Filter Tracking algorithm developed by Dr

John D Markel in his paper “The SIFT Algorithm for Fundamental Frequency Estimation” IEEE Transactions on Audio and Electro acoustics, December 1972.

7.) *Equalization Receiver for ISI Channels:*

Duration: 04/2006, 15 days
Team Size: 1

MAP based iterative equalization receiver system for ISI channels under coded environment were implemented and performance analysis of the system was done.
Frequency selective ISI channel was modeled as a rate 1 convolutional encoder. The channel ISI equalizer was then a MAP based convolutional decoder with soft output .The soft output from the MAP equalizer is fed to the turbo decoder as a priori information of the code bits. The a posteriori information of the code bits generated by the turbo decoder is used for the a priori information by the MAP equalizer.

8.) *Synchronous Digital Systems:*

Guide: B A Sujathakumari, ECE Dept, SJCE, Mysore, India.
Duration: 10/2004
Team Size: 1

Floating Point Multiplier
Booth’s Multiplier
Square Rooter
BCD-Binary Converter
Floating Point Adder/Subtractor
SBI & SBF Divider

The mentioned synchronous digital systems were implemented on a Xilinx FPGA, available in the college Electronics & Communication Engineering (ECE) department laboratory using VHDL. The work involved the following

1.) Complete design of digital hardware for the above systems.
Each system hardware had 2 components

- Data Processor: Responsible for the actual data inputs processing and generating data outputs for the digital system.
- Control Logic: Responsible for the control signals needed by the Data processor.

2.) Writing VHDL code for each digital hardware design.