

Vinay M K

Objective

To seek a lead position in speech, audio, image or video processing/coding and application of the same on practical real world systems. To move up along both the ladders of management and technical verticals.

Education

1999 – 2001 **Indian Institute of Technology** **Kanpur**
Master of Technology, Digital Signal Processing.

- CPI 10 (maximum) on a scale of 10.
- Thesis work accepted at ICASSP – 2002 and TENCON – 2001.

1994 - 1998 **National Institute of Engineering** **Mysore**
Bachelor of Engineering, Electronics and Communications

- University of Mysore second rank holder among more than 300 students.
- Passed with an overall (for all the eight semesters) aggregate of 83%.
- Secured the Chandrashekar memorial award for being the highest average scorer in mathematics among students of all branches for all the four semesters.

Summary of Experience

Apr 2001 – Till Date **Emuzed India Private Limited** **Bangalore**
Senior Design Engineer

- Working in audio and speech coding team, developing reference and optimized codes (on DSP and RISC platforms) of GSM-AMR (NB), MPEG-1, MPEG-2 & MPEG-4 audio coding. Error concealment solutions for GSM AMR speech codec.

July 2000 **FASET, Dept of CSE, IITK** **Kanpur**
Faculty

- Faculty for C++ (theory, lab & projects), as a part of summer vocational job.
- Handling a class of about 30 for one and a half months.

1999 – Feb 2001 **Dept of Electrical Engineering, IITK** **Kanpur**
Teaching Assistant

- Worked as one of the departmental web page developers, under the GATE scholarship for the masters program.

1998 - 1999 **Infosys Technologies Limited** **Bangalore**
Software Engineer

- Worked as a trainee on BSD-UNIX platform and retailing business domain.

Title	<i>GSM AMR (Narrow band) codec reference code development</i>
Group Size	2
My Role	Interface design, memory optimization (both stack as well as heap), reduction in code size, reduce function calls & provide support for optimization team.
Technical details	<ul style="list-style-type: none"> - Started with fixed point version available from ETSI – 3GPP site - Develop a re-entrant code/library - Reduce un-necessary heap variables - Optimize codebook search and other critical functions

Title	<i>GSM AMR (Narrow band) codec floating point to fixed point conversion</i>
Group Size	2
My Role	Conceptual design, writing fixed point versions of math functions like log, pow2, exp, sqrt etc.. Design of data flow to optimize computation and data flow complexities.
Technical details	<ul style="list-style-type: none"> - Estimating formats of various intermediate stages and design data path for efficient usage of precision and computing power. - Avoid re-computations by optimal storage. - Design of block exponent algorithms for efficient codebook search.

Title	<i>Error concealment solution for celp based encoded speech transmission on IP</i>
Group Size	2
My Role	Conceptual design, conceiving a suitable algorithm matching our interface, memory, computation power, delay constraints and minimum quality thresholds.
Technical details	<ul style="list-style-type: none"> - Quantifying the steps to be incorporated in our error concealment solution. - Solely decoder based, consistent interface independent of channel error resilient techniques were the initial constraints for algorithm design. - Suitable and efficient parameter estimation based on phoneme region. <p><i>Finer technical details of the project cannot be disclosed as it is the propriety of Emuzed Inc., But an overview of the same can be obtained by [1] in publication list above.</i></p>

Title	<i>Optimisation of GSM AMR NB codec on ADI Blackfin family of processors</i>
Group Size	3
My Role	Design of constant table usage avoiding extern variables.
Technical details	<ul style="list-style-type: none"> - Xdias a propriety vendor-application interface framework as developed by Texas Instruments was used. - Redesign of the algorithms to make efficient usage of dual MAC, vector operations and byte operations. - Optimisation of address calculations by employing local pointers - Effective usage of hardware loops and redesign of code to avoid stalls.

Title	<i>MPEG-1 audio layer 1 & 2 encoder reference code development & floating to fixed-point conversion</i>
Group Size	3
My Role	C level optimization of modules like rate control, psycho-acoustic model quantization etc., Implementing various math operations in fixed point

	arithmetic. Estimating optimal data formats based on audio quality improvement.
Technical details	<ul style="list-style-type: none"> - Reduction of complexity by reusing various parameters like energies, filter bank values etc., - Implementing psycho-acoustic model 1 & 2 in fixed-point arithmetic with fine-tuning for good audio quality.

Title	<i>MPEG –2 & 4 advanced audio codec development</i>
Group Size	3
My Role	Implemented stereo processing, TNS, psycho-acoustic, block switching modules in the encoder. Adding features like frequency transformations, stereo to mono and vice versa, etc on the decoder.
Technical details	<ul style="list-style-type: none"> - Encoder implementation involved adding stereo processing block, block switching, temporal noise shaping & psycho acoustic modules. - Fine tuning the psycho-acoustic model for improving the audio quality. - Optimization at design level for reduction in computational complexity. - Decoder implementation involved adding features like sampling frequency conversions for an already existing decoder library. - Solving tricky issues related to AAC_LTP object, stereo to mono conversion and back.

Title	<i>Optimisation of MPEG –2 advanced audio encoder on ARM9 of DM310 chip</i>
Group Size	1
My Role	Optimization of AAC encoder on ARM9TDMI
Technical details	<ul style="list-style-type: none"> - Implemented audio IO via McBSP serial port and AIC-23 on DM310. - The encoder is implemented on ARM sub system side of the SOC. - C level optimization

Title	<i>Float to fixed point conversion of TIA/EIA speech codec</i>
Group Size	2
My Role	Providing technical guidance for float to fixed point conversion of the celp based 14.4kbps speech codec.

Mtech Thesis

Nonlinear frequency warping in speaker normalization

- Application of various warping factor functions in speech recognition problem. A comparative study of warping methods with scale invariance transforms, in speaker normalization efforts.
- Frequency domain uniform and non-uniform warping implementation in ML-sense on a HMM based continuous-speech digit recognizer
- Uses HTK toolkit and also an independent implementation using C++ modules adapted from RES software.
- A simple normalization method proposed based on the work was accepted at TENCON-2001.
- Along with more adaptation and extra effort by two more students the complete work was presented as a conference paper at ICASSP-2002.

Guide Dr. S. Umesh, Asst. professor, IITK.

BE Project

Simulation of ARINC-429 Communication Protocol. National Aerospace Laboratories, Bangalore.

Abstract:

The project implements a part of the larger project meant for avionics suite testing. A user friendly, interactive application layer is built (Using C++ and OOP concepts), which accepts data in engineering format and converts it to ARINC standards as per ARINC 429 conventions. Bit oriented protocol is implemented with the help of an ARINC card, to set up the communication environment. The software is tested and verified on a twin computer set up, at National aerospace laboratories (NAL), Bangalore.

Guides : Mrs. Vijayalakshmi Shankar, Scientist NAL, and Bangalore.
Dr. H. S. Shivaram, Prof and head E&C dept, NIE Mysore.

Publications

- Vinay M K and Suresh Babu P V, “*Context based error recovery technique for GSM AMR speech codec*”, International conference on acoustic, speech signal processing (ICASSP) 2002, Orlando Florida, USA, May 13-17 2002.
- S Umesh, V Bharatkumar, Vinay M K, Rajesh Sharma and Rohit Sinha, “*A Simple approach non-uniform vowel normalization*”, International conference on acoustic, speech signal processing (ICASSP) 2002, Orlando Florida, USA, May 13-17 2002.
- Manish Arora, Suresh Babu PV and Vinay M K, “*RISC processor based speech codec implementation for emerging mobile multimedia messaging solutions*”, 14th International conference on digital signal processing, DSP-2002, Santorini, Greece, July 1-3 2002.

Soft copy of the thesis, above publications and other works of mine can be downloaded from my web page at <http://mkvinay.tripod.com/publications.htm>

References can be provided if required.