NEW TECHNOLOGY REDUCES SIZE AND COST OF COMMUNICATION SYSTEMS

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ABSTRACT

A communications system's function in a training environment is twofold. It must provide the necessary audio links among the students and instructors, and it must model the characteristics of real world devices. This has typically resulted in considerable amounts of hardware dedicated to specific processing tasks and an unwieldy dedicated line distribution system. A few companies have developed digital audio systems to take advantage of a time division multiplexed distribution system and to perform the primary mixing function. A natural extension of this is complete digital processing. The relatively new, single chip, Digital Signal Processors (DSPs) provide a cost effective means to implement signal processing algorithms and achieve the audio processing necessary to model a variety of communication devices.

This paper presents a communication system architecture that uses this new technology to cost effectively meet the processing and performance requirements encountered in the training and simulation environment. Prototype results have indicated a hardware reduction ratio on the order of 10 to 1 circuit boards for a typical (four person) operational flight trainer.

INTRODUCTION

An instructor giving direction to a student is an inherently different situation than an instructor role playing as an air traffic controller. The requirements on the audio for the two situations are derived from different criteria. Adequate intelligibility is the goal for the instructional audio, while high fidelity to the real world communication equipment is the objective of the simulation audio. These disparate motives can result in widely varying requirements for the trainer's communication system.

An audio channel can usually be described by its frequency response, signal to noise ratio, and distortion properties. The frequency response defines how the channel amplifies or attenuates the different frequency components of the input. The signal to noise (S/N) ratio is a measure of how loud the signal is compared to the background noise picked up or generated by the channel. The distortion is a measure of how much the output signal waveform differs from the input waveform due to new frequency components generated by non-linearities in the channel.

When intelligibility is the primary goal of the communication system, the requirements on the channel properties are static. Essentially noise and distortion are minimized and sufficient bandwidth is provided to pass the majority of the audio source's frequency components. A 3 KHz bandwidth is typical for voice channels since it allows most of the speech to pass while attenuating high frequency noise.

When simulating real world communication equipment, the trainer audio channel's characteristics must be dynamic. As a trainee switches from a radio to an intercom for instance, there would typically be an increase in bandwidth and a decrease in noise level. Other variation is inherent in the equipment. The signal attenuation as a radio moves away from the transmitter can be modeled as a time varying channel gain. Except for some special cases, sufficient fidelity can be achieved by modeling the channel bandwidth and the signal to noise ratio. Distortion effects can usually be neglected. Occasionally a specific type of distortion is intentionally added to a signal (Automatic Gain Control (AGC) and Voice activated circuits (VOX) can be considered a type of intentional distortion) and its effect is significant enough to require modeling.

Figure 1 depicts a simple audio channel model. Note that the modulation and demodulation process has been omitted. Except in some malfunction models the distortion produced by these processes is not significant and only the frequency response of the baseband signal and the channel noise is considered. In practice the transmitter and channel transfer functions are not realized separately, but are instead combined to reduce hardware in analog systems or processing time in digital systems. It should also be noted that very precise transfer functions are not necessary since the human ear is relatively insensitive to small changes in bandwidth (especially for speech). The goal is to provide a distinction between different pieces of equipment. Intercoms can have bandwidths as high as 6 KHz, while most radios are in the range of 1.5 to 3 KHz. Simple band pass filters are usually sufficient to provide the distinction. Occasionally, special transfer function models are required. A recent example was a 1020 Hz notch filter requirement in a receiver model.

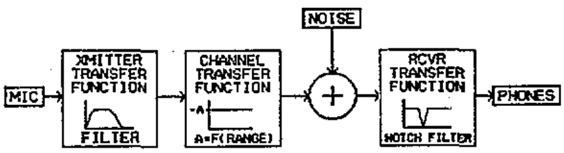

Figure 1

ANALOG TRAINER COMMUNICATION SYSTEMS

The traditional approach to trainer communication system design has been dedicated analog processing circuits with point to point wiring providing the distribution. Physically, this meant at least a cardfile of printed circuit boards and cabling to each headset interface. Every possible communication combination had to have a dedicated audio path and a full complement of the audio processing hardware. As the number of personnel increased linearly the amount of hardware and interconnections would rise exponentially. Even a relatively small four person Operational Flight Trainer (OFT) required ten to twenty analog audio processing boards.

System scaling has been a persistent problem with analog communication systems. To design a generic system with modular hardware that would handle the range of personnel encountered in training is hampered by the vast interconnection requirement. To add one independent headset, potentially required connecting to every piece of existing hardware. Although some creative methods have been found to limit the complexity of the hardware, most result in a decrease in performance (usually as restrictions on how general the audio mixes can be). Ideally, a system would have an open ended modularity in which any of the M outputs could be any weighted sum of the N inputs. This ideal is usually sacrificed for simpler systems in which audio channels are submixed, distributed, and remixed. The simpler scheme does not easily allow private communications and is difficult to reconfigure for equipment updates.

The primary function of a trainer's communication system is audio mixing. It can be described by a single matrix equation:

$$\begin{bmatrix} Y1 \\ Y2 \\ Y3 \\ \vdots \\ YM \end{bmatrix} = \begin{bmatrix} A11 & A12 & A13 \cdots & A1N \\ A21 & A22 & A23 \cdots & A2N \\ A31 & A32 & A33 \cdots & A3N \\ \vdots & \vdots & \vdots & \vdots \\ AM1 & AM2 & AM3 \cdots & AMN \end{bmatrix} \begin{bmatrix} X1 \\ X2 \\ X3 \\ \vdots \\ XN \end{bmatrix} \quad (1)$$

Each output, Yi, is the weighted sum of the N inputs, Xj. The weighing coefficient, Aij, is the relative volume level of the jth input in the ith output. The volume matrix is usually sparse since there is only a limited number of signals that can be listened to intelligibly. It is also dynamic. As changes are made by the personnel (switch selections, volume pots, etc.) it is the volume matrix which must be updated. Analog systems typically use summing amplifiers with independent variable gain on each input to provide the mixing function. Submixing is also common since it limits the amount of interconnections and summing hardware at the expense of a completely general mixing function (an output can only be the sum of previously submixed inputs).

The secondary functions of the trainer communication system are the transfer function models. Analog systems frequently use some type of adjustable high and low pass filters and some form of analog signal gating to vary the frequency response. Noise is often generated by amplification of the thermal noise in a semi-conductor and added with a summing amplifier.

Other functions that may be embedded in the communication system include tone generation, VOX, AGC, voice masking, etc. Like the primary and secondary functions each analog implementation uses dedicated circuits with little or no functional multiplexing. Each piece of hardware is dedicated to performing a single task.

DIGITAL SIGNAL PROCESSORS

Digital signal processing [1] is a branch of engineering concerned with discrete time systems. Signals are represented as sequences of numbers. Processing requires applying algorithms to the numeric sequence in such a way as to provide some useful function. The process may be analogous to an analog function (such as filtering) or be something uniquely digital (such as a Fast Fourier Transform). The fundamental elements are delay units, multipliers and accumulators. These are combined to execute the desired algorithm.

In contrast to analog systems, digital systems can trade off hardware and processing time. Addition of a new function may only mean adding a new software process and does not necessarily mean additional hardware. Because of the finite processing speeds of digital computers the algorithms were initially implemented in non-real time. As VLSI techniques improved, special building block components were devised that allowed real time signal processing. Although extremely fast, these components lacked the versatility of the general purpose computer. They are most effectively used in high throughput fixed process systems.

In 1983 Texas Instruments introduced a single chip Digital Signal Processor (DSP). It combined the hardware multiplier-accumulator with the programmability of microprocessors. Motorola, AT&T, Analog Devices and others have also introduced DSPs. Many are now producing second and third generation parts. Although not as fast as the building block components, these DSPs are capable of working in real time and provide the versatility of functional multiplexing.

Just as time multiplexed distribution allows the sharing of interconnection hardware, function multiplexing allows the sharing of processing hardware. Assuming an algorithm can be developed and sufficient processing time is available, any function or functions can be provided by a DSP. Once an architecture is devised, hardware circuit design is replaced by DSP code development. The programming of a DSP is similar to microprocessor assembly coding. A good foundation in digital signal processing theory is necessary for successful algorithm development. Third party software packages can aid in code development and analysis.

Selection of digital signal processing hardware is dependent on a number of system parameters. The number of channels, the maximum bandwidth, and the complexity of the processing algorithms, all play a major role in the processing budget. Since most DSP algorithms are multiply/accumulate intensive, the figure of merit most useful for comparison is the multiply/accumulate cycle time. For most processors this is equivalent to the instruction cycle time. Presently, 100 nsec is typical, although faster processors are becoming available.

The maximum analog bandwidth is the primary system parameter influencing DSP work load. The Nyquist Criteria [1], states that the rate at which an analog signal is sampled must be at least twice the highest frequency component. For audio systems this can range from 8 KHz (3 KHz BW) for voice only channels to 44 KHz (20 KHz BW) for high quality compact disc recording. Trainer communication systems are typically near the bottom of this range. Depending on the equipment to be simulated and the frequency range of the audio sources (voices, tones, noise, special functions) the sample rate for typical training systems can range from about 8 to 16 KHz. With a given sample rate and the 100 nsec instruction cycle time, the number of instructions executable in a sample period can be estimated. At 8 KHz 1250 instruction cycles can be processed in a sample period. Since instructions typically require 1 to 2 cycles to execute, approximately 800 can be executed in a sample period. At 44 KHz this drops to about 150. In training systems with 8 to 16 KHz sample rates, four to eight hundred instructions per period per processor are possible.

Even with efficient algorithms and coding, the four to eight hundred instructions per sample period are not sufficient to process all the audio in a trainer. Most will need multiple digital signal processors. The challenge is to design a modular architecture that allows the amount of hardware to be scaled to fit the needs of a particular training environment while still reducing the cost compared to an analog system.

THE DIGITAL COMM/NAV AUDIO SYSTEM

Features:

* Modular system expansion
* Minimal hardware, only two board types
* Single processor board for typical OFT
* VMEbus based, capable of sharing I/O System's hardware and host interface
* Time multiplexed audio distribution eliminates point to point wiring
* Transmitter and receiver transfer function modeling
* Channel noise modeling
* Special embedded features include: digital tone and noise generation, AGC and VOX simulation
* High bandwidth and low noise

HARDWARE

The Digital Comm/Nav Audio System is based on a modular channel architecture. An audio channel (Figure 2) is an input output pair, typically a headset (microphone and headphone), but may also be a record playback unit, cockpit speaker, real acoustic gear, or other audio equipment. Up to one hundred channels can be included in a single system and provisions are provided for joining entire systems.

The audio acquisition is accomplished on the Digital Audio Interface board. It is situated physically close to the analog audio source. This small board interfaces to the various headsets and audio equipment. It performs the analog to digital and digital to analog conversions. Audio samples are transmitted to and received from the Digital signal processor (DSP) on the Digital Audio Processor board. The minimal analog signal path and the serial data transmission provide a high degree of noise immunity.

The Digital Audio Processor is a multi-channel processor/distributor/interface board. It interfaces to the acquisition hardware, the host computer (through VMEbus), and the high speed audio distribution bus. Only the P1 connector is used for VME interfacing. P2 is used for the high speed audio distribution. Current prototypes have two channels per board but four channel production boards are expected. The digital signal processor provides all the audio processing for the channel.

Use of the VMEbus allows flexibility in system configuration. The boards can be placed in a VME chassis with CPU and I/O cards to form a completely independent sub-system or they can be inserted directly into a VME based I/O system and be Host computer driven. The former configuration is appropriate for large systems where off-loading the Host computer is desirable. The latter is suitable in smaller systems where minimizing hardware is the primary goal.

System configuration and layout are simplified since there are only two board types. Each headset plugs into a Digital Audio Interface board which is cabled to one channel of a Digital Audio Processor board. A single four channel board processes all of a typical OFT's audio. Scaling the configuration for larger systems requires inserting additional Digital Audio Processor boards in the VME bus. Figure 3 depicts the hardware configuration for a four channel OFT. A functionally equivalent analog system is included for reference.

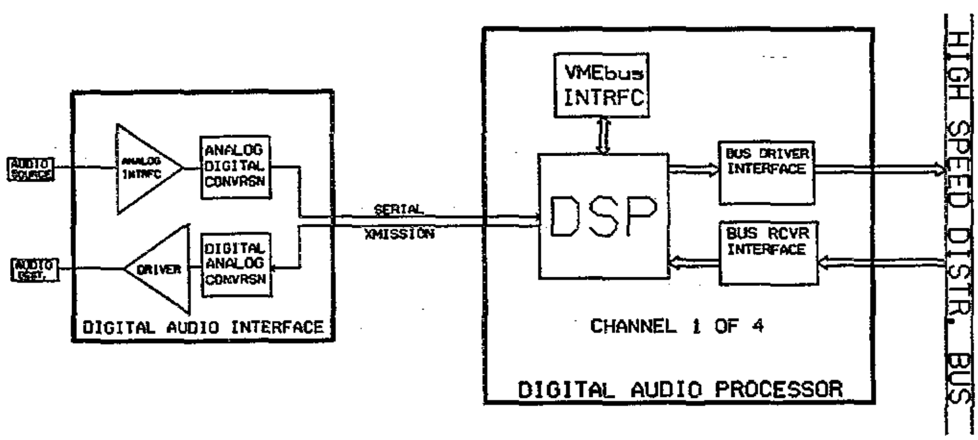

Figure 2

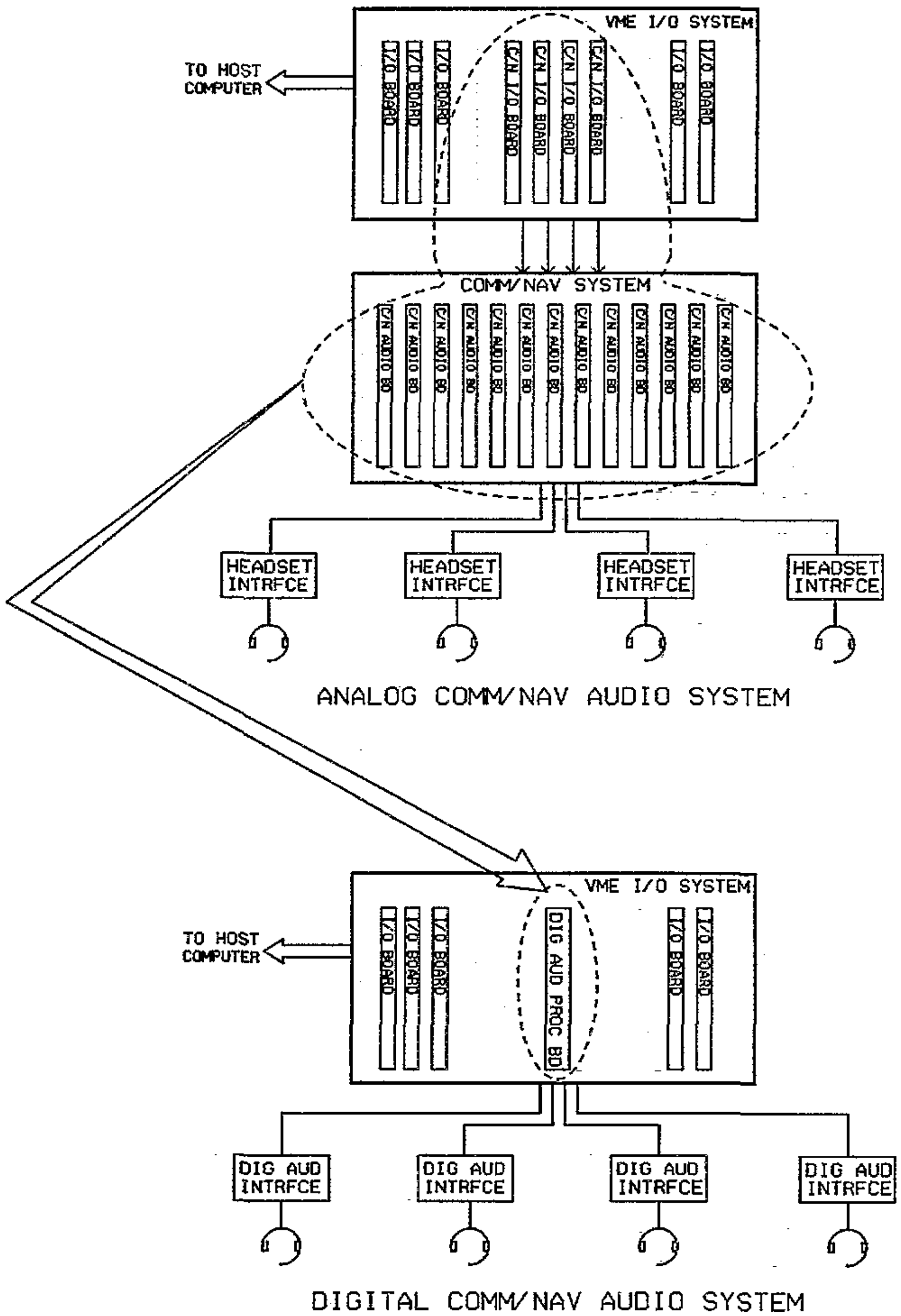
VME I/O SYSTEM
TO HOST COMPUTER
I/O BOARD
C/N I/O BOARD
COMM/NAV SYSTEM
C/N AUDIO BD
HEADSET INTRFCE
ANALOG COMM/NAV AUDIO SYSTEM
VME I/O SYSTEM
TO HOST COMPUTER
I/O BOARD
DIG AUD PROC BD
DIG AUD INTRFCE
DIGITAL COMM/NAV AUDIO SYSTEM

Figure 3

SOFTWARE

As apparent from the minimal amount of hardware, the Digital Comm/Nav Audio System relies heavily on the DSP code to perform all the communication functions. The sample flow through the processor routines is indicated in figure 4. The code is fixed in EPROM memory, but has been designed to be parameter driven. All real time variables are downloaded through the VME control bus. These include volume levels, filter coefficients, VOX thresholds, etc. Although not currently utilized, the system is capable of receiving actual DSP code through the VME download. This will allow future systems to provide new functions without hardware (or even EPROM) changes. Dynamic resource allocation is also possible.

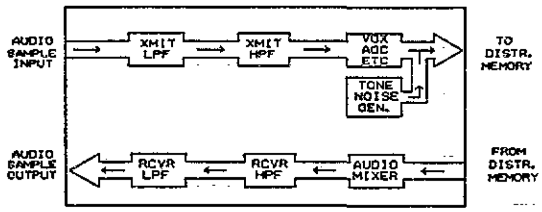

Figure 4

As discussed previously, Digital Signal Processors are extremely efficient at multiply/accumulate operations. In order to exploit this feature, algorithms are developed around sum of product functions. The mixing routine of equation (1) can be broken into M summations of the form:

$$Y_i = \sum_{J=1}^{N} A_{iJ} X_J \qquad (2)$$

This fits neatly within the system's architecture since each processor generates one output sample from the N inputs supplied from the distribution system. The collection of coefficients Aj for a given output (the volume vector) is downloaded from the host through the VME bus.

The filters can be easily shown to be sum of product operations [1]. Any transfer function that can be described by the ratio of two polynomials in the complex Z domain:

$$H(z) = \frac{\sum_{k=0}^{M} Bk \; z^{-K}}{1 - \sum_{k=1}^{N} Ak \; z^{-K}} \qquad (3)$$

can be realized by the difference equation:

$$y(n) = \sum_{k=1}^{N} ak \; y(n-k) + \sum_{k=0}^{M} bk \; x(n-k) \qquad (4)$$

In the mixing algorithm each output sample is the sum of every input sample multiplied by a volume coefficient. In contrast the filter algorithm acts on a single signal flow. The output (of the algorithm) is the sum of previous inputs [x(n-k)] and previous outputs [y(n-k)] multiplied by filter coefficients (ak and bk). M and N are the numerator and denominator polynomial orders (Eq 3). The filter coefficients determine the transfer function characteristics. By changing these coefficients in the VME download, transfer functions can be updated in real-time. Both infinite and finite impulse response (IIR and FIR) filters can be implemented. IIR filters tend to be more memory efficient (fewer coefficients) but lack the guaranteed stability of FIR filters. Although Equation 4 is an accurate description of the necessary filter algorithm, it is not necessarily the most computationally efficient. The actual filter code must also take into account finite register length, signal scaling, and truncation effects. Filters may also be used for purposes beyond bandwidth modeling. Speech compression, linear prediction, adaptive filters [2], may be used to enhance system performance.

There are a number of ways to approach the tone generator design. A straightfoward sine table look-up would be processing efficient but variation of the tone frequency would be difficult. Data memory is also at a premium in most DSPs. Using a sine function approximation to calculate each output is possible but tends to be inefficient with respect to processing time. A method that is both memory and processing efficient can be borrowed from analog oscillator design. A tone generator can be viewed as a feedback network with unity gain and zero phase shift at the oscillation frequency. The generalized transfer function synthesis used to design the filters can be used to design 'unstable filters' as oscillators. Second order transfer functions are sufficient to provide adequate Total Harmonic Distortion (THD). Variation of the filter coefficients adjusts the tone's frequency and amplitude.

Digital noise generation is equivalent to random number generation used in many computer systems. There are a number pseudo-random number sequence generators that are suitable for DSP coding. These typically have uniform probability density functions, but may be converted to others by use of an appropriate transform [3]. Since the noise in most communication systems is the result of a number of independently contributing sources, the resulting probability density is very close to Gaussian. This same process can be used for the simulation. By using a computationally efficient uniform random number generator and adding a number of its output together, a Gaussian distribution (by Central Limit Theorem [3], [4]) will result. Fidelity to actual equipment can also be increased by adding tones to the noise at points where analysis of the real equipment reveals strong frequency components. A 400 Hz component is common in aircraft systems because of the AC power in nearby instruments.

A number of desirable functions rely on the instantaneous RMS value of the input signal. VOX circuits switch the channel gain between zero and some fixed value depending on the RMS value of the input. AGC circuits vary the channel gain to maintain a constant output RMS level in response to changes of the input level. The RMS value of a signal over its past N samples is defined as:

$$\mathrm{RMS} = \sqrt{\frac{1}{N}\sum_{k=1}^{N} X_k^2} \qquad (5)$$

Even though the indicated sum of squares is in the multiply/accumulate form desired by DSPs, this calculation can consume an enormous amount of processing time if N is large. Fortunately, it is not necessary to calculate the entire sum every sample period. If a running total is maintained then each sample period the current sample can be added and the Nth oldest subtracted from the total. The square root and divide by N operations are difficult to perform on a DSP. Modifying the function's algorithm to depend on the Sum of Squares (SOS) of the input is typically easier than trying to approximate the operations. VOX simulation can be implemented by taking the calculated SOS and comparing it to a downloaded SOS threshold. If it is above the threshold the signal is passed unmodified. If not it is zeroed. A digital VOX simulation is preferable to an actual VOX circuit in the analog signal path since its parameters can be varied (threshold level and hysteresis, attack and decay timing, etc.) by the host computer. AGC circuits can be simulated similarly to VOX circuits but only within certain limits. At low signal levels there are fewer bits used to encode each sample. Scaling the channel gain to boost the signal level can result in noticeable signal to noise ratio degradation.

COST

The comparison of a four person analog trainer communication system with a four channel Digital Comm/Nav Audio System is depicted in Figure 3. The one Digital Audio Processor board functionally replaces the 13 Comm/Nav audio processor boards and the 4 I/O boards used to drive them. Because of the level and speed of the technology used on the Digital board it is considerable more expensive to manufacture than any single analog Comm/Nav board. But since so few are needed in a system, the digital system is much more cost effective. Reflectone expects to save 80% of the audio hardware costs on each trainer and 50% of the engineering design time. Further savings in the form of increased reliability and maintainability are also expected.

Conclusion

The single chip Digital Signal Processor has brought the promise of real-time signal processing into domain of small, cost conscience companies. The nature of communication systems makes them ideally suited to DSP development. The resulting cost-effective systems out perform their analog counterparts.

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Author Biography

Kevin Cahill received his BSEE degree from the University of South Florida in 1986 and is currently pursuing a MSEE in Communications and Digital Signal Processing. He joined Reflectone's Electronic Design Group in 1987 and has developed several Comm/Nav system designs for OFTs. For the past year he has been dedicated to the development of the Digital Comm/Nav Audio System for IR&D. He is also a member of the IEEE Acoustics, Speech, and Signal Processing Society.