A MICROPROCESSOR-BASED DIGITAL VOICE NETWORK

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ABSTRACT

The Digital Voice Network project is a 1984 IR&D program within the Microelectronic Systems Division of the Hughes Aircraft Company. The project is intended to advance the state-of-the-art in digital voice technology and demonstrate digital voice transmission using advanced microprocessor technology and token passing bus network architecture. This paper discusses the Digital Voice Network design architecture, voice terminal design and implementation, and finally future plans to satisfy digital voice requirements in a military environment.

Key words: Digital Voice, Distributed Processing, Token Passing Bus

INTRODUCTION

The field of communications continues to grow faster than any other field of technology. Future avionics systems will employ distributed processing in a move away from central computing systems; this will offer greatly improved reliability and increased fault-tolerant performance. With distributed processing, the microprocessor will permit merging computing and information transmission to allow dissimilar devices to communicate over a common transmission medium. Additionally, the microprocessor will be able to interrupt its current task to perform tasks left undone by other microprocessors. Distributed processing will also enable improved reliability in communication and control, a priority for tomorrow's military communication systems. Today, systems merge data with voice. When the bandwidth is available, video data surely will be added. This growth produces new system design problems in accessing networks and in efficiently processing data and voice.

One significant area affected by the move to distributed processing is that of digital audio distribution systems. This type of system offers significant technical advantage over the current analog centrally-switched intercommunications system. The distributed control

tolerates single-point failures and permits smaller and lighter packages, an extremely important requirement within the military. Operator workload is reduced by software control of the communication plan. Digital systems will take advantage of the new technology such as VLSI and VHSIC to decrease size and improve performance. Improvements in audio intelligibility and reduced ambient noise penetration can be achieved. Finally, a greater degree of security can be incorporated, both to thwart the interception of exterior communications by hostile forces and to isolate communications between terminals, ensuring the privacy of the commander's communication channel.

The microprocessor-based distributed Digital Voice Network (DVN) IR&D project, the subject of this paper, aims to address and satisfy some of the topics mentioned above.

DIGITAL VOICE NETWORK OBJECTIVES

There will be many challenges facing military voice communications in the coming years. The present and future efforts at Hughes are aimed at addressing some of the issues. Present Digital Voice Network IR&D efforts include the following:

1. Demonstration of digital voice transmission over a high-speed token passing bus. This includes both point-to-point communication and voice conferencing.

2. Development of an economical voice network, one that can be shown to be affordable and can achieve low life cycle costs.

3. Development of a test bed for future voice network enhancements that include:
    A. Noise reduction to improve audio intelligibility.

    B. Voice command recognition and voice synthesis to reduce operator workload.

    C. Network redundancy to improve system reliability.

    D. Network security to protect against undetected access.

NETWORK DESIGN

The network is configured as shown in Figure 1. Each of the three terminals is capable of sampling audio from four sources and providing two audio outputs. The input samples are digitized and assembled into packets which are then burst out onto a 10 MBps serial coaxial bus. A bus structure using a token passing protocol was selected for this application since it is highly deterministic, inherently collision free, and immune to centralized failures. In this protocol, a message token is passed from node to node on the

bus. Only that node whose unique address matches the token can transmit its message along with an incremented token. Thus, token passing provides a predictable network; guaranteed maximum access times can be calculated.

TERMINAL DESIGN

A. Hardware Implementation

The DVN Terminal block diagram is shown in Figure 2. The Bus Multiplexer (BMUX) provides the interface to the fast bus, satisfying all protocol requirements. It contains an internal 8086 microprocessor and sufficient memory to buffer large incoming and outgoing messages and is capable of accepting incoming messages at the full bus rate. The BMUX has been designed to be compatible with the Intel Multibus standard.

The Processor provides control of the terminal. It stores messages from the BMUX, performing all required processing such as conferencing and volume control, and sends the data to the Audio board when requested. In addition, it accepts digitized audio data from the Audio board when available and sends it to the BMUX. Finally, the Processor is responsible for the control and monitoring of the Entry/Display Terminal. The Processor consists of an 80286 microprocessor, memory and associated circuitry on a single Multibus compatible card. Two serial RS232 ports are supported by the card, one of which is dedicated to the Entry/Display Terminal.

The Entry/Display Terminal allows for operator control and monitoring of the Digital Voice Terminal and the Network. Connectivity and volume control can be accomplished, and terminal/network status can be monitored.

The function of the Audio board is to digitize samples of four audio inputs which are then transferred to the Processor or the BMUX via the Multibus interface. In addition, the board accepts digital data from the Multibus, converts it to analog and provides two audio outputs. The board is capable of operating at selectable sample rates, with all channels operating simultaneously. The four audio inputs are filtered, then applied to sample and hold circuits and multiplexed into a single analog to digital converter. The output of the converter is loaded into an onboard RAM and, after a selectable number of samples have been written, the contents are read by the Processor via the Multibus. Simultaneously, digital data from the Multibus is loaded into another onboard RAM, then sequentially converted to analog, demultiplexed and filtered to provide two audio outputs.

B. Software Implementation

The software for the Digital Voice Network terminal has been written in PL/M and compiled into 80286 object code for execution. The software is modular, with each module's entry and exit conditions being defined at the top level to ensure compatibility and expandability. The DVN top level software implementation, shown in Figure 3, illustrates the major modules and their relationships. The Kernel provides flow control and interrupt handling routines. Error reporting and exception handling modules reside within the Kernel. During power up, reset or restart, this module provides initialization sequencing and creates a table of resident Audio boards. The Kernel controls all system timers and timer interrupt vectors. Finally, the Kernel is responsible for periodically transmitting the connectivity data and other status of the terminal to the other terminals on the fast bus for local display.

BMUX Control handles the initialization of the BMUX and any commands directed to buffer creation and deletion, generation of Input Control Blocks and Output Control Blocks, and any required commands for processing data.

The Voice Processing Module controls the interface to the Audio board and handles all functions related to packet switching overhead, such as bits/sample adjustment, volume control, and channel conferencing.

The Operator Interface Module provides Control and monitoring of the Entry/Display Terminal. This allows operator inputs to control the communication paths, volume control, and the BMUX through the command screen. The terminal's monitor displays the system screen upon power up, which allows the operator to configure the terminal. The tables generated from this information are used by the other software modules.

SYSTEM EVALUATION

System evaluation is being conducted to verify expected performance of the Digital Voice Network to demonstrate both point-to-point communication and voice conferencing. First, two terminals will be configured as a two-way communications network. Performance will be evaluated in the area of microprocessor utilization to determine how much of the processor's capacity is being used for the tasks required. Voice intelligibility will be assessed using signal-to-noise measurements and Modified Rhyme tests.

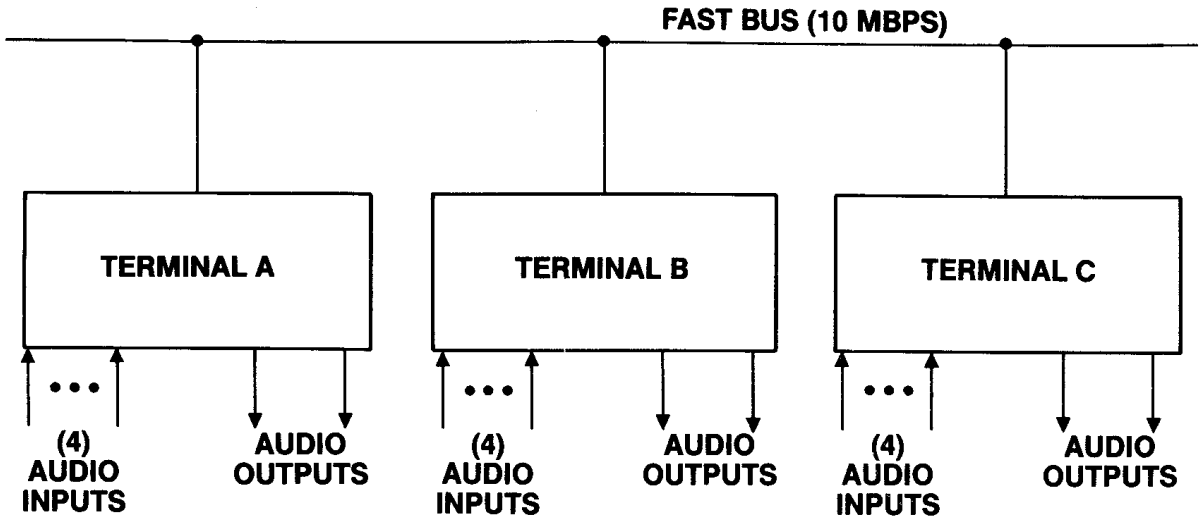
Once two-terminal communications have been established, a third terminal will be added to evaluate voice conferencing performance. This third terminal will differ from the other two in that it will simulate up to 30 terminals with respect to bus loading. Channel capacity performance evaluation will be demonstrated in two ways. First, the system's performance

under a worst-case frame time will be evaluated by simulating a large number of terminals on the bus. Second, the system's performance under conditions of a worst-case terminal workload will be evaluated by directing additional messages to one of the two terminals. Network performance and audio intelligibility under this fully-loaded condition will be assessed.

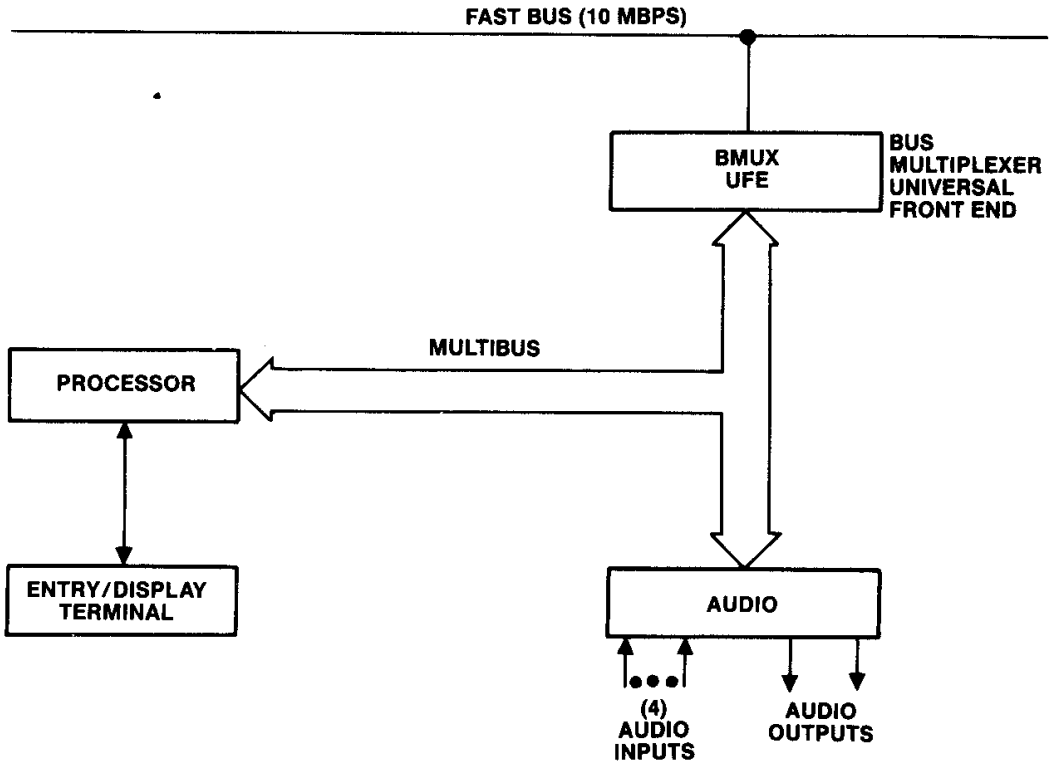
CONCLUSION

This IR&D effort is being performed with two major goals in mind: 1) to demonstrate the transmission of digital voice over a high-speed token passing bus and 2) to provide the ability to research key voice technology areas crucial to satisfying the future voice requirements of the military.

The test bed that we have developed will allow for implementing enhancements in the voice terminal. Audio inputs will be pre-processed with a noise reduction algorithm by means of a dedicated processor. Algorithms will be generated for voice command recognition. These will be incorporated into the terminal to provide operator control of terminal functions. A dual-redundant bus structure will provide fault-tolerant, graceful degradation of network functions in the event of bus failure. Finally, hardware and software will be implemented to isolate secure from clear voice/data on a shared bus. These and other enhancements will be researched and then incorporated into the terminal in an attempt to satisfy digital voice requirements in a military environment.

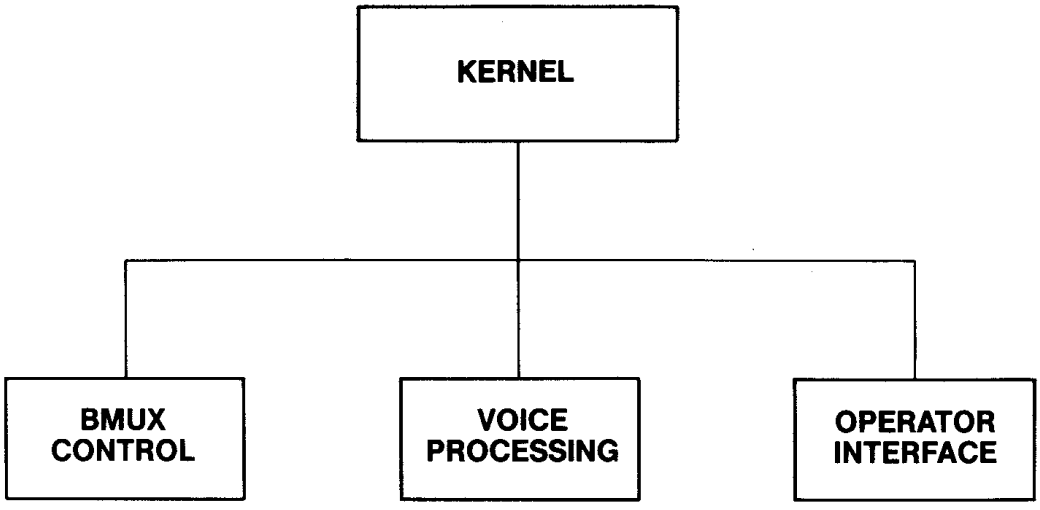


NETWORK CONFIGURATION
FIGURE 1



DVN TERMINAL BLOCK DIAGRAM

FIGURE 2



DVN TOP LEVEL SOFTWARE IMPLEMENTATION
FIGURE 3