Solid-State Storage-Based Telephone Answering Machine Implementation With SX Microcontroller

Application Note 17

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1.0 Overview

In recent years, there has been a surge in telephone answering machines that store messages in solid-state storage devices. The current generation of telephone answering devices (TAD) uses digital signal processors (DSP) to compress digitized voice so that it can be stored in conventional DRAM or FLASH memory. This approach requires complicated compression algorithms and multiple components.

This application note describes a new approach that replaces DSP chips with a high-performance 8-bit microcontroller from Scenix Semiconductor and multi-level memory from Invox Technology. Telephony functions, such as Caller ID, DTMF detection, Bell 202 detection and Ring detection, are implemented in software in the microcontroller. Answering machine-specific functions, such as message storing, DTMF generation and outgoing message storage are implemented in analog storage. The main advantage of this design, based on a configurable 8-bit microcontroller and analog storage, is to reduce cost and complexity and enhance voice quality. This design greatly simplifies the development effort by eliminating a relatively expensive DSP device without sacrificing the quality of the end product.

2.0 Basic TAD Blocks

Figure 2-1 shows the main blocks for the TAD. This section describes the functions of each block in detail. The solution should include lightning and surge protection, reduction of the 90-volt ring signal, and 48 volt off line signal and 6 volt on-line signal.

2.1 ANALOG FRONT END

All telephone answering machines include an analog front end to pre-process and condition incoming and outgoing voice messages. The complexity of the front end is determined by the physical layer connection to the outside world. The main job of the analog front end is to condition the higher line voltage used in the telephone line so that they are compatible with the low-level signals around the integrated circuits.

2.2 RING DETECT

In a cell phone the master processing unit with the phone or base station itself generates ring signals. This logic signal can be fed to the main processor of the TAD to initiate the TAD functions. In a land line based telephone answering machine, the ring signal is produced at the central office and is composed of a 90-volt rms 20 Hz signal that is impressed across the telephone line. In this design, this ring signal is detected by the microcontroller.

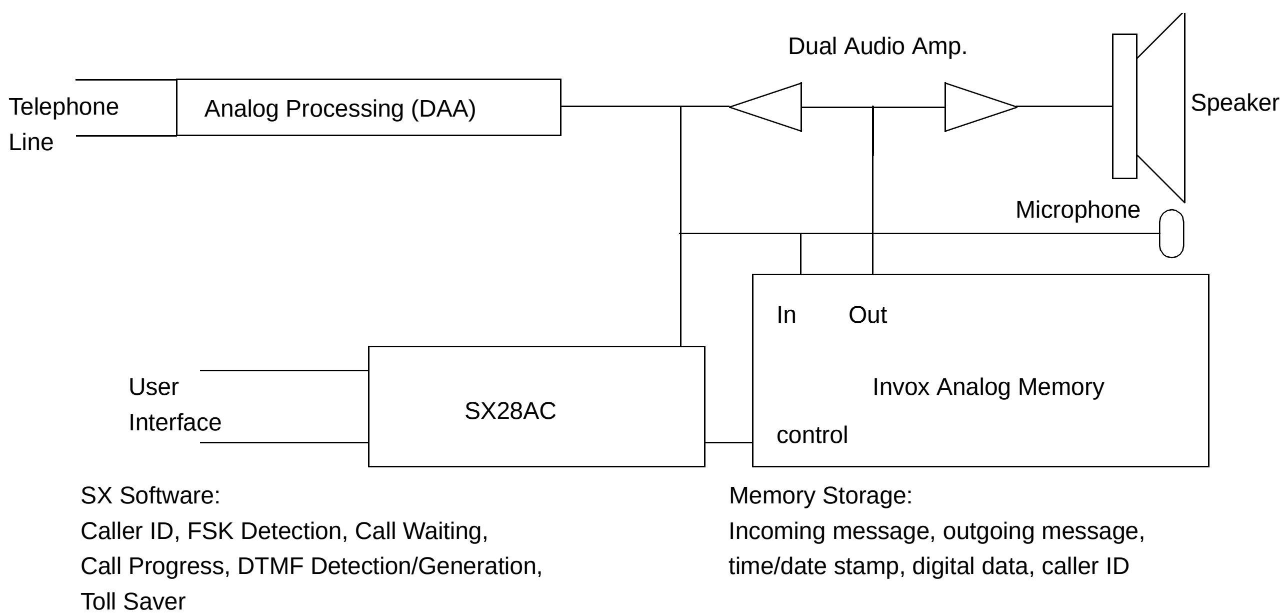


Figure 2-1. Basic TAD Blocks

2.3 MASTER CONTROL

Master control refers to message management, call management, and memory management as well as the user interface. In this design the same microcontroller is used to do the master control**.**

2.4 DTMF DETECTION AND GENERATION:

All modern answering machines must be able to detect DTMF tones in order to allow remote access. Some also must be able to generate DTMF tones to enable dial out and forwarding the messages. In this design, the DTMF generation and detection is implemented in software by the same microcontroller. DTMF tones can also be pre-stored in the analog memory chip and played back onto the telephone line under the control of the microcontroller unit.

2.5 CALLER ID

In United States, Caller ID function requires FSK (Frequency Shift Keying) data between the first and the second ring to be detected and demodulated. This data is then processed so that caller information can be displayed and recorded. In this design, Caller ID function is implemented by the SX device in software. The Caller ID information can be stored in the analog memory.

2.6 OUTGOING MESSAGES/VOICE PROMPTS

The same analog memory used to record incoming messages can be used to record the outgoing message. The SX device used for the master control accomplishes this task by reserving certain sectors within the analog storage array for the outgoing message. Some answering machines also include the built in prompts such as time and date stamps. These prompts can be factory programmed into the analog memory in sectors that are write protected.

2.7 CALL PROGRESS DETECTION

Call progress detection involves sensing of the line voltages and off-hook tones. This also can be done by the microcontroller in software.

3.0 Design Implementation

3.1 PROCESSOR UNIT

In the past, telephony functions, such as, FSK (frequency-shift key) data generation and detection, DTMF (dual-tone, multi-frequency) dialing generation and detection, and Caller ID could not be implemented with an 8-bit embedded microcontroller because performance levels were not high enough to support them. As a result, either DSP or ASIC (application specific IC) had to be designed or a 16 or 32-bit microcontroller were used. Now, the 8-bit Scenix Semiconductor SX Series microcontroller, which has performance of 100 MIPS (million instructions per second) and a deterministic interrupt architecture, overcome this challenge by providing the ability to perform these functions in software.

Unlike other microcontrollers that add functions in the form of additional silicon, the SX Series uses its industry-leading performance to execute functions as software modules, or Virtual Peripheral™ modules. These modules are loaded into a high-speed (10 ns access time) on-chip flash/EEPROM program memory and executed as required. In addition, a set of on-chip hardware peripherals is available to perform operations that cannot readily be done in software, such as, comparators, oscillators etc. The Virtual Peripheral modules used in this design are: 16-bit timers, DTMF detection/generation, FSK detection, Caller ID, Call waiting, Call progress and LCD drive.

To achieve the lowest implementation cost, the design uses discrete components, rather than a module for the DAA (Digital Access Arrangement) block. An opto-isolator and a transformer provide coupling to the telephone network that complies with the Bell 202 standard. The cost of the components in such a circuit is significantly less than that of a DAA module.

Table 3-1. Virtual Peripheral Memory Utilization

	DTMF Detect	FSK Detect	Ring Detect	Caller ID
Program Memory (Words)	394	78	32	119
Data Memory (Bytes)	45	16	4	32

3.1.1 DTMF and FSK Detection

The DTMF signal is applied to the SX28AC I/O pin after it passes through the analog-processing block. The MCU converts this from analog to digital by sampling it in software using the Goertzel discrete Fourier transform (DFT) algorithm. Eight separate DFTs simultaneously sample the signal looking for the row and column frequencies that identify digits within the DTMF matrix. To be properly selective on the target frequencies (within 5 Hz), a performance level of at least 50 MIPS is required. In addition, the standard for the DTMF tone duration allows it to be as short as 50 milliseconds (ms), which the SX28AC can easily handle. Most other 8-bit MCU implementations require 150 ms or more to detect the tone.

The FSK signal from the analog-processing block is applied to the I/O pin of the SX28AC. Detection of the FSK data for the caller id (1300 Hz representing a logic 1 and 2100 Hz representing a logic 0) and its conversion to a digital format is done by an on-chip hardware comparator using a form of zero-crossing detection. The comparator is one of the basic set of silicon peripherals included in the SX28AC. Table 1 shows the actual program memory and data memory sizes required to implement the Virtual Peripheral modules in software.

3.2 RECORDING AND PLAYBACK DEVICE

Non-volatile multi-level flash memory from Invox Technology is used as the recording and playback device in the design. Up to thirty minutes of audio recording and playback can be accommodated using an IVM1700 device. Invox Technology devices can be cascaded for longer duration of recording or greater digital storage. Invox Technology devices utilize multi-level analog recording to store voice signals. This allows the device to reproduce audio signals in their natural form, eliminating the need for encoding and compression; which introduce distortion. The smallest addressable memory unit is called a “sector”. Each sector is composed of 3008 memory cells. During audio recording one memory cell is used per sample clock cycle. IVM1700 devices allow the sampling rate to be adjusted from 4 to 8 kHz. When recording is stopped an end of data (EOD) bit is programmed into the memory. This prevents play back of silence when recording is stopped before the end of a sector. During playback, the stored signals are retrieved from memory, smoothed to form a continuous signal and finally amplified before being fed to an external speaker or the telephone line. All memory management is handled by the SX microcontroller.

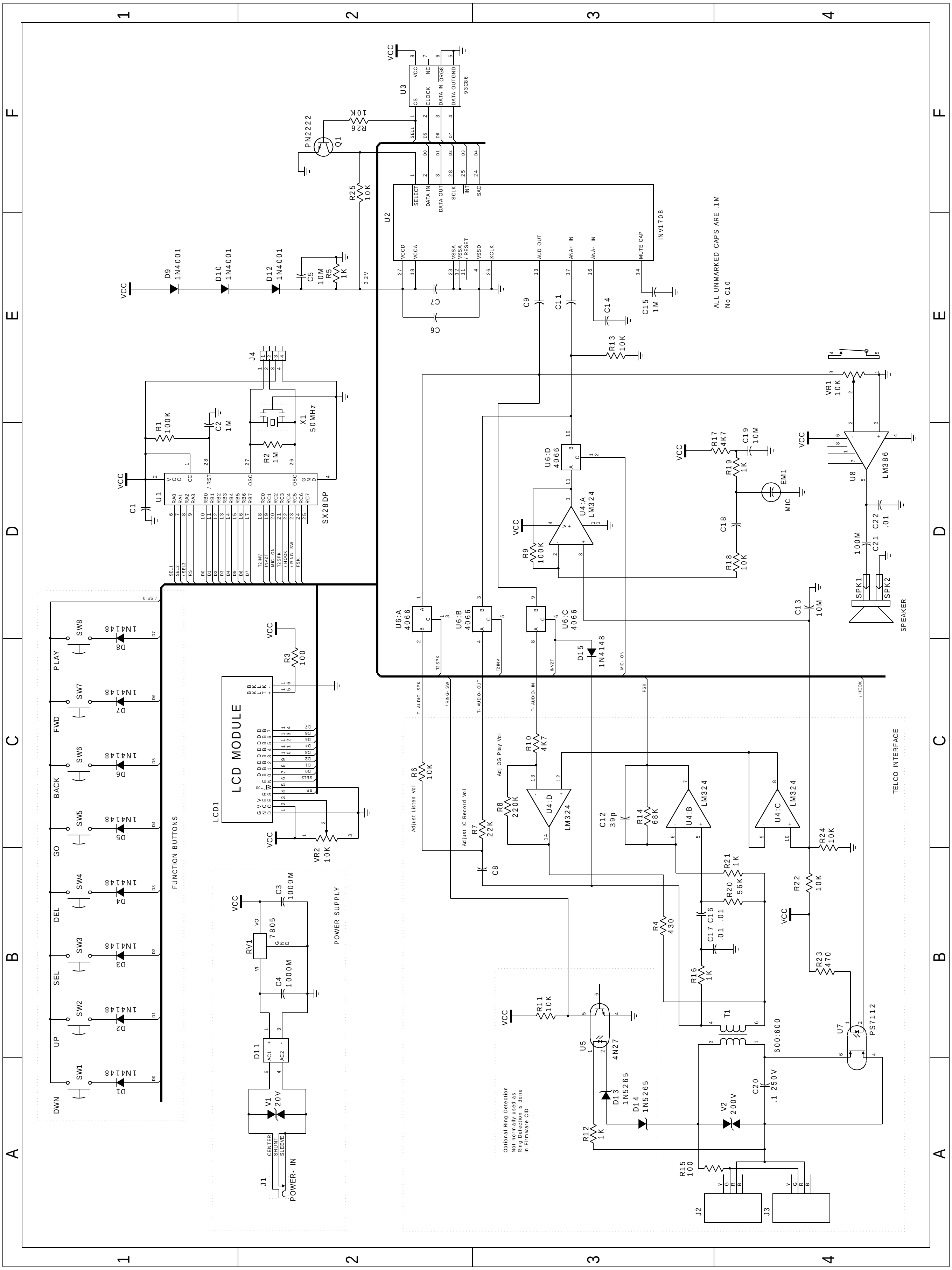
The SX device communicates with the IVM1708 through a simple Serial Peripheral Interface (SPI) port. The SPI port can run on as little as three wires or as many as seven depending on the amount of control necessary. The IVM1700 devices also provide digital storage capability. Digital storage is used to accommodate storage of caller ID information as well as time and date stamps.

4.0 Conclusions

Since majority of TAD functions are performed by the SX device in software, this design eliminates the need for separate ICs for DTMF detection, ring detection, Caller ID etc., thus providing savings in board area and cost. This design greatly simplifies the development effort by eliminating a relatively expensive DSP without sacrificing the quality of the end product.

5.0 Reference

- Jack Quinn, ‘Digital Data Communication’, Prentice Hall, 1995.
- Scenix Semiconductor Inc., ‘Modem Reference Design Using the 50 MHz SX28AC 8 bit MCU’, 1998.
- Invox Technology, ‘TAD design based on the IVM1700 family’, Application note, 1998.



INV1708
SX28DP
LCD MODULE
93C86
FUNCTION BUTTONS
POWER SUPPLY
TELCO INTERFACE
SPEAKER
ALL UNMARKED CAPS ARE .1M
No C10

Lit #: SXL-AN17-01

Sales and Tech Support Contact Information

For the latest contact and support information on SX devices, please visit the Scenix Semiconductor website at www.scenix.com. The site contains technical literature, local sales contacts, tech support and many other features.

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