PCT-AN Product Brief

Introduction

PCT series are PCI form factor Computer Telephony cards from SST. This range of cards are ideally suitable for providing cost effective Computer Telephony solutions (IVRS, Predictive dialers, etc.). PCT-AN-4 is a 4-port Analog Loop Start Telephony Interface card. PCT-AN-2 has 2 ports. Cards can also be supplied with 3 ports. If more than 4 ports are required multiple cards can be plugged into the same system. SST has the technology to build 8-port card depending on market demand.

Features of the Architecture

All PCT series telephony cards use similar architecture. All are based on PCI bus. They support both 5V signaling and 3.3V signaling. All the cards have product IDs and Serial numbers embedded in them, which can be used by integrators to lock their software.

PCT cards use Programmable DSPs. The number of DSPs on-board is decided based on the current and future processing requirements. Having programmable DSPs on-board helps to add new features at a later date without modifying the hardware. This provides investment protection for the customers.

The Firmware running on the DSP is responsible for all the interface and protocol related functions. Firmware upgrade is as simple as installing the new software and restarting the application. So, the card need not be sent to the service center for Firmware upgrades.

Block Diagram

PCT-AN-4 has four 2-wire analog telephone line interfaces (PCT-AN-2 has two), a four-channel CODEC, a programmable 16 bit fixed point DSP, a PCI interface IC and an EEPROM.

Each of the four Line Interfaces supports,

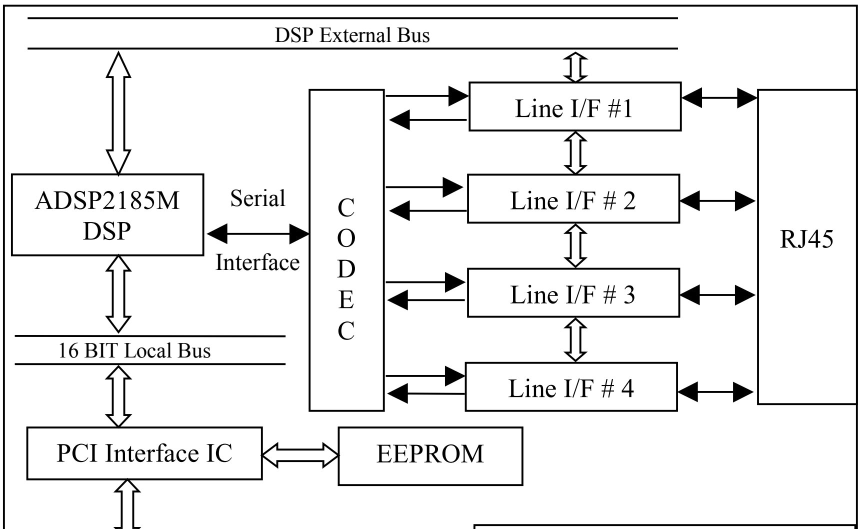
- Receiving and sending of analog voice signals
- Ring detection
- Hook switch control
- On-hook reception

and other functions required by the switch. The four-channel CODEC converts the analog signals into digital data and pass it to the DSP.

There is one fixed point DSP on-board and it is responsible for

- Transferring the speech data from the CODEC to the host CPU memory during recording
- Transferring speech data from the host CPU memory to the CODEC during play
- Detecting DTMF and Pulse digits, Caller ID information (FSK and DTMF), Call Progress Tones and other tones from the data received from CODEC
- Pulse and Tone dialing
- And passing the status of the telephony port to the application

The PCI Interface IC is responsible for interfacing the DSP to the PCI bus. The PCI configuration details are stored in an EEPROM.



Block Diagram of PCT-AN-4

Features

- Supports WIN NT 4.0, WIN 2000 and WIN XP Operating Systems
- Half-size PCI Plug-in board
- Network Interface: Analog Loop Start Interface
- Supports popular APIs for call processing (TAPI) and Media interfacing (WAVE API). Applications written using TAPI and WAVE API can be migrated to this board very easily
- Maximum of four analog Telephony ports per card for PCT-AN-4 and two in case of PCT-AN-2
- All ports can be used for making incoming and outgoing calls
- Supports pulse and Tone (DTMF) dialing
- Supports Call Progress Tone Detection. An Automatic Tone Configuration utility is provided for configuring the card to detect PSTN and various EPABX tones
- Supports DTMF and FSK based Caller ID identification.
- Loop reversal Detection for called party answer detection
- Supports Positive Voice Detection
- Supports Silence Detection
- Supports DTMF digit detection and Generation after the call is established
- Supports Dial Pulse Detection and FAX/Modem Answer Tone Detection
- Supports Loop Current Drop Detection
- Supports SI (Special Information) Tone Detection
- Supports PABX related features such as call forward and call transfer
- Supports simultaneous play and record (Full Duplex) of voice signals on all ports. It supports, 8 kHz sampled A-law coding, 8 kHz sampled µ-law coding and 8 kHz sampled 16 bit linear wave formats
- Analog Interface confirms to analog loop start PSTN/EPABX network specifications.
- Multiple boards (max 4) can be supported on the same system

Technical Specification

- Number of Voice Ports: 2/3/4
- On-board Processing Power: 73 MIPS

Host Bus Requirement

- 33 MHz, PCI Ver 2.1 compliant Bus PCI Bus Master/slave card
- Universal 32-bit card (Can work in the predominantly existing 5V signaling and future 3.3V signaling environments)
- Card Size: Height: 4.2 inches, Length: 6.9 inches

OS Support

- Supports WIN NT 4.0, WIN 2000 and WIN XP

Application Development Support

- Supports TAPI 2.1 for call control
- Supports Wave API for media control

Analog Network Interface Specification

- The Analog Network Interface Confirms to analog loop start PSTN/EPABX network specifications. Some of the important specifications are given below.

AC & DC Specification

- Impedance: 600 Ohm (Nominal)
- Loop Current operating Range: 16 to 70 mA

Speech Coding

- Sampling Rate: 8000 Hz
- PCM coding: 8 bit A-law, 8 bit µ-law and 16 bit Linear

DTMF Dialing

- Digits Supported: 0 to 9, #, *, A, B, C, D (Other TAPI specified dialable characters are also supported)
- Frequency Accuracy: ± 0.1% Maximum
- Dialing Rate: 20 digits/sec to 4 digits/sec (Configurable)
- Pulse Duration: 20 msec to 200 msec (Configurable)

- Transmit Level: -3 dBm to –15 dBm (Configurable)
- Twist: 0 to ±6 dB (Configurable)

Pulse Dialing

- Digits Supported: 0 to 9
- Dialing Rate: 10 pulses per second
- Make/Break: Configurable

DTMF Detection

- Digits Supported: 0 to 9, #, *, A, B, C, D
- Frequency Accuracy tolerance: 1.5% (verified through simulation)
- Pulse Duration: 50 msec minimum (verified through simulation)
- Detection Level: up to –30 dBm (verified through simulation)
- Acceptable Twist: ±9 dB maximum (verified through simulation)

Dial Pulse Detection

- Supported

Positive Voice Detection

- Supported

Silence Detection

- Supported

SIT Detection

- Supported

Hang-up Detection

- Supports Loop Current Drop Detection and Hang-up tone detection

FAX/Modem Answer Tone Detection

- G3 FAX
- Data Modem

Caller identification

- Supports DTMF and FSK based caller ID detection

Power Supply Requirement

- +5VDC: 500 mA (Max)
- -12VDC: 60 mA (Max)

Operating Environment

- Operating Temperature: 0°C to 50°C
- Storage Temperature: -10°C to +70°C

Other Related products from SST

- PCI based E1 Network Interface board (PCT-E1-30)
- Host based Speaker independent Speech Recognition to run on this card
- Rhapsoidia Text-to-Speech system for English

Contact us

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