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EP-436 Base Station Version 0.50
Description Document

1.0 Scope

This document describes the EP-436 base station. This document also describes the connections to the outside world and the line interfaces.

2.0 Definitions

DSP1 = Digital Signal Processor #1 (handles the air interface)
DSP2 = Digital Signal Processor #2 (handles the vocoder)
uC = H8S/2134 Microcontroller
HPI = Host Port Interface
McBSP = Multi-channel Buffered Serial Port
RFM = RF Module
IF = Intermediate Frequency
AA = Auto-attendant
OGM = Outgoing Message
MOH = Music on Hold
L1 = Line Interface #1
L2 = Line Interface #2
L3 = Line Interface #3
L4 = Line Interface #4
PSTN = Public Switched Telephone Network
PBX = Private Branch Exchange
PCB = Printed Circuit Board
PBCA = Printed Circuit Board Assembly
CID = Caller Identification
RSSI = Receive Signal Strength Indicator
ATE = Automated Test Equipment

3.0 Overview

The EP-436 base station is a 4-line wireless PBX. It is capable of linking four handsets to four different PSTN lines simultaneously. (Handset to handset communication is done independent of the base.) The multiple access is achieved using TDMA. The base hardware supports the following functions:

Caller ID detection (Type 1 and Type 2)

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3-way call conferencing
Call forwarding
Auto attendant
Music on hold

The base has three processors: two DSPs and one uC. DSP #1 handles the air interface processing. It communicates with the IF undersampler (a combined A/D and D/A) through the EPLD. The IF undersampler sends 10.7 MHz IF signals to the RFM and receives 10.7 MHz IF signals from the RFM. (10.7 MHz filtering is done inside the RFM for transmit, but the filter is located on the B1 board for receive.) DSP1 also communicates with DSP2 via a serial port, for the purpose of sending and receiving voice data (compressed).

DSP #2 is responsible for voice compression and decompression (vocoder), auto attendant, auto routing, and interfacing to the four-channel codec. Auto routing involves connecting an air interface time slot to the appropriate four channel codec time slot (and therefore to the appropriate line interface). Voice data from the codec must be compressed by the vocoder before being sent to DSP1. Voice data from DSP1 (the air interface) must be decompressed before being sent to the codec. The four-channel codec is connected to DSP2 by a full duplex serial port. The analog side of the codec is connected to the four PSTN lines.

The uC tracks the state of the base (i. e. who is talking to whom). It tells DSP2 which air interface time slot needs to be routed to which PSTN line, but DSP2 does the actual routing of the digital voice data. The uC also keeps track of system configuration data in the EEPROM, and processes the CID data from all four CID chips (one for each line). The uC monitors each line for ring detect and hang-up detect. Also, it controls the off-hook relays and the MOH analog multiplexers, allowing the uC to pick up any line or to put any line on hold.

4.0 Electrical Interface

The base station has the following connections to the outside world:

DC power input jack
RJ11 PSTN dual line input jack (2)
Antenna connector
Music on hold input jack
Momentary push-button switches (2)
LEDs (5)

The DC power input shall be 7.0 V nominal. The rated current at 7.0 V shall be 1.0 amps. It will accept the plug from the Engenius supplied (UL approved) AC-DC wall adapter. It is preferable

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that the wall adapter shall be a linear regulator, not a switching regulator (for improved noise performance). Filtering and hold-up capacitors are provided inside the base station.

The line input jacks are implemented by two dual modular jacks. Each dual modular jack contains two RJ11. The first dual modular jack is for L1 and L2. The RJ11 jacks have the following configuration:

Pin Number	Standard Color of Phone wire in plug	Function
1	Unused	Unused
2	Unused	Unused
3	Red	Ring (line 2)
4	Green	Tip (line 2)
5	Unused	Unused
6	Unused	Unused

L3 and L4 have the same configuration on the second dual modular jack as is detailed above for L1 and L2 on the first dual modular jack.

Also, a standard telephone can be plugged into the RJ11 jack, but it won't have access to base station PBX features.

Each line interface shall meet the requirements or FCC part 68 and EIA-470B. The line interface will present a balanced 600 ohm load (roughly) to the phone line in "off hook" condition. Ring detect and caller ID detect (Type 1 and Type 2) will be performed on the PSTN side of the line interface. There will be adequate isolation to meet regulatory requirements. DC loop current is limited to less than 100 mA.

5.0 Mechanical

The base station has two PCBAs and an RFM. The B1 PCBA contains most of circuitry. The RFM is mounted to the B1 PCBA and connected to it by a 20-pin connector. The B2 PCBA only contains 5 LEDs and is connected to the B1 PBCA through a cable to a six pin connector. Surface mount components were used as much as possible, and all of the components are placed on the top side of the B1 board. Most test points are accessible on the top and bottom side of the B1 board. The B1 board is approximately 9.8 in X 6.8 in and is a six layer board (top, bottom, two layers of internal signal routing, one power plane, and one ground plane). Maximum component height on the top of the PCB is 20 mm. Maximum allowable component height on the bottom is 3.1 mm. The PCB is laid out so that there is maximum distance between RF antenna and the line interfaces. There are provisions for a shielding can around the codec and audio circuitry, the line interfaces, and the digital circuitry. The inside of the plastic enclosure shall be coated with conductive paint.

The footprint for the memory device that contains default OGMs and DSP2 program code can accommodate a socket or the device itself. The footprint for the uC can accommodate a socket or the device itself. There are provisions for a flash memory device (with or without a socket) for DSP1. JTAG connectors (14 pin headers) are provided for each DSP, and a 6-pin header is provided for uC flash programming capability.

6.0 Line Interfaces

The line interface is based on the Philips TEA1062 line interface IC. The design is based on the recommendations from their data sheet. Each line interface circuit contains an off-hook relay, a 2-wire to 4-wire converter, and two isolation transformers. Each of the four line interface circuits is identical. Each line interface also contains a ring detect circuit and a hang-up detect circuit.

The PSTN side of the line interface (before the relay) contains a PTC and 300V MOV for protection.

After the relay, there is a diode bridge rectifier and the 2-wire to 4-wire converter. The 2-wire to 4-wire converter is based on the Philips TEA1062. The 4-wire side of the 2-4 wire converter is connected through isolation transformers to the analog inputs and outputs on the four-channel codec.

Each line also has a “hang-up” detect circuit. This circuit simply detects whether the VCC power is on for the TEA1062. The TEA1062 is on when the off-hook relay is on and the central office has connected the line (closed the loop). This causes the hang-up detect circuit output to be asserted low. When the loop is opened (either by the PSTN central office or by opening the relay), the output is asserted high. This feature is primarily useful for terminating “call-forwarding” calls (one line talking to another without a handset involved), because the central office opens the tip line for at least 100 ms when the person on the outside line hangs up. The hang-up detect output for each line interface is connected to a uC GPIO pin.

Caller ID detection circuitry is based on the Mitel MT88E43 (one for each line).

8.1 Music on Hold

The music on hold (MOH) analog signal is a differential signal that is routed through an isolation transformer, RF filtering, and an attenuator. The attenuated signal is connected to a differential amplifier, which improves common mode rejection and converts the differential signal to a single ended signal. The attenuator is adjustible, but the differential amplifier has a fixed gain of 2. The amplifier output is routed to the inputs of the analog multiplexers (two 74HCT4053s). The uC can command these multiplexers to put the L1, L2, L3, or L4 caller on hold, which would directly connect the MOH analog signal to the appropriate line. If the line is not on hold, the analog output of the four-channel codec for that line is routed through the analog multiplexer to the appropriate line interface.