the charge contact. Protection from ESD is afforded by a bypass capacitor installed at the charge contact.

When the battery voltage drops below the minimum working voltage of the MCU, the phone will not function properly again if the MCU is not properly reset. Therefore, circuits have been implemented to insure that the battery has sufficient charge for proper operation. The reset pin of MCU will be released till the battery voltage charge back to the min. operating voltage.

About the function of low voltage detection, the RF chip can provide an internal reference to which battery charge is compared and the detection voltage are programmable by MCU. In VT9108 MK4, two level of voltage will be detected. If the battery voltage drops below 3.3 VDC, the low battery detect pin in chip will be activated to inform the MCU and causes the MCU to notify the user by producing an audible tone and flashing the LED. Then, MCU will change the detection voltage to 2.9V to noticfy the MCU stop all the operation before the voltage drop below the operation voltage of the RF chip. The detection voltage will change back to 3.3V once the on-cradle signal detected. The typical hysteresis of the comparator is 18mV.

2.3 Audio Circuits

Audio circuits are necessary to condition speech for RF transmission and reception. The conditioning includes amplification, filtering, pre-emphasis / de-emphasis and compression / expansion, all of which ensures that the speech is received and transmitted with maximum clarity and legibility.

Pre-emphasis/de-emphasis is used to improve signal-to-noise ratio (SNR) which is, as a consequence of frequency or phase modulation, degraded at high audio frequencies. Compression/expansion is also used to improve the perceived SNR by reducing the noise vulnerability of low-level signals. The compression process amplifies low level signals more than it does for high level signals. Thus, by compressing the dynamic range of the audio before transmission, noise picked up during transmission has less of an effect on the low-level signals.

After receiving the transmission, the expansion process maintains this improved SNR while restoring the low level signals back to their original levels.

The audio circuits are implemented by using the compressor and expander inside the combo chip. It provides compression/expansion, amplification and muting all in a clean, simple way.

2.3.1 Base Unit

The audio circuits in the base provide for speech exchange between the telephone line interface and the RF module that communicates with the handset. Figure 1 shows the circuitry for audio processing in the base unit.

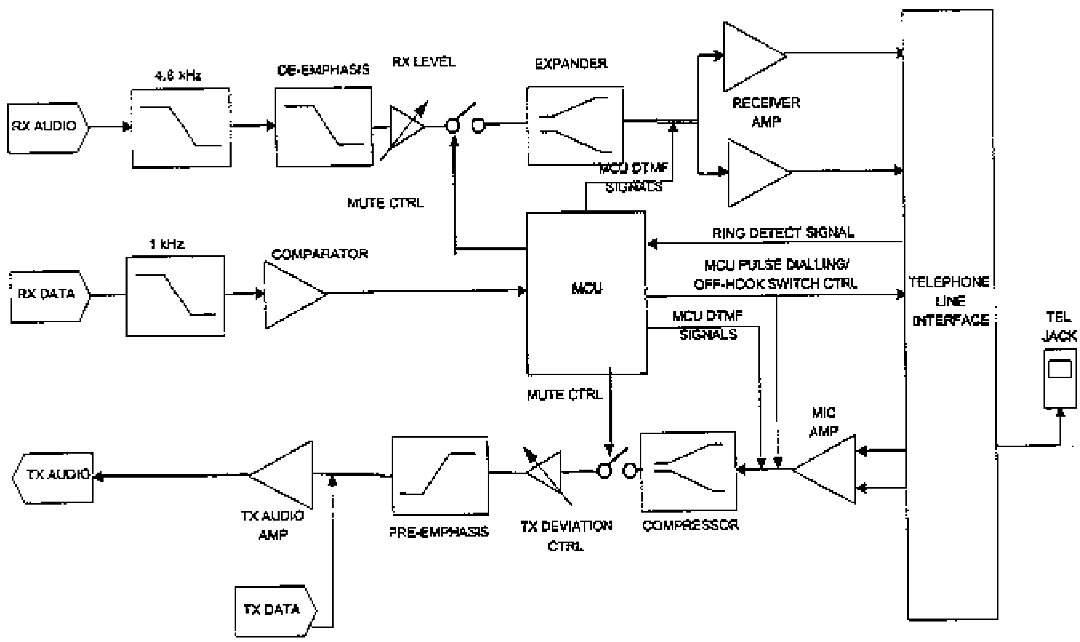


Figure 1. Audio Circuits in Base Unit

2.3.1.1 Transmit Direction (RF module to telephone line)

The transmit audio is transmitted from the handset to the base using frequency modulation (FM). The FM signal from the handset enters the base RF module, where the signal undergoes filtering, down-conversion, and finally demodulation. The baseband audio then leaves the RF module via the demodulator (for a deviation of ±25 kHz).

The DETO pin is connected to a passive low-pass filter with a -3 dB cut-off set at about 4.8 kHz. The filtered audio is then passed to an de-emphasis filter where the de-emphasis occurs across the entire audio band (300 Hz to 3400 Hz) at a rate of 6 dB/octave or 20 dB/decade. After de-emphasis, the transmit audio signal is proceeded through level control and the expansion process. The level control,

Level	SPK_OUT(dB)
1	0/Mute
2	3.5
3	7.0
4	10.3
[redacted]	

For other products, in order to make the volume levels be the same as the speakerphone. The volume levels are set as below:

[redacted]

4.8.4 Line Playback :

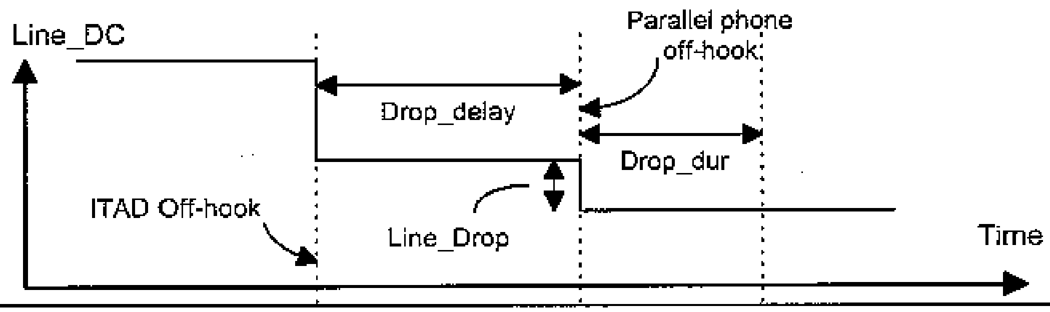
In the application circuit, the gain from LIFPOUT to T&R is about 0dB. When LIFPOUT is at –2dBm (600mVrms), T&R level is also –2dBm. The design of ITAD module for LIFPOUT output level is derived from: when the ALC goes to the "saturated" region and this level is recorded to the DSP, the corresponding LIFPOUT will be –2dBm (which is the maximum output level).
Under normal line condition, most of time ALC is "saturated" by the voices of the line and hence ICM line playback level will be around –2dBm

4.8.5 ALC & DTMF Detection During Line Playback :

In line playback mode, DSP's internal Line Echo Canceller (LEC) is on to cancel out the sidetone feedback for DTMF detection. At the same time, to make LEC operate normally, ALC cannot be implemented. **Therefore the current design uses another line input LIFMIN2 to by-pass the external ALC. During line playback, the DSP will monitor the line signal through LIFMIN2 to skip the external ALC circuit. The maximum allowable input level at this pin is** [redacted]**. Also if the gain from T&R to LIFMIN2 is changed (now is –8dB), the various thresholds** DFD_play, VOX_play & CPD_play **must be adjusted.**

4.9 Parallel-phone Detection using LINE_DC:

The parallel phone detection mechanism can use the internal A/D inside the DSP chip in both Mode 1 & 2. It will monitor the T&R DC conditions and goes on-hook if parallel phone is detected. The diagram below illustrates the detection algorithm:



When ITAD goes off-hook, it will wait for *Drop_delay* before the A/D converter will read the Line_DC value and save. It will then start to compare the Line_DC with the stored initial value. This *Drop_delay*, in US models, is set to 1 second to prevent US West CO problem. If an external parallel phone is constantly picked up after *Drop_delay*, ITAD MK3 will recognize as a valid parallel phone pick-up if the following conditions are fulfilled:

The pick-up duration is long than *Drop_dur.*

The Line_DC drop is more than *Line_Drop* compared with the stored initial value.

As the levels and timing are different from the line-interface circuit and different countries, so some parameters (*Line_Drop, Drop_delay, Drop_dur)* are open for the user to tune according to his requirements. On the other hand, Line_DC circuit should be able to provide good immunity to ac signals in T&R, just like that in the application circuit. Otherwise, mis-triggering of parallel phone detection may occur.

4.10 Power Failure Detection & Flash Protection :

ITAD MK3 has a built-in power failure detection to prevent Flash corruption in case of power failure during the Flash's writing cycle. It is done by another A/D converter inside the DSP. To protect the Flash, upon detection of power failure, the DSP will reset the Flash by asserting MA4 and then stop the writing cycle by forcing itself into idle state. During power-up, MA4 is kept low so as to make the Flash being reset.

which has a range of about 15 dB, is used to set the transmit audio level at the tip and ring of the telephone line, The transmit level can vary from component tolerances and variations.

After then, the audio is passed to a final stage of amplification. The output of this amplifier is connected to telephone line interface. The output of the DTMF generator circuit is also coupled in at the input to this final stage of amplification.

The transmit audio chain can be disabled at the expander amplifier on chip by a mute command from MCU. This function is used to mute the transmit audio chain when data is being received from the handset so that data noise does not enter the telephone line. When transmitting data from the base to the handset, the mute function is used to disable the receive audio circuits.

2.3.1.2 *Receive Direction (telephone line to RF module)*

The receive audio signal from the telephone line makes its way through the line interface and the speech circuit before reaching the receive direction audio circuits. From the speech circuit, the audio undergoes a first stage of amplification and light low-pass filtering. Following the amplifier, the receive audio is compressed and fed directly to the pre-emphasis stage. The compressor does a straight 2-to-1 conversion; the dynamic range is reduced by one half. The compressor amplifier is also used to sum in DTMF feedback so that the tones can be heard from the handset's receiver. Pulse dialling feedback is accomplished similarly by summing in the hook switch signal level at the same location.

From the compressor output, the receiver audio signal enters an Tx audio level control circuit. . The level control, as in the transmit direction, has a range of about 15 dB and is used to set the level applied to the RF module. This level control thus sets the FM deviation and is necessary to compensate for component tolerances and variations in the sensitivity of the FM circuits.

Transmit data is resistively combined in after the level control from which point it shares the rest of the audio circuits with the receive audio. However, either only audio or only data will be present at any given time to prevent corruption of the signals. To further minimise the chance of data corruption, the receive audio circuits are disabled after compressor using the mute function as mentioned in above section. Muting this part of the receive audio chain ensures that any noise or signals on the telephone line do not interfere with data transmissions to the handset. The output is then coupled to the RF module's frequency modulator.

2.3.2 Handset

The audio circuits in the handset provide for speech exchange between the audio transducers (ear-piece receiver and microphone) and the RF module that communicates with the base. Figure 2 shows the circuitry for audio processing in the handset.

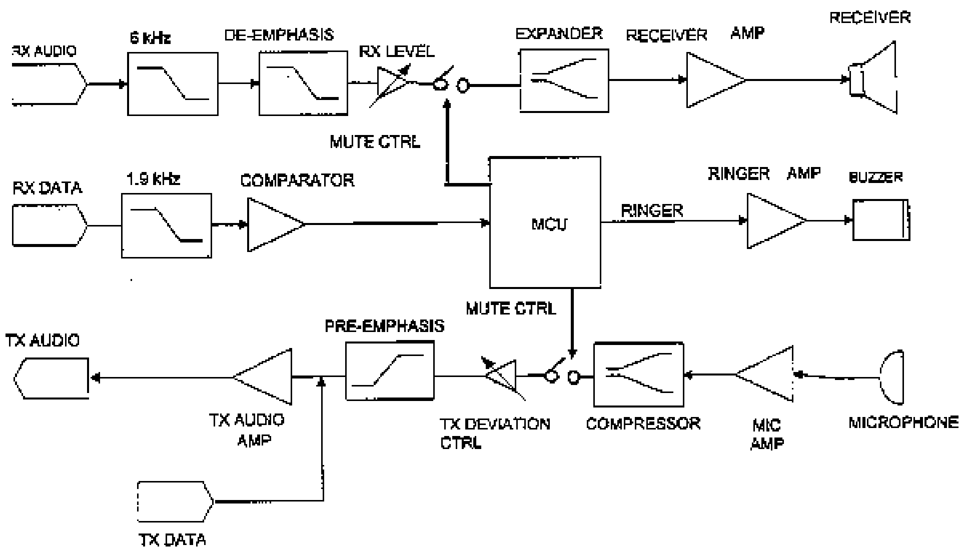


Figure 2. Audio Circuits in Handset

2.3.2.1 Receive Direction (RF module to ear-piece receiver)

The received audio is transmitted from the base to the handset using FM. The FM signal from the base enters the handset RF module where the signal undergoes filtering, down-conversion, and finally demodulation. The baseband audio then leaves the RF module via the demodulator IC (for a deviation of ±25 kHz). From the RF module, the audio is low-pass filtered and directed to both an audio channel and a data channel. The audio undergoes this split in directions because both data and audio share the same circuit.

The DETO pin is connected to a low-pass filter with a -3 dB cut-off set at about 6 kHz. The filtered audio is then passed to a de-emphasis filter where de-emphasis occurs across the entire audio band (300 Hz to 3400 Hz) at a rate of 6 dB/octave or 20 dB/decade. After de-emphasis, the audio undergoes received audio level control and the expansion process. The level control, which has a range of about 15 dB, is used to set tolerances and variations.

After the expander, the audio is passed to a final stage of amplification. A volume control (selectable to "high", "mid.", and "low" volume) is processed through the feedback resistor of this amplifier. This process ensures the compliance of min. 12 dB volume change from the lowest to the highest (FCC Part 15: 68.317).

The receive audio chain can be disabled by a mute command from MCU. This function is used to mute the receive audio chain when data is being received from the base so that data noise is not heard at the receiver. When transmitting data from the handset to the base, the mute function is used to disable the transmit audio circuits.

2.3.2.2 Transmit Direction (microphone to RF module)

In the transmit direction, the speech is picked by an electret condenser microphone and passed into a microphone amplifier stage. The speech or transmit audio undergoes a first stage of amplification and light low-pass filtering. Following the amplifier, the transmit audio is compressed and fed directly to the pre-emphasis stage. The compressor does a straight 2-to-1 conversion; the dynamic range is reduced by one half.

From the compressor output, the transmit audio signal enters a pre-emphasis circuit with an audio level control. The pre-emphasis, like the de-emphasis, is set at a rate of 6 dB/octave or 20 dB/decade throughout the entire audio band (300 Hz to 3400 Hz). The level control, as in the receive direction, has a range of about 15 dB and is used to set the level applied to the RF module. This level control thus sets the FM deviation and is necessary to compensate for component tolerances and sensitivity variations of the FM circuits. Extra range control has been implemented to compensate for microphone variations.

Transmit data is resistively combined in after the pre-amphasis from which point it shares the rest of the audio circuits with the transmit audio. However, either only audio or only data will be present at any given time to prevent corruption of the signals. To further minimise the chance of data corruption, the transmit audio circuits are disabled at the compressor output using the mute command from MCU as mentioned in above section. Muting this part of the transmit audio chain insures that any noise or signals from the microphone do not interfere with data transmissions to the base.

Following the transmit audio level control which programmed by the MCU, the audio goes through another stage of amplification and light low-pass filtering. It is then coupled to the RF module's frequency modulator.

2.4 *MCU Circuits*

A CMOS 4K x 8-bit microcontroller from Toshiba (TMP87C405M) is used to control all the functions in the base. The B/U MCU can be clocked by the 8-MHz signal which come from the RF chip CLKO pin or clocked by an external seperated 8MHz X'tal. The H/S MCU is clocked by a 4-MHz resonator seperately. The reason why not using the clock which can be provided from the RF chip CLKO pin is for current saving during "sleep" mode.

The base MCU controls such functions as DTMF generation, data communications, telephone signalling detection and ATE interfacing, while the handset MCU controls such functions as data communications, sleep/wake-up sequence, battery maintenance and ATE interfacing.

2.4.1 Base Unit

2.4.1.1 RSSI

When a communication channel is required, the handset searches for a channel that is unoccupied or has a very low-level interference. The H/S MCU does this by using the received signal strength indicator (RSSI) function of the demodulator to determine if a channel is available. The demodulator's RSSI, coupled with its carrier detect, supplies the MCU with a signal whose state indicates the status of a channel. A power threshold is set by the demodulator and when a channel's total power exceeds this threshold, the channel is declared to be occupied.

2.4.1.2 DTMF Generation

To minimise costs, power consumption, and space, the base MCU is used to generate the DTMF tones instead of a dedicated DTMF generator IC. The base MCU generates the tone waveforms by using a 1% R-2R ladder network connected to six of its ports to produce a 6-bit digital-to-analog (D/A) converter. The D/A converter's output is then passed through an low-pass filter to clear the waveforms of high frequency ripple caused by the D/A conversion.

As mentioned in previous sections, the DTMF tones are combined in at the input of the last amplifier stage in the transmit direction for transmission onto the telephone line. The transmit audio chain is disabled before the expander during DTMF dialling to stop audio from the RF module from entering the telephone line and interfering with the dialling. Also, the DTMF tone will feedback to handset receiver by sidetone signal path.

2.4.1.3 Data Communications

The data is transmitted between the handset and base at 1000 bps using Manchester Coding format. The two frequencies used in the keying are 500 Hz and 1000 Hz. Logic 0 and logic 1 are represented by falling edge and rising edge of 1kHz data respectively. A separate modem chip is not required in the design since the MCU generates and decides all the data. The level for data transmissions is set to produce about ±50 kHz of deviation.

Received data from the handset is passed form the demodulator IC to undergoes low-pass filtering. The received data is split off to the receive data chain; as mentioned in a previous section. It is shared by both the audio and data, thus requiring a splitting junction . This received data chain consists of low-pass filtering and a comparator to restore the data to its original condition. After conditioning, the data is coupled directly to the MCU for analysis.

2.4.1.4 Reset Circuit

The reset circuit for the base MCU consists of two transistor switch and support components. The circuit is designed to reset the MCU when the power supply drops to about 3.4 VDC and below. This insures that if the power supply drops to a level where logic levels may become indeterminate, the MCU will be reset to a known condition, potentially preventing erroneous operation. In addition to being connected to the reset circuit, the MCU's active low reset line is connected to the power rail via an RC network. This RC network ensures that after a reset, the MCU's reset line is brought back to a logic-high cleanly and continuously.

2.4.1.5 EEPROM

An EEPROM 93C46M8 is used in the base to store all tuning parameters for the RF chip, the current active channel, and the security code so that the information is not lost in the event of a power failure.

2.4.1.6 ATE Interface

ATE test points are available on the base to facilitate testing using automated test equipment (ATE). The ATE uses these test points to access the signals required to complete a base alignment. Connections from the base to its ATE jig are facilitated by a bed of nails. Base to ATE communication is accomplished through a dedicated port on the MCU .

2.4.1.7 LED Indicators

The base features one LED indicator to show the base unit when the handset batteries is being charged, In-use mode and channel resynchronisation.

2.4.2 Handset

2.4.2.1 Data Communications

The data is transmitted between the handset and base at 1000 bps using Manchester Coding format. The two frequencies used in the keying are 500 Hz and 1000 Hz. Logic 0 and logic 1 are represented by falling edge and rising edge of 1 kHz data respectively. A separate modem chip is not required in the design since the MCU generates and decides all the data.

The level for data transmissions are set to produce about ±50 kHz of deviation. This level was determined empirically to provide optimum data sensitivity. In the idle state, the transmit data port on the MCU is set to high impedance.

Received data from the handset is passed form the demodulator output to the data comparator where it undergoes low-pass filtering. After the splitting junction at the demodulator output, this receive data chain consists of low-pass filtering and a comparator to restore the data to its original condition. After conditioning, the data is coupled directly to the MCU for analysis.

2.4.2.2 *Keypad Control*

The keypad is arranged as a 5 x 4 (row x column) matrix with each row pulled up to V_BAT by a pull-up resistor. When the MCU is ready to accept a key press, columns 0 to 3 are set low. As soon as a key is depressed, the MCU will detect this by sampling the rows (i.e., a row will be pulled low). To determine what column was connected to the row, identifying what key was pressed, the MCU sets all columns high and then sequentially sets them low until the previous detected row is pulled low. During the sleep mode, only <Phone>, <Channel>, <Off> and <Program> keys will wake up the MCU via the interrupt request (IRQ) line. The MCU will not respond to other keys in sleep mode since it is not scanning.

2.4.2.3 *Ringer*

The ringer used in the handset is basically a magnetic transducer. It takes an electrical signal from the MCU and uses it to create a varying magnetic field to vibrate a metal diaphragm. The vibrating diaphragm, in turn, produces acoustic waves forming sound energy. The ringer draws a large amount of current when it rings and is therefore powered directly from the battery. There are four different ring signals produced by the MCU to provide four distinct ringing tones. Each signal is a combination of two frequencies. The frequencies are listed in the table below.

Ring Tone Number	Frequency #1 (Hz)	Frequency #2 (Hz)
1	2500	1250
2	2500	625
3	1250	625
4	625	313

Table 1. MCU Ringer Tone Frequencies

The MCU ringing tones are coupled to the ringer via transistors. These transistors are biased to insure that ringer output is consistent from handset to handset.

2.4.2.4 *ATE Interface*

ATE test points are available on the handset to facilitate testing using automated test equipment (ATE). The ATE uses these test points to access the signals required to complete a handset alignment. Connections from the handset to its ATE jig uses a single 6-pin header. Handset to ATE communication is accomplished through a dedicated port on the MCU.