Contents

1 General

The VT1411 is based on the VTECH VT9108 ADL(MK4). The VT1411, like the VT9108 MK4, is a next generation product within the VTECH 9108 MK4 line of 900 MHz analog cordless telephones. The VT1411 offers essentially the same functionality as the current VT9108 MK4.

1.1 Hardware

The VT1411 consists of a mobile handset and a fixed base unit. The handset contains one printed circuit boards (PCB) which hold all its electronics - the baseband audio circuits and the radio frequency (RF) circuits. The majority of the discrete components on the PCB are surface mount and use formats such as those in the 0402 size.

The base unit electronics are split over two parts — consists of one double sided PCB for the RF and audio electronics. The double-sided PCB contains the RF electronics and audio electronics, the line interface, power management, baseband controller and parts of audio chain circuits, and is comprised of through-hole (line interface) as well as surface-mount components (rest of the circuit).

1.2 Overview

This document describes the theory of operation of the VT1411. Section 2 provides the technical description of the baseband audio module, including telephone line interface, power management circuits, audio circuits, and microcontroller unit (MCU) circuits. Though similar in some cases, there are sufficient differences that the handset and base unit baseband circuitry are described in separate subsections under each functional heading. Section 3 provides the technical description of the RF module, including the antenna circuits, receiver circuits, and transmitter circuits. Unlike the baseband, the handset and base unit RF circuitry are essentially identical, and any differences applicable are highlighted within the description of each functional heading.

2 Baseband Circuit

The baseband audio circuit includes circuitry for the telephone line interface (base only), power management function, audio connection, and the microcontroller.

The following sections explain the individual blocks in the baseband module in detail.

2.1 Telephone Line Interface

The VT1411 telephone line interface couples audio and line signalling to and from the telephone line while isolating the phone from the telephone line. The interface provides basic operation for the audio and facilitates pulse dialling and ring detection. Isolation is achieved by inductively coupling audio through a transformer and by coupling line signalling through opto-couplers. The isolation is necessary for the phone to meet the FCC 1.5 kV high-pot requirement. The interface also provides protection from high voltage transients and surge currents.

2.1.1 Line Protection and Filtering

The audio signal from the central office or PBX is carried by the telephone line (tip and ring) to the phone jack. The two lines are also used to carry the ring signal (40 - 15 V_{rms}, 15.3 - 68 Hz) and various line signalling (i.e., DTMF, dial tone, etc.)

A fuse is installed in the telephone loop to limit the loop current to no greater than 250 mA. A varistor is used to limit the voltage across the line interface should a high voltage transient appear on the telephone line (i.e., lightning strike). Further high voltage protection (1.5 kV) is afforded by the audio transformer's isolation.

2.1.2 Ring Detect

The ring signal across the tip and ring is detected by an AC coupled opto-coupler. Zener diodes are used to set the threshold detection voltage and to meet the no-ring response/impedance requirement for EIA-470. Resistors limit the current flow into the opto-coupler and maintain the necessary ringing impedance as specified in EIA-470. The diode across the opto-coupler input provides a discharge path for the coupling capacitor during negative ring cycles.

On the opto-coupler output side, a pull-up resistor is used to set the transistor's collector current (i.e., sensitivity control) when a ring signal is detected. The output of the opto-coupler is connected to the MCU where the ring signal is analysed for validity. A typical ringing pattern from the central office is one second "on" and four seconds "off".

2.1.3 Pulse Dialling or Off-Hook Switch

For pulse dialling (8 - 11 pps, 58 - 64% break, interval 53 - 80 ms), an opto-coupler is used to make and break the telephone line loop. In order for the phone to function normally irrespective of the polarity of the tip and ring, a diode bridge is used to ensure the potential on the collector of the opto-coupler is positive with respect to its emitter. The opto-coupler input is connected to the MCU from which the required state of the opto-coupler is controlled. For pulse dialling, the opto-coupler is simply pulsed off and on at the appropriate rate. To set the phone off-hook, the opto-coupler is activated which closes the telephone loop.

2.1.4 Speech Circuit

To minimize cost, a speech network IC is not used in the VT1411 design. Instead, an isolation transformer with supporting hardware is used to provide all the speech network functions. The speech circuit provides line impedance matching and sidetone cancellation.

Matching (or return loss) is optimised when the termination impedance equals the source impedance. The effective impedance looking into the VT1411 is a combination of all the components' impedance in the line interface. This effective impedance was derived empirically by fine tuning the resistor across the audio transformer's secondary. The speech circuit matches a line impedance of 600 Ω (EIA-470: 4.5.2.3) while the transmit, receive and sidetone frequency responses are set with a 900 Ω line impedance (EIA-470:4.1.11 - 4.1.3).

The telephone line to audio signal conversion (or vice versa) is accomplished by transmitting audio to the transformer's secondary via a transmit amplifier, and a receive amplifier receives audio directly from the transformer's secondary.

Sidetone cancellation is accomplished by taking transmit audio (the sidetone) and resistively combining it with out-of-phase transmit audio. In a real-world situation, the match between the line interface and the telephone line is not perfect. This slight mismatch results in some transmit audio being enter into the receive direction. The sidetone cancellation signal in VT1411 is created by combining the transmit audio signal from another output of the transmit amplifier which are inherently 180° out-of-phase (i.e., the sidetone source), then feed to the receive path.

2.2 Power Management

2.2.1 Base Unit

VT1411 base power circuits consist of DC power regulation and charging circuits for the handset. The base unit operates on a regulated 5 VDC power supply. The power is supplied to the regulator via a UL-approved 9 VDC, 200 mA power adapter. During normal operation, the base unit draws about 100 mA of current (with an additional 40 mA when the handset is in the cradles).

2.2.1.1 Power Supply

DC power is supplied to the base via a UL-approved AC-to-DC power adapter rated at 9 VDC, 200 mA. The power from the adapter is then regulated down to 5 VDC. Filter capacitors are connected to both sides of the 5 VDC regulator to ensure AC variations are eliminated form the power lines. All circuits except for the Tx RF chain are powered at all times. The Tx RF chain is controlled by the MCU by programming the RF chip and is turned on when communications with the handset is required. The Tx_PWR is switched off when the Tx RF chain is not needed to minimise the use of the RF spectrum space during the idle state.

2.2.1.2 Handset Charge Circuit

To reduce costs by keeping circuits simple, the handset charge circuit is designed to supply a charging current to a cradled handset regardless of whether the battery is fully charged or not. This current varies with the charge on the battery and is limited to 0.1 C or 10% of the battery capacity by a limiting resistor. The charge circuit is supplied directly from the 9 VDC, 200 mA power adapter which insures that power is available to charge the handset battery.

In the VT1411, the handset battery has a capacity of 400 mAHr, thus the maximum charging current is set to approximately 40 mA. The specification of 0.1 C allows a battery to be constantly charged without damaging the battery. The handset charge circuit components have been selected to withstand shorting the charge contacts on the cradle. The handset charge circuit also provides a signal to the MCU for cradle detection.

2.2.1.3 ESD Protection

The charge contacts for the handset are vulnerable to electro-static discharge (ESD) because they are exposed to the outside world. Since the contacts are connected directly to the base circuits, ESD can damage some of the base internal circuits if no protection is implemented. Therefore, a number of measures have been taken to protect internal circuits from ESD damage.

Charge contact have LC filtering on them to bypass ESD. Low voltage spare gaps (arc at ~500 V based on 1 kV/mm electric discharge through air) are also implanted in the PCB layout between charge contacts and a special ESD ground. This ESD ground channels any ESD discharge directly to the telephone line and AC adapter, preventing discharges from entering the main circuits.

2.2.2 Handset

The VT1411 handset power is supplied by a three-cell battery with a nominal voltage of 3.6 VDC. This voltage will direct apply to all circuit including the RF circuit part. The variation on the performance can fall within the tolerance.

In order to achieve a long standby time, the handset conserves power by "sleeping" when not in use and occasionally "waking up". In the "sleep" condition, the handset supplies power only to those circuits deemed essential for proper operation such as the MCU and memory. The RF chip will be programmed to "Inactive" mode and all operation will stop including the oscillator circuit. In the "wake-up" condition, the handset powers the RF chip to "Rxmode" that allow it to receive data. This function is necessary to detect if the base requires the handset to act on a condition such as an incoming call. With this sleep/wake-up sequence, handset is able to achieve a six-day stand-by time. To reduce the cost, external wake-up circuit are insteaded by entering "Ideal mode" or "Normal mode" that under MCU control.

2.2.2.1 Battery Maintenance and Low Voltage Detection

The battery is recharged via a cradle contact on the base. The handset has a corresponding charge contact at the bottom of the handset chassis. The charge contact is protected from a short to ground by a diode placed in line with the battery connection. The diode prevents the battery from discharging from 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 VT1411, 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 3.0V 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.

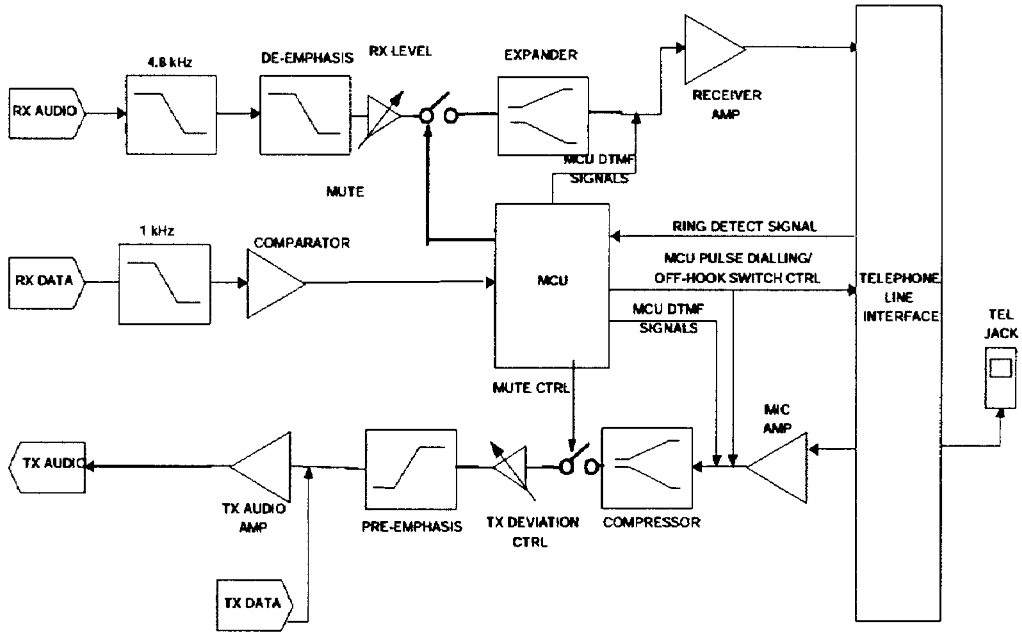


Figure 1. Audio Circuits in Base Unit

2.3.1.1 Transmit Direction (RF 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, where the signal undergoes filtering, down-conversion, and finally demodulation. The baseband audio then leaves the RF 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, 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)

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. 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 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 that communicates with the base. Figure 2 shows the circuitry for audio processing in the handset.

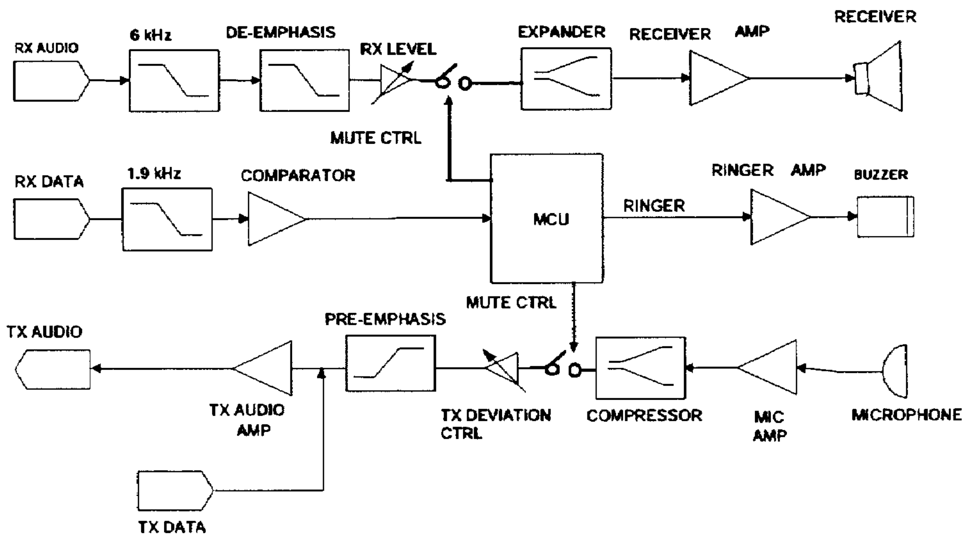


Figure 2. Audio Circuits in Handset

2.3.2.1 Receive Direction (RF 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 where the signal undergoes filtering, down-conversion, and finally demodulation. The baseband audio then leaves the RF via the demodulator IC (for a deviation of ±25 kHz). From the RF, 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 "VOL3", "VOL2", "VOL1", and "VOL0" volume) is processed through the software. This process ensures the compliance of 15 dB volume change from the lowest to the highest.

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. 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 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 ±25kHz of deviation.

Received data from the handset is passed from 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 3.3 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 ±25 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.

2.4.2.5 LED Indicators

The handset features a single LED indicator that is used to show that the handset is in-use or low battery.

3 RF Module

The basic function of the radio frequency (RF) circuits on both handset and base is to provide a full duplex wireless link between the handset and base units of the telephone. This is accomplished by setting up two simultaneous communications links between the handset and base RF. The RF receiver and transmitter circuitries essentially provide a link between the microphone and speaker in the handset to the telephone line in the base. In this way, the phone performs exactly as a corded phone, except without the cord.

The frequency at which the handset (operating at 3.6V) transmits to the base is centred around 925.65 MHz, and the frequency at which the base (operating at 5V) transmits to the handset is centred around 903.5 MHz. The VT1411 uses a wideband FM scheme to directly modulate audio signals onto the RF carriers.

The following tables outline the frequencies and corresponding channel numbers used by the RF. The handset uses a high-side local oscillator (LO) while the base uses a low-side LO to down-convert the incoming signal.

Channel #	Transmit Frequency (MHz)	Receive Frequency (MHz)	Rx LO Frequency (MHz)
1	925.05	902.3	891.6
2	926.1	903.35	892.65
3	927.15	904.4	893.7
4	923.4	905.45	894.75
5	924.45	906.5	895.8
6	925.8	903.05	892.35
7	926.85	904.1	893.4
8	923.1	905.15	894.45
9	924.15	906.2	895.5
10	925.5	902.75	892.05
11	926.55	903.8	893.1
12	927.6	904.85	894.15
13	923.85	905.9	895.2
14	925.2	902.45	891.75
15	926.25	903.5	892.8
16	927.3	904.55	893.85
17	923.55	905.6	894.9
18	924.6	906.65	895.95

Channel #	Transmit Frequency (MHz)	Receive Frequency (MHz)	Rx LO Frequency (MHz)
19	925.95	903.2	892.5
20	927	904.25	893.55
21	923.25	905.3	894.6
22	924.3	906.35	895.65
23	925.65	902.9	892.2
24	926.7	903.95	893.25
25	927.75	905	894.3
26	924	906.05	895.35
27	925.35	902.6	891.9
28	926.4	903.65	892.95
29	927.45	904.7	894
30	923.7	905.75	895.05

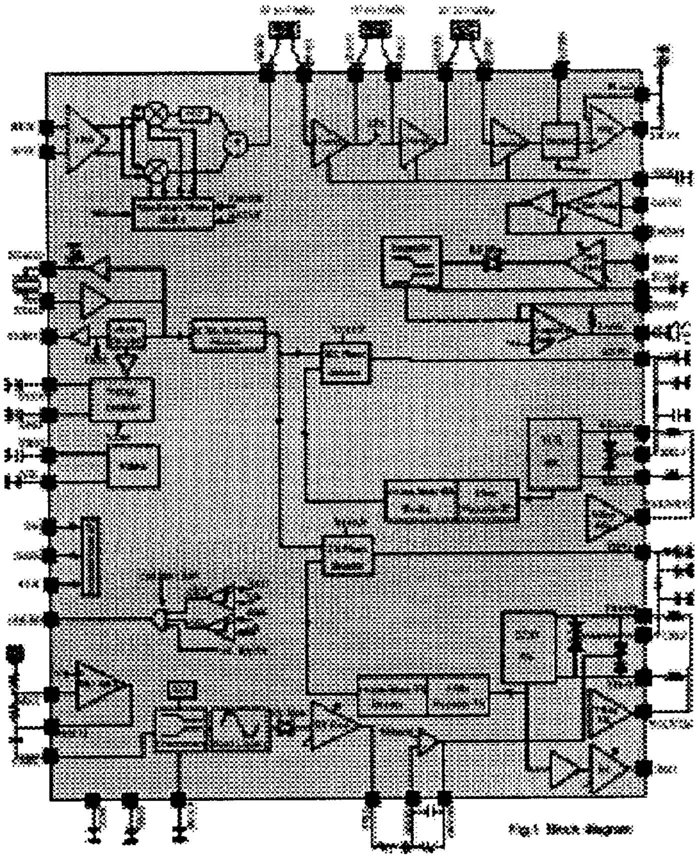
Table 2. Handset Frequencies

Channel #	Transmit Frequency (MHz)	Receive Frequency (MHz)	Rx LO Frequency (MHz)
1	902.3	925.05	935.75
2	903.35	926.1	936.8
3	904.4	927.15	937.85
4	905.45	923.4	934.1
5	906.5	924.45	935.15
6	903.05	925.8	936.5
7	904.1	926.85	937.55
8	905.15	923.1	933.8
9	906.2	924.15	934.85
10	902.75	925.5	936.2
11	903.8	926.55	937.25
12	904.85	927.6	938.3
13	905.9	923.85	934.55
14	902.45	925.2	935.9
15	903.5	926.25	936.95
16	904.55	927.3	938
17	905.6	923.55	934.25
18	906.65	924.6	935.3
19	903.2	925.95	936.65
20	904.25	927	937.7
21	905.3	923.25	933.95
22	906.35	924.3	935
23	902.9	925.65	936.35
24	903.95	926.7	937.4
25	905	927.75	938.45
26	906.05	924	934.7
27	902.6	925.35	936.05
28	903.65	926.4	937.1
29	904.7	927.45	938.15

Channel #	Transmit Frequency (MHz)	Receive Frequency (MHz)	Rx LO Frequency (MHz)
30	905.75	923.7	934.4

Table 3. Base Unit Frequencies

Both the handset and base RF follow the same block with only minor changes to incorporate the different transmit and receive frequencies. The block diagram of the RF chip is shown below:



There are seven important input/output signals that are necessary for operation of the RF section (this does not include the separate supply lines for both Tx and Rx sections). A 8.0 MHz reference is present for use in the frequency synthesisers. The accuracy of this 8.0 MHz input will affect the accuracy of the transmit and receive frequencies. In order to ensure proper operation of the RF, the 8.0 MHz reference signal must be at least $128mV_{p-p}$ in amplitude. Also present is the three-line data (SPI) bus on which data is transferred to the synthesisers to set both the transmit and receive frequencies.

The modulation input allows analog voice and digital data (signalling) to be modulated directly onto the Tx carrier. There are three outputs from the RF: Rx audio, Rx data and RSSI. The Rx data output is the demodulated signal which is subsequently filtered. The Rx audio output is the recovered analog voice that is sent to the audio circuits for additional processing. The RSSI output gives an indication of received signal strength. This is set to be high when the input signal is -90dBm or less at the antenna.

The RF performs a single down-conversion of the incoming RF signal to 10.7 MHz where it is demodulated. The transmit section directly modulates the RF carrier.

The following sections explain the individual blocks in the RF module in detail.

3.1 Antenna Section

3.1.1 Antenna

The antenna is a device that allows effective conversion of energy from air to the RF circuitry. The base has a short antenna with approximately 0-dB gain relative to an isotropic radiator, while the handset has a spiral antenna with approximately -3 dB gain. The duplexer and filters that follow the antenna require a 50 Ω match to operate properly. The antenna is approximately matched to 50 Ω and requires a simple lump-component matching network to achieve this. If a network analyser is attached to the BFA connector after disconnecting the duplexer, the antenna match may be measured. In order to achieve a good 50 Ω match, one must be careful not to obstruct the antenna, as any object near the antenna will affect its impedance.

3.1.2 SAW Duplexer

The duplexer itself is simply composed of two microstrip lines and discrete capacitors that shift each filter's out-of-band match to high impedance. To ensure that the duplexer is operating correctly, the match looking into the filters from the BFA connector may be measured. To do this, it is necessary to remove the 0 Ω resistor that connects the antenna to the duplexer. A return loss of approximately 15 dB should be measured for both the Tx and Rx bands.

The duplexer provide two functions: The first one is to effectively pass the correct frequencies to the Rx and Tx sections of the RF. It is important that these it has a low insertion loss in order to ensure a low front-end noise figure. It is also designed to provide > 30 dB rejection for the opposite band. This means that the transmit carrier will be attenuated by at least 30 dB before entering the receive section of the phone. A plot of H/S duplexer (Tx -> ANT) is shown below:

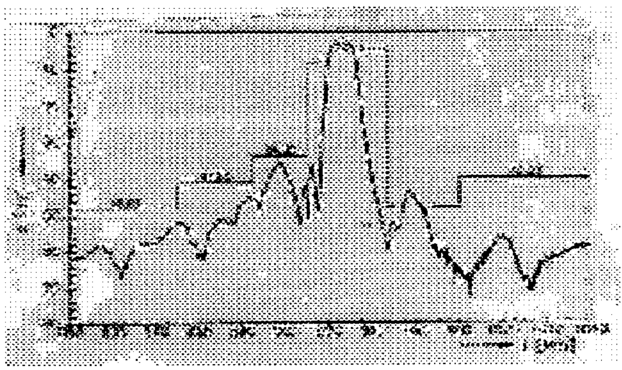


Figure 2.1 : Frequency response of H/S duplexer (Tx -> ANT)

In figure 2.1 the insertion loss is less than 4dB at 902 - 905 MHz while the 925 - 928 MHz band has >30dB attenuation. This duplexer is used for the Rx filtering and Tx filtering in the handset. In addition, the frequency response of H/S duplexer (ANT -> Rx) is shown below:

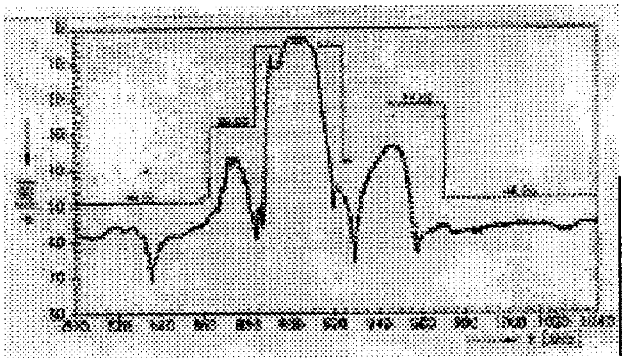
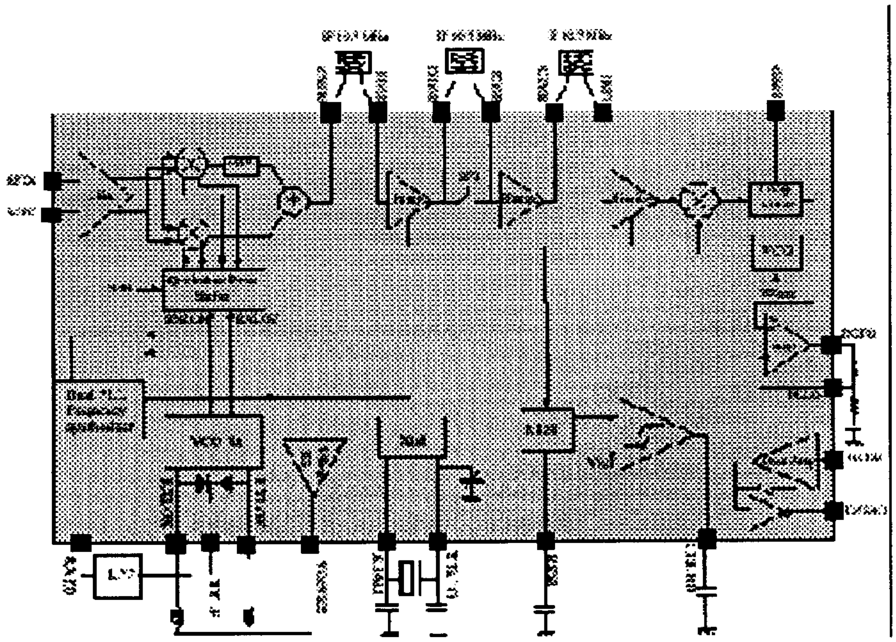


Figure 2.2: Frequency response of H/S duplexer (ANT -> Rx)

3.2 Receive Section

The block diagram of FM receiver is shown below:



3.2.1 BALUN matching and differential low-noise amplifier

The purpose of the differential low-noise amplifier is to provide enough gain and the noise figure of the Rx section is fixed to a value as low as possible. It must provide a good 50 Ω match to both the SAW duplexer by the BALUN matching. This amplifier must also have good power handling capability due to the limited filtering which precedes it. The BALUN matching ensures the gain of this stage is not too wide-band, further improving its performance by allowing it to effectively reject signals that are far out of its passband.

3.2.2 Rx VCO and PLL

The crystal local oscillator and reference divider provide the reference frequency for the Rx PLL. The programmed divider value for the reference divider is selected based on the crystal frequency. VCOs and varicaps are integrated. Resonance inductors is shared between bonding wires, leadframe of the package and external inductors. For MK4, 8MHz X'tal is selected as a reference.

The Tx PLL is combined together with Rx PLL in the block "Dual PLL frequency synthesizer". The synthesiser receives channel information from the audio through the SPI bus. It also requires a stable clock reference which is supplied from the 8M Xtal.

A passive loop filter is employed connecting to the VCO to improve the phase noise and to allow relatively fast lock time.

3.2.3 Rx Mixer, IF Filter & IF Amplifier

The function of the mixer is to combine the incoming signal with a LO signal in order to convert the desired signal to the 10.7 MHz intermediate frequency (IF). The mixer output is coupled off of the combo chip to a 10.7 MHz ceramic IF filter before it is injected into IF ampilifer 1 integrated within the combo chip. The output of IF ampilifer 1 is connected to input of IF ampilifer 2 by shorting the SFS switch (software controlled). The second 10.7MHz IF filter is not used in this design. The output of IF amplifier 2 is connected to another 10.7MHz IF filter.

The 3dB bandwidth of 10.7MHz IF filter is 110kHz. By using this in series configuration, the resultant 3 dB bandwidth is 80kHz (as shown in figure 3.1). It can provide the +-150kHz adjacent channel rejection >30dB.

The limiter provide a 6dB gain for the 10.7MHz Rx signal path. It is used to limit the signal amplitude entering the demodulator.

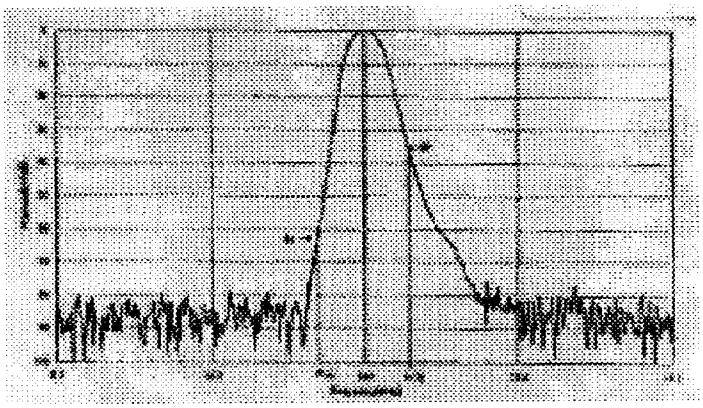


Figure 3.1 : Frequency response of 2x 10.7MHz IF filters (in series)

3.2.4 FM-PLL Demodulator

The FM-PLL demodulator is a wideband FM demodulator with programmable ability allowing to calibrate the center frequency of the VCO within the FM PLL to align the frequency as close as the norminal value (i.e.: 10.7MHz). The quadrature voltage may be observed at the ATE test point connector through DETO pin. This voltage should nominally be 1.4 V for both the base and handset when a signal is centre tuned.

The recovered audio signal from the demodulator has a peak-to-peak amplitude of approximately 0.43 V (for 25 kHz peak-to-peak modulation). One path from the recovered audio port is filtered through a low-pass data filter and passed back into the baseband.

3.2.5 Data Comparator & Data Phase Shifter

The data comparator is an inverting hysteresis comparator. The open collector output is current limited to control the output signal slew rate. An external bandpass filter is connected between DETO and DATAI (AC coupled). An external capacitor can be added to further reduce the slew rate.

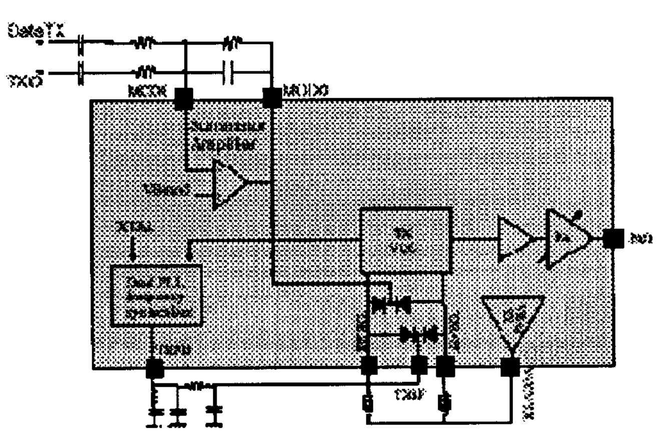
The data phase shifter allows 180 degree phase change of the received data for programming convenience.

3.2.6 RSSI Comparator

The UAA3515 provides an RSSI detect (i.e.: CDLBD pin) that is proportional to the input signal level, sending to the MCU in the baseband section.

3.3 Transmit Section

The block diagram of transmit section is shown below:



3.3.1 Tx Amplifiers

A programmable power amplifier is used to provide the necessary gain for the transmit section. It amplifies the signal from the Tx VCO to apply the correct output power into the antenna (-3 dBm for both handset and base).

There is a Chebyshev low-pass filter between duplexer Tx_in port and PAO pin. The cutoff frequency of the filter =915MHz, 3 poles and ripple <.1dB. The source and load impedence = 50 ohms. It is used to reduce the 900 MHz harmonics power level.

3.3.2 Tx VCO and PLL

The basic operation of the Tx VCO is the same as the Rx VCO, except that the Tx VCO is modulated by the transmit voice and data. The audio deviation is adjusted to a nominal value of 25 kHz by adjusting the summator amplifier gain.

The Tx PLL is combined together with Rx PLL in the block "Dual PLL frequency synthesizer". The loop filter cut-off frequency is about 60-70 Hz. This allows the data and audio modulation to include frequencies down to about 100 Hz. The power-up time of the Tx PLL is not critical.