TITLE	Theory of Operation
MODEL	Sony SPP-900 Analog 900MHz Cordless Telephone

Theory of Operation of the
SONY SPP-900
Analog 900MHz Cordless Telephone

Revision History:

Revision	Description	Page	Effective Date
0	Initial Release	All	Nov 10/98

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Title	Project Manager

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Contents

1 General

The Sony SPP-900 is based on the VTECH VT9111 (ADL MK2A). The SPP-900, like the VT9111, is a next generation product within the VTECH 910ADL line of 900 MHz analog cordless telephones. The SPP-900 offers essentially the same functionality as the current VT9111.

1.1 Hardware

The SPP-900 consists of a mobile handset and a fixed base unit. The handset contains two printed circuit boards (PCBs) which hold all its electronics - one for the baseband audio circuits and the other for the radio frequency (RF) circuits. The majority of the discrete components on both PCBs are surface mount and use formats such as those in the 0603 size. The two PCBs are connected to each other via flexible ribbon cable.

The base unit electronics are split over three PCBs — two main PCBs (one double sided and one single-sided) for the RF and audio electronics, and a small LED PCB. The double-sided PCB, which sits on top of the right up corner of the single-sided PCB, contains the RF electronics. The single-sided PCB contains the line interface, power management, baseband controller and audio chain circuits, and is comprised of through-hole (line interface) as well as surface-mount components (rest of the circuit). Finally, the LED PCB contains two LEDs and a tact switch for the Page Key on the base unit (also through-hole components). The single-sided PCB, the LED PCB, and double-sided PCB are connected together via flexible ribbon cable.

1.2 Overview

This document describes the theory of operation of the SPP-900. 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.

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2 Baseband Module

The baseband audio module includes circuitry for the telephone line interface (base only), power management function, audio connection, and the microcontroller. The block diagrams for the base and handset baseband modules are shown in Figures 1 and 2, respectively.

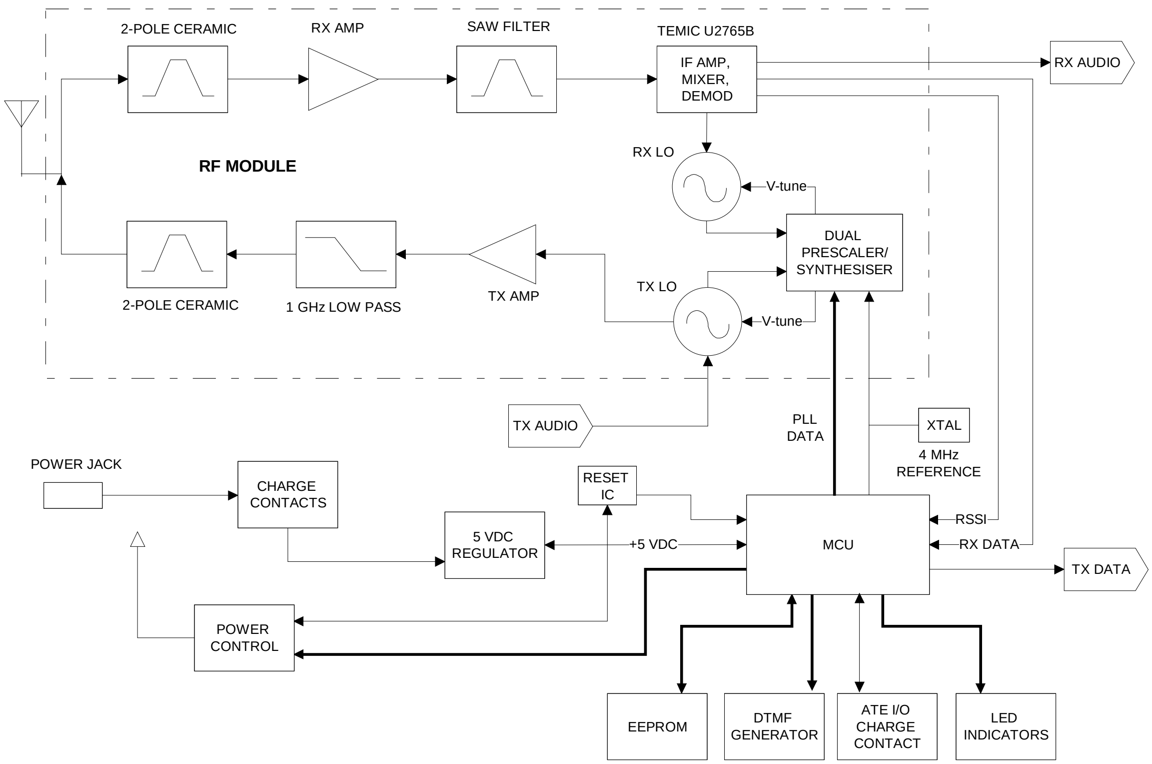

Figure 1. Sony SPP-900 Base Unit Baseband Module Block Diagram

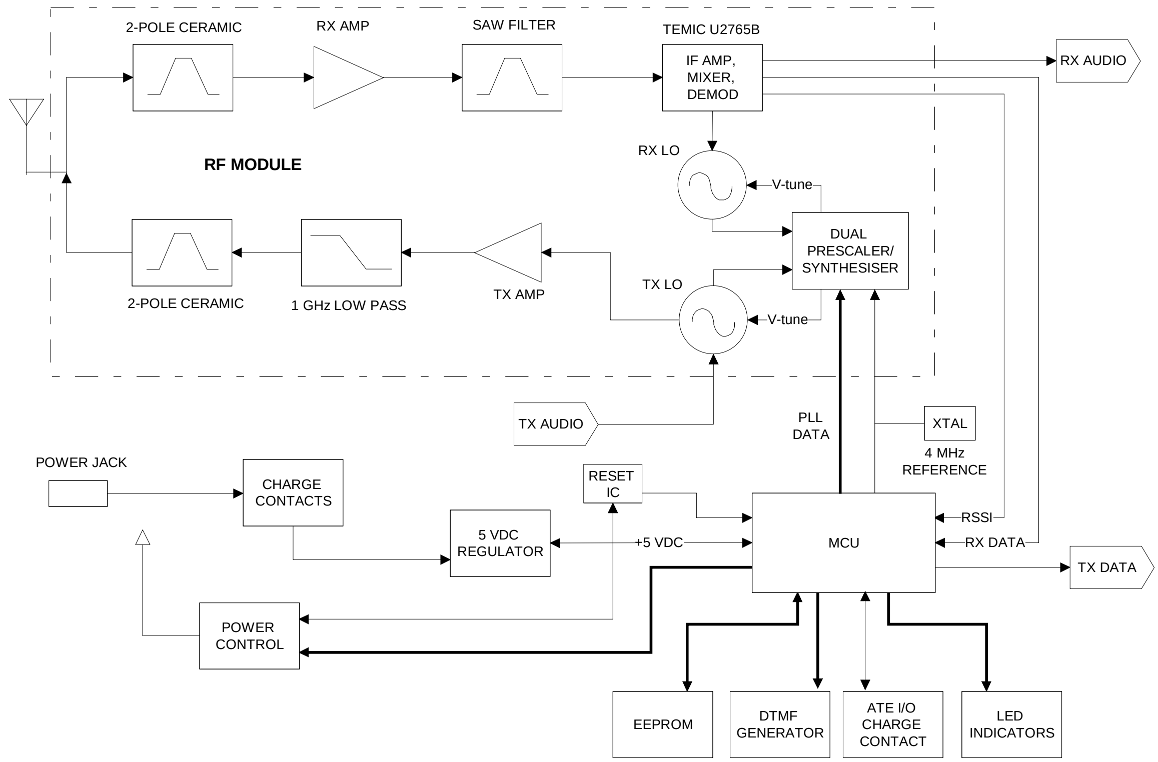

Figure 2. Sony SPP-900 Handset Baseband Module Block Diagram

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

2.1 Telephone Line Interface

The SPP-900 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 2-wire to 4-wire conversion 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 and high voltage spark gaps found in the layout of the audio PCB. The voltage spark gaps are designed to arc across at about 2 kV allowing the phone to undergo the 1.5 kV hi-pot test while preventing component damage from higher voltages.

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". The presence of a ringing signal at the base is indicated by flashing the "Line" LED.

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. The off-hook condition is indicated by turning on the "Line" LED on the base.

2.1.4 Speech Circuit

To minimise cost, a speech network IC is not used in the SPP-900 design. Instead, an isolation transformer with supporting hardware is used to provide all the speech network functions. The speech circuit provides line impedance matching, 2-wire to 4-wire conversion, and sidetone cancellation.

Matching (or return loss) is optimised when the termination impedance equals the source impedance. The effective impedance looking into the SPP-900 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-470:4.1.1l - 4.1.3).

The 2- to 4-wire (or 4- to 2-wire) conversion is accomplished by transmitting audio to the transformer's secondary via a transmit amplifier with differential outputs, and a differential input 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 SPP-900 is created by combining the transmit audio signals from the two outputs of the transmit differential 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

SPP-900 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 60 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. An LED is used to indicate the presence of the 5 VDC supply. All circuits except for the Tx RF chain are powered at all times. The Tx RF chain is supplied by a switched Tx_PWR line 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 ample power is available to charge the handset battery.

In the SPP-900, the handset battery has a capacity of 600 mAHr, thus the maximum charging current is set to approximately 60 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 and an LED labelled "Charge" to indicate the on-cradle condition.

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.

Since the MCU is connected directly to one charge contact and its ground reference is connected to another, care must be taken to prevent ESD from damaging the MCU and corrupting the ground reference. All charge contacts have LC filtering on them to bypass ESD. Low voltage spare gaps (arc at ~200 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. A spark gap can also be found between the ESD ground and the antenna.

2.2.2 Handset

The SPP-900 handset power is supplied by a three-cell battery with a nominal voltage of 3.6 VDC. This voltage is then regulated to 3 VDC and distributed throughout the handset circuits. The handset ringer is the only circuit that operates directly off the handset battery.

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. In the “wake-up” condition, in addition to the vital circuits, the handset powers the circuits 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, the SONY CPX-91 handset is able to achieve a seven-day stand-by time.

2.2.2.1 Power Control

There are five power lines used in the handset. Four of the lines are supplied by a single 3 VDC adjustable regulator and one line comes straight off the battery. The power lines give the handset the flexibility of powering down circuits that are not needed to minimise current consumption and to prevent audio signals from interfering with data and vice versa. The MCU controls the switched power lines through transistor switches.

The five handset power lines are:

- MCU_PWR - a full-time 3 VDC regulated line that is used to supply all of the handset circuits except for those in the RF modules and microphone biasing
- Rx_PWR - a switched in a 3 VDC regulated line that supplies the RF circuits associated with receiving and demodulating an incoming RF signal
- Tx_PWR - similar to the Rx_PWR, but supplies the circuits associated with modulating and transmitting the RF signal
- V_ANA - a switched 3 VDC regulated supply and is used to bias the microphone; this supply is turned on with the Tx_PWR supply and is implemented to isolate noise from the RF section from the microphone circuits
- V_BAT - a direct line from the battery and thus can vary from about 3 VDC to 4 VDC; it is used to power the ringer which requires a good voltage and current supply to operate properly; the ringer can also produce noise that can find its way onto its power supply so the V_BAT line provides some isolation from the rest of the handset’s circuits.

2.2.2.2 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 heart of the battery maintenance circuit, and for that matter the power supply, is the adjustable regulator mentioned earlier. The regulator features an internal reference to which battery charge is compared. If the battery voltage drops below 3.3 VDC, a low battery line is activated to inform the MCU. The latter action, in turn, causes the MCU to notify the user by producing an audible tone. If the battery falls below 2.8 VDC, the regulator turns off the power to prevent the handset from being used while it operates improperly. A slight hysteresis has been designed into the point where the low battery indicator is turned off when charging. The low battery indicator is disabled when the battery voltage exceeds 3.35 VDC.

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 around a Toshiba compandor IC (TA31103F for the base and TA31103FN for the handset). This IC provides compression/expansion, amplification and muting all in a clean, simple package. The IC is a good compromise between parts cost, flexibility, size, and performance. Refer to the IC's data sheets for a detailed description of its operation.

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 3 shows the circuitry for audio processing in the base unit.

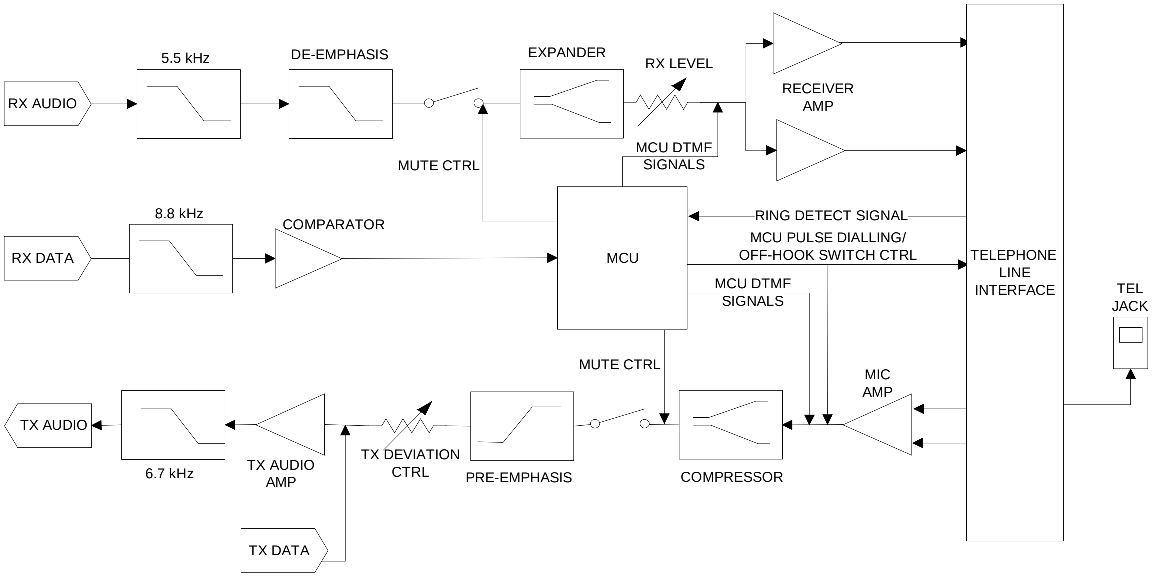

Figure 3. Audio Circuits in Base Unit

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 IC at about -16 dBv (for a deviation of ±25 kHz). From the RF module, the audio is then fed into a buffer amplifier where the audio is low-pass filtered and directed to both an audio channel and a data channel. The audio undergoes this split in directions after the buffer amplifier because both data and audio share the same circuits up to and including the buffer amplifier. The filtering at the buffer amplifier provides some rejection at higher frequencies (>40 kHz).

The buffer amplifier output is connected to an active third-order low-pass filter with a -3 dB cut-off set at about 5.5 kHz. The filter has unity gain in its passband. The filtered audio is then passed to an active 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 audio undergoes the expansion process and is passed through a transmit audio level control. The level control, which has a range of about 12 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 the level control, the audio is passed to a final stage of amplification. The output of this amplifier is taken differentially which leads to the rest of the speech circuit and telephone 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 by a mute function on the compander IC. 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. To minimise costs, the transmit audio mute function also simultaneously disables some of the receive audio circuits. 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 integrated level control circuit. . The level control, as in the transmit direction, has a range of about 12 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 at the compressor using the mute function as mentioned in Section **Error! Reference source not found.**. 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.

Following the receive audio level control (or deviation adjust), the audio goes through another stage of amplification and light low-pass filtering before being passed to an active third order low-pass filter. The filter's -3 dB cut-off is et to approximately 6.7 kHz and has unity gain in the passband. The third order low-pass filter's 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 4 shows the circuitry for audio processing in the handset.

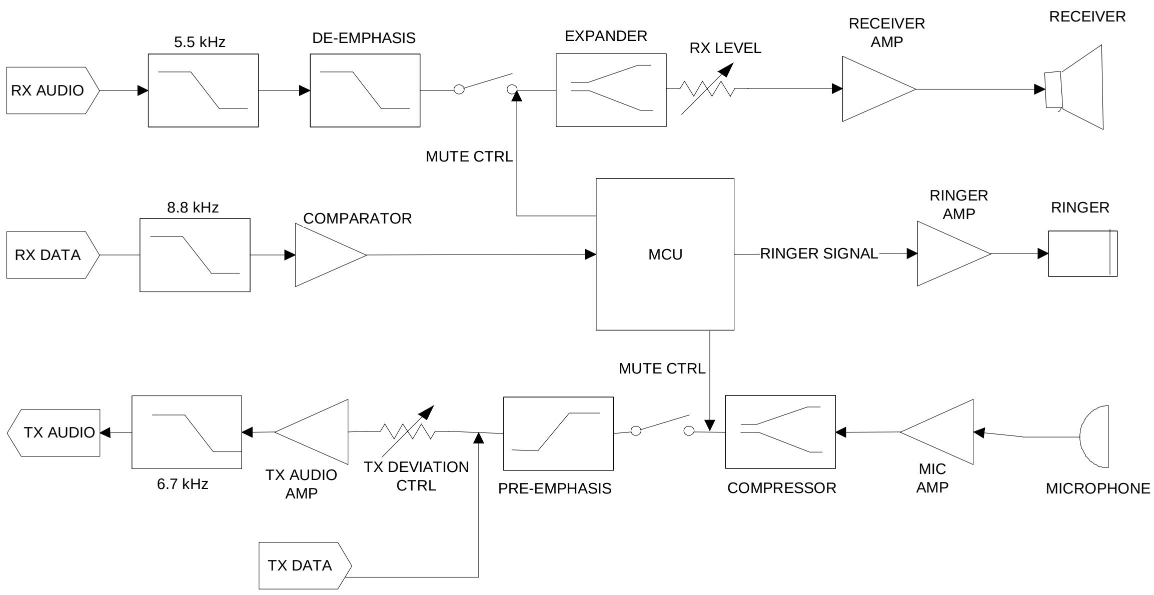

Figure 4. Audio Circuits in Handset

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

The receive 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 at about -16 dBv (for a deviation of ±25 kHz). From the RF module, the audio is then fed into a buffer amplifier where the audio is low-pass filtered and directed to both an audio channel and a data channel. The audio undergoes this split in directions after the buffer amplifier because both data and audio share the same circuits up to and including the buffer amplifier. The filtering at the buffer amplifier provides some rejection at higher frequencies (> 40 kHz).

The buffer amplifier output is connected to an active third order low-pass filter with a -3 dB cut-off set at about 5.5 kHz. The filter has unity gain in its passband. The filtered audio is then passed to an active 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 the expansion process and is passed through a transmit audio level control. The level control, which has a range of about 20 dB, is used to set tolerances and variations. Specifically, extra adjustment range has been implemented to compensate for variations in the receiver's sensitivity.

After the level control, 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 12 dB volume change from the lowest to the highest (FCC Part 15: 68.317).

The receive audio chain can be disabled at the expander amplifier by a mute function on the compander IC. 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. To minimise costs, the receive audio mute function also simultaneously disables some of the transmit audio circuits. 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 integrated 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 20 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 before the level control from which point it shares the rest of the audio circuits with the receive audio. However, either only audio or only date 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 using the mute function as mentioned in Section 0. 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 audio level control (or deviation adjust), the audio goes through another stage of amplification and light low-pass filtering before being passes to an active third order low-pass filter. The filter's -3 dB cut-off is et to approximately 6.7 kHz and has unity gain in the passband. The third order low-pass filter's output is then coupled to the RF module's frequency modulator.

2.4 MCU Circuits

A CMOS 8K x 4-bit microcontroller from Motorola (68HC705P6CDW) is used to control all the functions in the base. The MCU is clocked by a 4-MHz crystal. 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 base searches for a channel that is unoccupied or has a very low-level interference. The base 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 in lieu 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 active third order low-pass filter to clear the waveforms of high frequency ripple caused by the D/A conversion. The -3 dB cut-off for this filter is set at about 4.6 kHz and has unity gain in the passband.

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 at the expander during DTMF dialling to stop audio from the RF module from entering the telephone line and interfering with the dialling. As well, the DTMF tones are summed in at the compressor stage in the receive direction for audio feedback at the handset receiver.

During DTMF tone generation, the MCU's modem functions are disabled to ensure that the MCU has enough resources to produce distortion free tones. If the modem functions are not disabled, the data communications associated with the modem functions, in conjunction with the DTMF sample generation, could overload the MCU and result in missed samples.

2.4.1.3 Data Communications

The data is transmitted between the handset and base at 625 baud using Minimum Shift Keying (MSK) format. The two frequencies used in the keying are 300 Hz and 600 Hz, which represents logic 0 and logic 1, 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 ±40 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 IC to the buffer amplifier where it undergoes low-pass filtering. From the buffer, the receive data is split off to the receive data chain; as mentioned in a previous section, the buffer amplifier is shared by both the audio and data, thus requiring a splitting junction at the buffer 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.

2.4.1.4 Clock Reference

The MCU crystal frequency is 4 MHz and is shared with the RF module where it is used as a reference for the phase-locked loop (PLL). A trimmer capacitor is connected on one end of the crystal is used to pull the frequency into specification. The trimmer capacitor compensates for component tolerances, crystal variations, and any other parasitics that may affect the oscillating frequency. The PLL reference is tapped off the MCU's oscillator circuit via a buffer amplifier to prevent loading of the oscillator circuit.

2.4.1.5 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.

2.4.1.6 EEPROM

An EEPROM 93C46M8 is used in the base to store speed dial numbers, the current active channel, and the security code so that the information is not lost in the event of a power failure.

2.4.1.7 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 which is connected directly to a charge contact.

As with other charge contacts, the ATE I/O contacts are protected from ESD using LC filtering and additional protection is afforded at the MCU's port by using protection diodes.

2.4.1.8 LED Indicators

The base features two LED indicators to show when the base unit is in-use, and when the handset batteries are being connected and being charged.

2.4.2 Handset

2.4.2.1 Data Communications

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2.4.2.3 Keypad Control

The keypad is arranged as a 4 x 5 (row x column) matrix with each row pulled up to MCU_PWR 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.

2.4.2.4 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	1250	312

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.

2.4.2.5 Sleep Timer

The sleep timer wakes up the handset from sleep mode by pulling the IRQ line on the MCU low. The sleep timer circuit is made up of a comparator, an RC network and supporting components. The circuit acts basically like a one-shot activated by the Rx_PWR line. With the MCU asleep, the receive power is off allowing an RC network to charge up. The charge on the capacitor is compared to a set reference and at a certain point, toggles the comparator output low. The low pulls the IRQ low, waking up the MCU.

When the MCU awakes, it executes its interrupt routine to check for base data. To do this, the MCU powers the Rx_PWR line that, in turn, resets the IRQ line high. As soon as the MCU completes its interrupt routine, the handset goes back to sleep if no base data was detected and the sequence is repeated.

2.4.2.6 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 10-pin header. Handset to ATE communication is accomplished through a dedicated port on the MCU that is connected directly to a charge contact.

As with other charge contacts, the ATE I/O contacts are protected from ESD using LC filtering and additional protection is afforded at the MCU's port by using protection diodes.

2.4.2.7 LED Indicators

The handset features a single LED indicator that is used to show that the handset is in use. There are four surface-mount LEDs located in the middle of the keypad to provide for backlit keypad as required.

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 modules. The RF receiver and transmitter circuitry 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 3V) 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 SPP-900uses 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	Receive Frequency	Rx LO Frequency
1	925.05 MHz	902.30 MHz	913.00 MHz
2	925.35 MHz	902.60 MHz	913.30 MHz
3	925.65 MHz	902.90 MHz	913.60 MHz
4	925.95 MHz	903.20 MHz	913.90 MHz
5	926.25 MHz	903.50 MHz	914.20 MHz
6	926.55 MHz	903.80 MHz	914.50 MHz
7	926.85 MHz	904.10 MHz	914.80 MHz
8	927.15 MHz	904.40 MHz	915.10 MHz
9	927.45 MHz	904.70 MHz	915.40 MHz
10	927.75 MHz	905.00 MHz	915.70 MHz
11	925.20 MHz	902.45 MHz	913.15 MHz
12	925.50 MHz	902.75 MHz	913.45 MHz
13	925.80 MHz	903.05 MHz	913.75 MHz
14	926.10 MHz	903.35 MHz	914.05 MHz
15	926.40 MHz	903.65 MHz	914.35 MHz
16	926.70 MHz	903.95 MHz	914.65 MHz
17	927.00 MHz	904.25 MHz	914.95 MHz
18	927.30 MHz	904.55 MHz	915.25 MHz
19	927.60 MHz	904.85 MHz	915.55 MHz
20	923.10 MHz	905.15 MHz	915.85 MHz
21	923.40 MHz	905.45 MHz	916.15 MHz
22	923.70 MHz	905.75 MHz	916.45 MHz

Channel #	Transmit Frequency	Receive Frequency	Rx LO Frequency
23	924.00 MHz	906.05 MHz	916.75 MHz
24	924.30 MHz	906.35 MHz	917.05 MHz
25	924.60 MHz	906.65 MHz	917.35 MHz
26	923.25 MHz	905.30 MHz	916.00 MHz
27	923.55 MHz	905.60 MHz	916.30 MHz
28	923.85 MHz	905.90 MHz	916.60 MHz
29	924.15 MHz	906.20 MHz	916.90 MHz
30	924.45 MHz	906.50 MHz	917.20 MHz

Table 2. Handset Frequencies

Channel #	Transmit Frequency	Receive Frequency	Rx LO Frequency
1	902.30 MHz	925.05 MHz	914.35 MHz
2	902.60 MHz	925.35 MHz	914.65 MHz
3	902.90 MHz	925.65 MHz	914.95 MHz
4	903.20 MHz	925.95 MHz	915.25 MHz
5	903.50 MHz	926.25 MHz	915.55 MHz
6	903.80 MHz	926.55 MHz	915.85 MHz
7	904.10 MHz	926.85 MHz	916.15 MHz
8	904.40 MHz	927.15 MHz	916.45 MHz
9	904.70 MHz	927.45 MHz	916.75 MHz
10	905.00 MHz	927.75 MHz	917.05 MHz
11	902.45 MHz	925.20 MHz	914.50 MHz
12	902.75 MHz	925.50 MHz	914.80 MHz
13	903.05 MHz	925.80 MHz	915.10 MHz
14	903.35 MHz	926.10 MHz	915.40 MHz
15	903.65 MHz	926.40 MHz	915.70 MHz
16	903.95 MHz	926.70 MHz	916.00 MHz
17	904.25 MHz	927.00 MHz	916.30 MHz
18	904.55 MHz	927.30 MHz	916.60 MHz
19	904.85 MHz	927.60 MHz	916.90 MHz
20	905.15 MHz	923.10 MHz	912.40 MHz

Channel #	Transmit Frequency	Receive Frequency	Rx LO Frequency
21	905.45 MHz	923.40 MHz	912.70 MHz
22	905.75 MHz	923.70 MHz	913.00 MHz
23	906.05 MHz	924.00 MHz	913.30 MHz
24	906.35 MHz	924.30 MHz	913.60 MHz
25	906.65 MHz	924.60 MHz	913.90 MHz
26	905.30 MHz	923.25 MHz	912.55 MHz
27	905.60 MHz	923.55 MHz	912.85 MHz
28	905.90 MHz	923.85 MHz	913.15 MHz
29	906.20 MHz	924.15 MHz	913.45 MHz
30	906.50 MHz	924.45 MHz	913.75 MHz

Table 3. Base Unit Frequencies

Both the handset and base RF modules follow the same block diagram shown below with only minor changes to incorporate the different transmit and receive frequencies.

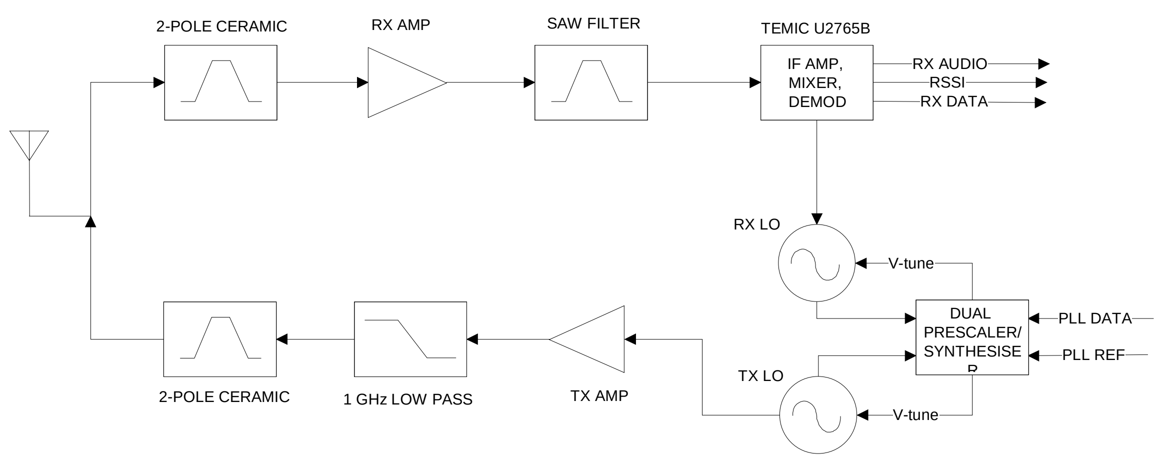

Figure 5. Sony SPP-900 RF Module Block Diagram

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 4.0 MHz reference is present for use in the frequency synthesisers. The accuracy of this 4.0 MHz input will affect the accuracy of the transmit and receive frequencies. In order to ensure proper operation of the RF module, the 4.0 MHz reference signal must be at least $1.1V_{p\text{-}p}$ 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 module: audio, Rx data and RSSI. The Rx data output is the demodulated signal which is subsequently filtered. The audio output is the recovered analog voice modulation 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 module 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 module circuitry. The base has a ½-wave antenna with approximately 0-dB gain relative to an isotropic radiator, while the handset has a 1/4 wave 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 microstrip 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 Duplexer

The duplexer ensures that the two bandpass filters do not interact with each other. It accomplishes this by making each filter see high impedance from the opposite filter in its own passband. This is necessary to ensure that both filters work effectively when connected together. If the duplexer were not present, mismatches from one filter would cause the passband of the other to be distorted and this would degrade performance.

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.

3.1.3 Rx, Tx Bandpass Filters

The Rx and Tx bandpass filters provide two functions. The first is to effectively pass the correct frequencies to the Rx and Tx sections of the RF module. It is important that these filters have a low insertion loss in order to ensure a low front-end noise figure. These filters are also designed to provide > 25 dB rejection for the opposite band. This means that the transmit carrier will be attenuated by at least 25 dB before entering the receive section of the phone. A plot of the low band filter is shown below.

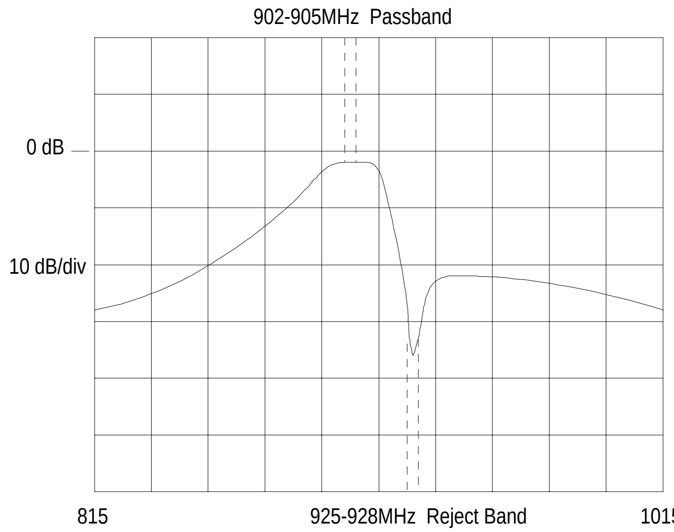

Figure 6. Low Band Ceramic Filter Response.

For this filter the insertion loss is less than 3dB at 902 - 905 MHz while the 925 - 928 MHz band has >25dB attenuation. This filter is used for the Rx filter in the handset or the Tx filter in the base. The high band filter characteristic is shown below. This filter is used for the handset Tx filter and base Rx filter.

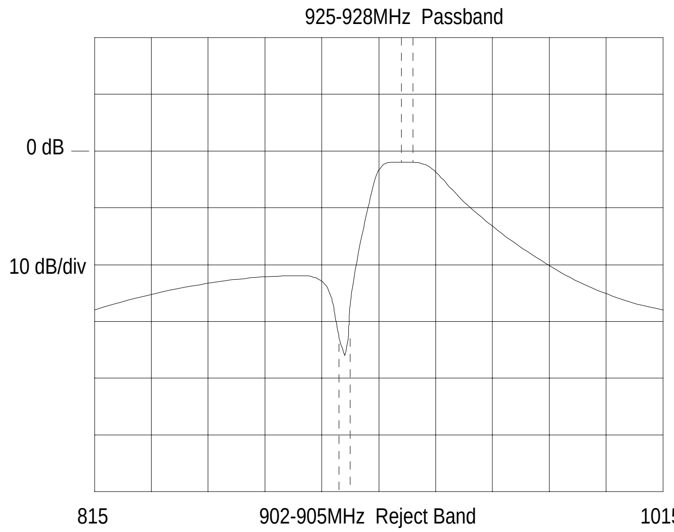

Figure 7. High Band Ceramic Filter Response

3.2 Receive Section

3.2.1 Rx Amplifiers and SAW Filter

The purpose of the first Rx amplifier is to provide enough gain so that 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 Rx bandpass filter and the surface acoustic wave (SAW) filter. This amplifier must also have good power handling capability due to the limited filtering which precedes it. The design employs a collector inductor to improve the output power capability of the transistor. This form of matching also ensures that 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.

A SAW filter is placed after the first Rx amplifier. This filter is responsible for the bulk of the filtering in the receive section. It provides more than 40 dB of image rejection and Tx carrier suppression. The insertion loss of this filter — less than 4 dB, typically 3 dB — is relatively high due to its SAW implementation. To keep the noise figure low, an amplifier is inserted before the SAW filter. (Otherwise, the noise figure of the phone would increase by the 4-dB loss associated with the SAW filter.)

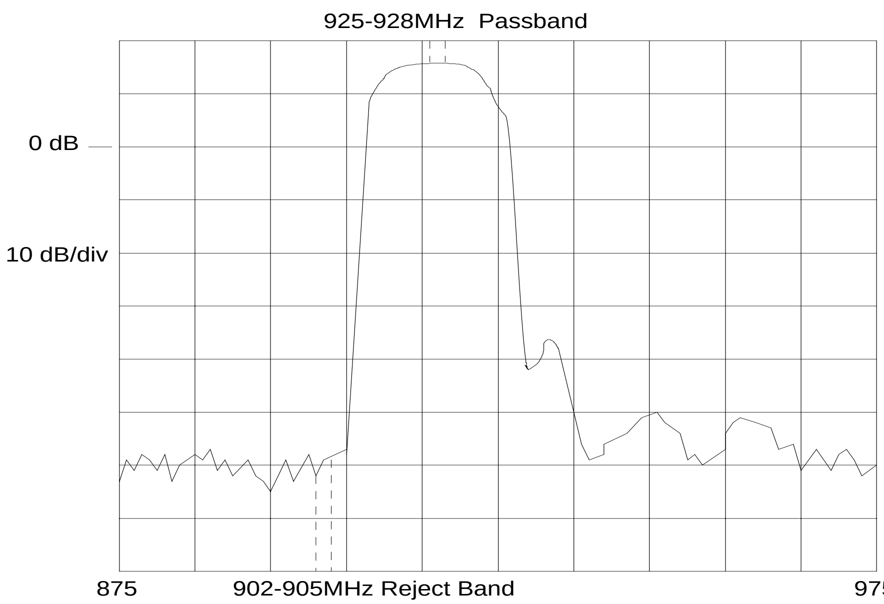

Figure 8. Rx Front-end Response.

3.2.2 Mixer, IF Amplifier, FM Demodulator

The SPP-900 RF design uses an integrated solution that provides a number of different receiver functions on a single silicon chip. The Temic U2765B IC combines a, mixer down-converter, IF amplifier and FM demodulator onto one device.

1.1.1.1 Rx Mixer

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 LO and RF signals are coupled onto pins 26 and 3 respectively on the Temic IC. The mixer output is coupled off of the Temic IC by a 10.7 MHz transformer to a 10.7 MHz ceramic IF filter before it is injected into limiter 1 integrated within the Temic IC. The output of limiter 1 is coupled off of the Temic IC into another 10.7 MHz ceramic IF filter.

3.2.2.2 IF Amplifier Stage

There is a two-stage limiting amplifier integrated with the Temic IC. Both of the limiters require external 10.7 MHz ceramic IF filters.

3.2.2.3 Demodulator

The quadrature circuit is made up one inductor, one resistor, and two capacitors. One of the two capacitors is a variable capacitor that allows tuning of the circuit. The quadrature voltage may be observed at the ATE test point connector. This voltage should nominally be 1.2 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 0.31 V (for 50 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 module.

3.2.2.4 RSSI Comparator

The U2765B provides an RSSI voltage that is proportional to the input signal level that is then sent to the MCU in the baseband section.

3.2.3 Rx VCO and LO Buffer

The Rx voltage-controlled oscillator (VCO) is a Colpitt's type oscillator operating at about 450 MHz with a frequency selective network tuned to about 900 MHz on the collector. The frequency of oscillation is controlled by a varactor diode in the tank circuit connected to the base of the transistor. This diode is connected to the loop voltage from the Rx synthesiser. Rough tuning is achieved with a variable chip capacitor. This capacitor is used to centre the tuning voltage to ensure reliable operation over a wide temperature range and also to compensate for variations in component values.

The 450 MHz LO for the phase-locked loop is coupled off the emitter of the VCO transistor. This is lightly coupled to ensure that the VCO is not loaded by the PLL. The LO buffer isolates the PLL from the VCO, preventing the Tx VCO from interfering with the Rx VCO and vice versa. The 900 MHz Rx LO signal for the mixer is coupled off the collector of the VCO transistor.

3.2.4 Rx Synthesiser

The PLL and pre-scaler for both the Tx and Rx sides are combined into one IC. The synthesiser receives channel information from the audio module through the SPI bus. It also requires a stable 4.0 MHz reference which is supplied from the MCU section.

A passive loop filter is employed to connect the synthesiser to the VCO. This tuning voltage may be observed at the ATE test point connector.

The loop filter cut-off frequency is set to about 1 kHz to allow relatively fast power-up times.

3.2.5 IF Filtering

The choice of 10.7 MHz as an IF frequency allows the use of relatively inexpensive filters. Two ceramic filters are used to achieve the desired adjacent channel suppression. Two different bandwidth filters are used, 230 kHz and 150 kHz, so that any shifting in the passband does not reduce the bandwidth excessively.

3.3 Transmit Section

3.3.1 Tx Amplifiers

A single transistor is used to provide the necessary gain for the transmit section. The transistor amplifies the signal from the Tx VCO to apply the correct output power into the antenna (-5dBm for both handset and base).

3.3.2 Tx VCO

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 through an additional (second) varactor in the tank. The audio deviation is adjusted to a nominal value of 50 kHz peak-to-peak by adjusting a T-resistor network.

3.3.3 Tx Synthesiser/PLL

The Tx PLL is combined into one IC with the Rx PLL as described in Section 3.2.4. The loop filter cut-off frequency is about 70-80 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.

APPENDIX – Alternate RF Module for SPP-900

The SPP-900 cordless telephone may use one of two alternate RF modules in the design. Both designs provide the same functionality and essentially the same performance. The main RF module was detailed in Section 3 of this document. The block diagram for the alternate RF module is shown below.

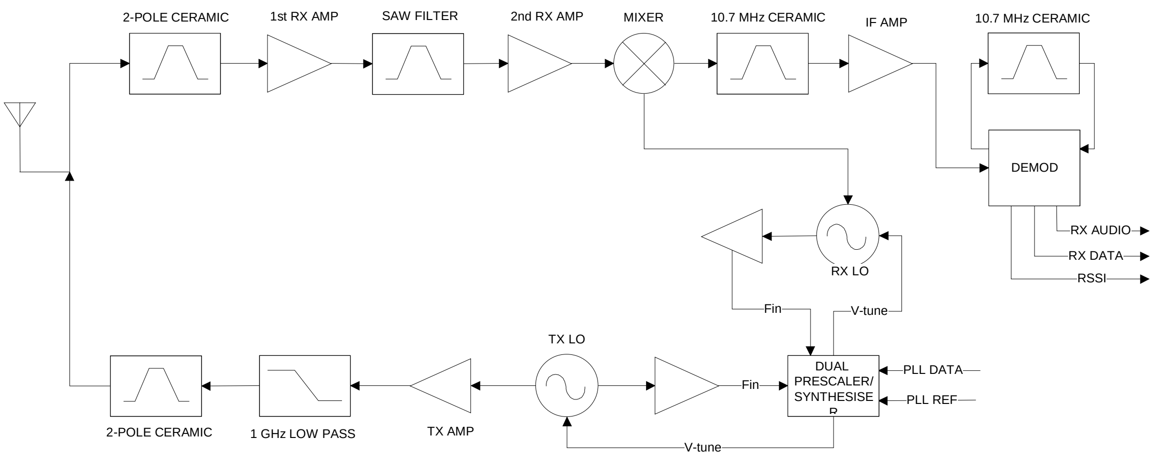

Figure 9. Alternate RF Module Block Diagram

The alternate RF module employs a discrete mixer, the Siemens CF739R, to down-convert the received 900 MHz signal to a 10.7 MHz IF signal which is then fed into the Motorola MC13156 FM demodulator to be demodulated. In the main RF module, the Temic U2765B IC replaces both of these ICs to provide the mixer and FM demodulator as well as the IF amplifier functions on a single chip.

The other area of change involves the Tx and Rx phase locked-loop (PLL). The Toshiba TB31202FN IC used in the alternate replaces the Toshiba TB31202BFN IC. With the TB31202FN, external buffer transistors are required for amplifying the RF signal into the PLL. In the main RF module, the TB31202BFN provides internal buffers so the external buffers are not required.

The Tx and Rx oscillator circuitry remains the same as both RF module designs, but the layouts and shield cans in both the handset and base unit PCBs are different to accommodate the different ICs used.

Handset Circuit Changes

The following table identifies all the differences in the RF circuit in the handset. Additional details can be found by referring to the attached schematics (43310000.sch and 4296000?.sch for the main and alternate RF modules, respectively). The item designator indicated in the table refers to the alternate RF module and can be found on the schematics for ease of reference.

Item	Change	Reason for Change
1	Receiver LNA transistor *Q2* and gain circuitry is changed (smaller gain required)	Only the Temic IC requires the larger gain.
2	FM demodulator IC *U1* and mixer circuitry changed replaces the Temic U2765B IC	Alternate RF module uses discrete approach.
3	PLL IC *U2* changed to TB31202FN and external Tx and Rx buffers *Q10* and *Q9* and associated circuitry are added	No on-board buffers available in PLL part.
4	Matching circuitry changed	Modifications for discrete approach
-	Component value changes: • change C1 & C2 to 3.6 pF • change C3 & C52 to 8.2 pF • change C5 to 6.8 pF • change C6 to 5.6 pF • change C7 to 2.4 pF • change C12 to 3.9 pF • change C54 to 2 pF • change C56 to 47 pF • change C57 to 150 pF • change C69 to 7.5 pF • change R30 to 10 kΩ • change R28 to 12 kΩ • change R31 & R40 to 100 Ω • change R38 to 11 kΩ	

Base Unit Circuit Changes

The following table identifies all the differences in the RF circuit in the base unit. Additional details can be found by referring to the attached schematics (43960002.sch and 43790000.sch for the main and alternate RF modules, respectively). The item designator indicated in the table refers to the alternate RF module and can be found on the schematics for ease of reference.

Item	Change	Reason for Change
5	Receiver LNA transistor *Q24* and gain circuitry is changed (smaller gain required)	Only the Temic IC requires the larger gain.
6	Remove Zener diode D5	Unlike Temic IC, the demod IC in alternate RF module design supports 5V operation
7	FM demodulator IC *U1* and mixer circuitry	Alternate RF module uses discrete

	changed replaces the Temic U2765B IC	approach.
8	PLL IC *U2* changed to TB31202FN and external Tx and Rx buffers *Q10* and *Q9* and associated circuitry are added	No on-board buffers available in PLL part.
9	Matching circuitry changed	Modifications for discrete approach
-	Component value changes: • Change C2 to 3.6 pF • Change C3 to 8.2 pF • Change C4 to 7.5 pF • Change C5 to 4.3 pF • Change C6 & C12 to 3.3 pF • Change change C54 to 1 pF • Change C67 to 2.7 pF • Change C76 to 27 nF • Change C89 to 2.7 nF • Change R33 & R35 to 75 Ω • Change R34 to 130 Ω • Change R37 to 100 Ω • Change R38 to 15 kΩ • Change R47 to 30 kΩ	