March 9, 2006

Dr. Andrew Rawicz
School of Engineering Science
Simon Fraser University
Burnaby, British Columbia
V5A 1S6

Re: ENSC 440 – Design Specifications for a Wireless Cell Phone Docking Station

Dear Dr. Rawicz:

The attached document is the *Design Specifications for a Wireless Cell Phone Docking Station* from Websa Technology Ltd. We are developing a mobile phone to home phone communication system which allows the user to dock his or her cell phone and use normal corded or cordless home phones to make and receive cellular phone calls.

The design specification provides, in detail, the design requirements of our Wireless Docking Station to be achieved at the end of the prototype product in terms of three aspects: architectural overviews, selected components description, and the prototype implementation. The function requirements and test plans for each module stated previously in the *Functional Specification for a Wireless Cell Phone Docking Station* will be met by this design specification.

Websa Technology consists of five talented, innovative, and dedicated fifth-year engineering students: Wilson Kwong (CEO), Andy Leung (CFO), Stephen Au-Yeung (COO), Edwin Wong (CTO-Hardware) and Bobby Ho (CTO-Software). If you have any questions about this document, or the project in general, please feel free to contact us at websa-ensc440@sfu.ca. Thank you.

Sincerely,

Wilson Kwong
Chief Executive Officer
Websa Technology Ltd.

Enclosure: ***Design Specifications for a Wireless Cell Phone Docking Station***

Design Specification

WEBSA

Wireless Cellphone Docking Station

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Submitted to: Andrew Rawicz
Steve Whitmore
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Date: March 9th, 2006

Executive Summary

Imagine being able to use your cell phone like you would with a normal telephone. Imagine the ability to just pick up the phone and dial without having to worry about poor reception or running out of batteries. Websa Technology Ltd. has the solution with the design of the Wireless Mobile-Dock, a docking station which allows one to dock his/her cell phone and use normal corded or cordless home phones to make and receive cellular phone calls.

According to CTIA The Wireless Association for the telecommunications industry, at the end of 2005, wireless subscriptions will have nearly reached 2 billion worldwide [1]. That is, 30.8% of the total world population of approximately 6.5 billion [2] currently uses a wireless subscription service. In the first six months of 2005, total wireless revenues for the US alone was $55.7 Billion USD [1]. The impressive market is made possible by the constantly increasing number of wireless service subscribers. With these numbers, Websa aims to provide wireless subscribers a more comfortable and convenient way to use their cell phones with the introduction of the Wireless Mobile-Dock.

The Wireless Mobile-Dock is divided into three critical modules: Base Station Module, Wireless Transmission Module, and Receiver Station Module. The prototype model for each module has its own set of functionalities to satisfy for evaluation purposes. The basic functions of each module are outlined below:

- **Base Station Module (BSM):**
  Interface with a docked cell phone to extract necessary signals for transmission to the receiver station module.

- **Wireless Transmission Module (WTM):**
  Transfer, with minimal packet loss, information between BSM and RSM.

- **Receiver Station Module (RSM):**
  Provide the interfacing to a normal corded or cordless telephone without generating any loss of user familiarity for telephone systems.

Upon completion of the proof-of-concept prototype (with targeted completion date set for April 2006), the Websa design team will be more equipped to further assess potential marketing ability of the product.

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Glossary

ADC	Analog-to-Digital
ATMS	Audio to Mobile
AFMS	Audio from Mobile
BSMS	Base Station Module State
BSM	Base Station Module
CPG	Call Progress Generator
CS	Cell-Phone Status
DAA	Data Access Arrangement
DAC	Digital-to-Analog
DIP	Dual in-line package
DS	Dialing String
DTMF	Dual-Tone Multi-Frequency
FIFO	First in First Out
GBW	Gain-Bandwidth Product
IC	Integrated Circuit
ISM	Industrial, Scientific, and Medical
ITU	International Telecommunication Union
MCU	Microcontroller Unit
MIB	Module Indication Bit
MISO	Master In Slave Out
MOSI	Master Out Slave In
PCB	Printed Circuit Board
PDIP	Plastic In-line Dual Package
RF	Radio Frequency
RSM	Receiver Station Module
RX	Receive
SCLK	SPI Clock
SMPI	Smart Mobile-to-Phone Interface
SPI	Serial Peripheral Interface
SRD	Short Range Device
SS	Slave Select
TDD	Time Division Duplex
TX	Transmit
UART	Universal Asynchronous Receiver-Transmitter

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1 Introduction

The *Wireless Mobile-Dock* (WMD) system is a wireless cell phone docking station that provides users the ability to make outgoing and receive incoming cellular phone calls via the use of a normal corded or cordless telephone. The WMD system will help users maximize comfort, convenience and efficiency while engaging in conversations over their cell phones. In addition, the WMD system can potentially lower monthly expenses for the user through the possible elimination of a telephone land line. The intended completion date of the WMD system is April 2006, where the final prototype will satisfy the minimum requirements previously stated in Websa's *Functional Specification for a Wireless Cell Phone Docking Station*.

1.1 Scope

This document describes the design specifications that must be met by Websa Technology's *Wireless Mobile-Dock* system. It also explains how the design will meet the functional requirements set out for the system. The design specifications are written for the implementation of the proof-of-concept models only, as described in our Functional Specifications [3]. The development for the WMD system is broken down into three sub-modules: the base station module (BSM), the receiver station module (RSM) and the wireless transmission module (WTM).

1.2 Intended Audience

This design specification is intended for use by all members of Websa Technology Ltd., and is intended to ensure that the WMD System developed by Websa meets all specified requirements. In addition, this document may serve as a tool for marketing to arrange various sales strategies in the future.

2 System Overview

Websa's Wireless Mobile Dock consists of three main modules: the Base Station module (BSM), Wireless Transmission module (WTM) and Receiver Station module (RSM). Figure 2-1 shows the functional interaction between the modules. Note that the WTM module is embedded into both the BSM and RSM.

General module descriptions and system operation are given as followed. The BSM is responsible for providing cell phone docking and interfacing capability. It will constantly monitor the cell phone status and upon receiving an incoming cell phone call interrupt, the BSM will send a notification to the RSM via the Smart Mobile to Phone Interface (SMPI). Signals from the SMPI are relayed by the WTM. The RSM, upon receiving notification of an incoming call, will attempt to ring the telephone. When the telephone is picked up, the RSM will notify the BSM and the BSM will request the cell phone to begin delivering the voice signals. As soon as the WTM streams the voice signals, a

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conversation can be established. A similar signal relay concept is applied for voice signals generated from the RSM. An analogous process is applied to outgoing calls.
The essential requirement for the *Wireless Mobile-Dock* is that users must not feel a discontinuity in their conversation or hear echoing of their own voices. The WMD system aims to provide clients with the same experience as if they were using traditional land line telephone systems.

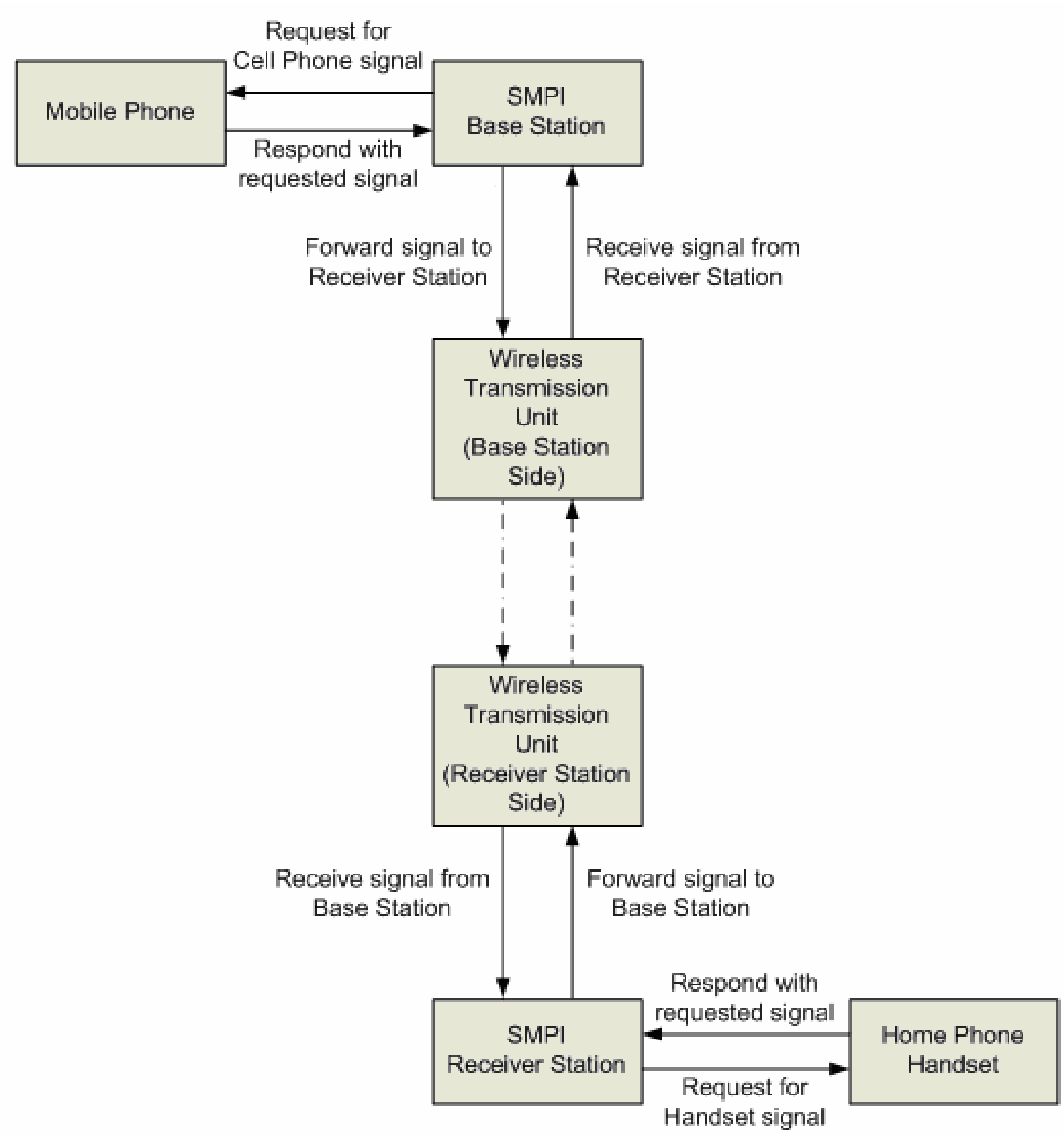

Figure 2-1: Functional interaction between modules.

3 Base Station Module

3.1 Architectural Overview

The base station module (BSM), as shown in Figure 3-1, consists of four individual components: a docked mobile phone, a microprocessor, a Digital-to-Analog converter (DAC), and a signal amplifier. The BSM is responsible for controlling the mobile phone, monitoring the mobile phone's status, and receiving/transmitting command and audio signals between the mobile phone and the wireless transmission module (WTM). A microprocessor (PIC16F877A) connects the BSM together with the WTM. Thus, the microprocessor behaves as the bridge of the communication between the mobile phone and the wireless transmission unit. Due to the specification of the WTM, all signals are transmitted in a digitized package format only and the MCU will be responsible for digitizing the incoming audio signals from the mobile phone using its own built in Analog-to-Digital converter (ADC). In addition, a DAC device is required for recovering the digitized audio signal sent from the receiver station module (RSM) thru the WTM. The signal amplifier is used to amplify the low-voltage output from the mobile phone data-port to match the dynamic range of the ADC. The following figure is a block diagram of the architectural overview of the BSM.

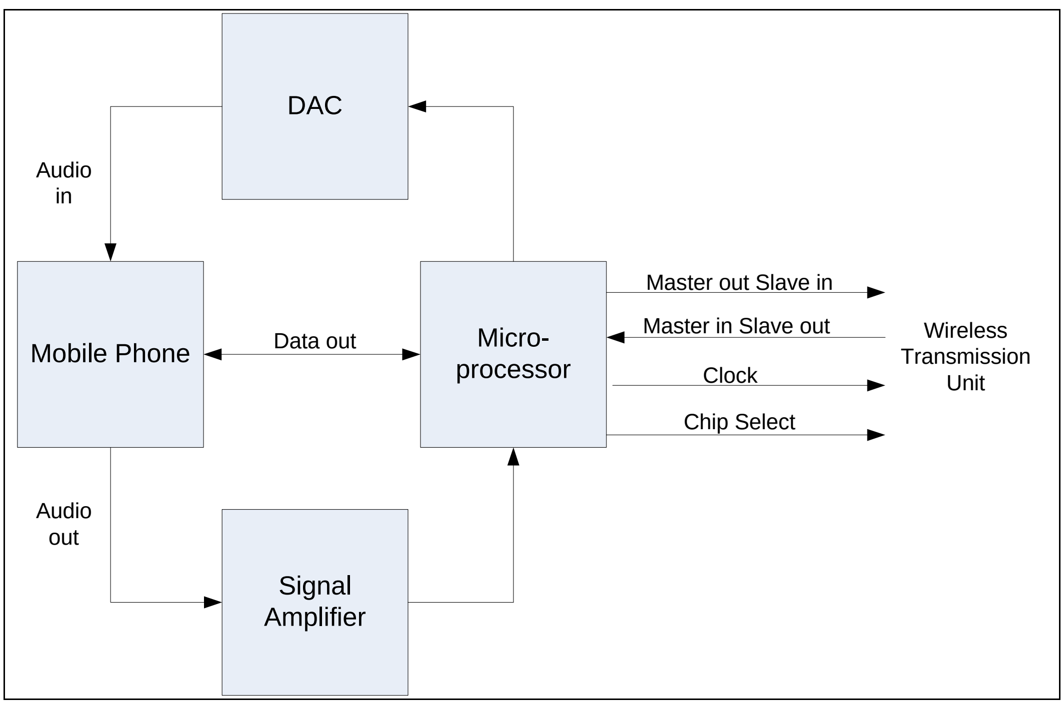

Figure 3-1: Architectural Overview of Base Station Module.

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3.2 Mobile Phone

3.2.1 Component Description

A Sony Ericsson T310 mobile phone (Figure 3-2 [4]) is used for the prototype model development due to the ease of communication establishment with the mobile phone through its data port. A serial data cable is utilized to establish the communication between the data port and the microprocessor's I/O pins. The mobile phone's data port layout is shown in Figure 3-3 [5] with the pin assignment and their corresponding descriptions presented in Table 3-1 [5].

Figure 3-2: Sony Ericsson T310 Mobile Phone.

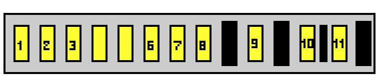

Figure 3-3: 11-pin Data Port Connector.

Table 3-1: Pin Function Description.

bottom Pin	Name	Direction	Description
1	ATMS	←	Audio to mobile
2	AFMS/RTS	→	Audio from mobile/RTS
3	CTS/ONREQ	—	CTS/Mobile Station On REQuest
4	data in	←	Data to mobile (Rx)
5	data out	→	Data from mobile (Tx)
6	ACC in	←	Accessory control to mobile. Used as Rx in some models (i.e. T68) for flashing.
7	ACC out	→	Accessory control from mobile/handsfree sense. Used as Tx in some models (i.e. T68) for flashing.
8	AGND	—	Audio signal ground + 0V reference
9	flash	—	Flash memory voltage + Service
10	DGND	—	Digital ground
11	Vcc	—	DC + for battery charging + External accessory powering

The four most important pins are pins 1, 2, 4, and 5, which represent the Audio to Mobile (ATMS), Audio from Mobile/RTS (AFMS), Data to Mobile (data in) and Data from Mobile (data out), respectively. They are the core pins for communication between the mobile phone and the BSM for voice and data signals. The data in/out pins are directly connected to the MCU I/O pins because the signals do not need to be processed in any way. The ATMS pin will need to pass thru the DAC before entering the data port while the AFMS will first go through a signal amplifier to match the dynamic range of the MCU's ADC as previously stated.

3.2.2 Component Implementation

The MCU controls and monitors the mobile phone's status thru Attention commands (AT commands) that requests the mobile phone to perform certain functions. Table 3-2 lists the AT commands used for controlling the mobile phone [6].

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Table 3-2: Attention Commands list.

AT Commands	Functions
Call Control	
ATA	Answers incoming call
ATH	Terminates an active call
ATD	Dial command
Interface Commands	
ATV	DCE Response Mode
GSM Mobile Equipment, Control, Status	
AT+CKPD	Keypad Control
AT*ECAM	Call Monitoring
Audio Control	
AT*EALR	Audio Line Request
GSM Network Service	
AT+CHLD	Call Hold and Multiply
AT+CCWA	Call Waiting Notification

Whenever the mobile phone has a change in status, it will output a custom indication to the MCU I/O pins as a data signal. Base on the information, the MCU will recognize the current state of mobile phone and perform the corresponding function, as determined in our system event flow (Section 6.1). Table 3-3 and Table 3-4 summarize the essential status indications [6].

Table 3-3: Status Indications in DCE Response Mode.

Result code (ATV1)	Result code (ATV0)	Description
OK	0	Acknowledges execution of a command
CONNECT	1	A connection has been established; the DCE is moving from command state to on-line data state
RING	2	The DCE has detected an incoming call from the network
NO CARRIER	3	The connection has been terminated, or the attempt to establish a connection failed
ERROR	4	Command not recognized, command line maximum length exceeded, parameter value invalid, or other problem with processing the command line
NO DIALTONE	6	No dial tone detected
BUSY	7	Engaged (busy) signal detected
NO ANSWER	8	"@" (Wait for Quiet Answer) dial modifier was used, but remote ringing followed by five seconds of silence was not detected before expiration of the connection timer

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Table 3-4: Status Indications of Call Monitoring.

<ccstatus>	Description
0	IDLE
1	CALLING
2	CONNECTING
3	ACTIVE
4	HOLD
5	WAITING
6	ALERTING
7	BUSY

3.3 Microprocessor

3.3.1 Component Description

The microprocessor selected is Microchip's PIC16F877A, which consists of a 40-pin plastic dual in-line package (PDIP). The pin diagram of the microcontroller is shown in the Figure 3-4 [7].

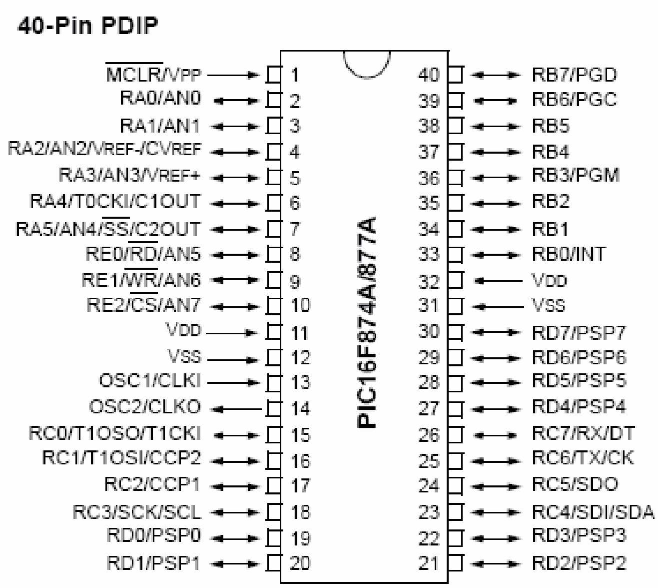

Figure 3-4: Pin Diagram of PIC16F877A Microprocessor.

This specific microprocessor is selected because of its versatility and ease of implementation and programmability. It operates at an incredibly high speed, approximately 200ns, in executing an instruction with a fast settling time of 1.6µs per bit for the ADC module. Its large number of I/O pins is also an advantage for the implementation of the UART interface, the ADC, and the wireless transmission unit. In addition, the chip has potential for higher utilization for future features of the system.

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The PIC16F877A is powered by 5V DC [7], while an external crystal oscillator (at 20MHz) is required to operate the microprocessor clock. A recommended implementation of the external crystal oscillator is shown in Figure 3-5, where *C1* and *C2* should be between 15 to 33 *p*F because of the high speed operation. R_S is not required as we are not using strip cut crystal oscillators [7].

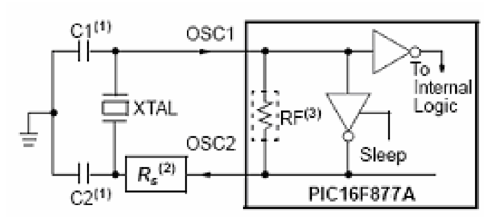

Figure 3-5: Crystal/Ceramic Resonator Operation Circuit.

Another manufacturer recommended circuit is implemented at the input to the Master Clear pin. This circuit, shown in Figure 3-6, prevents the voltage input to the pin from exceeding its specified maximum, which will result in both a Reset and current consumption outside of device specification [7].

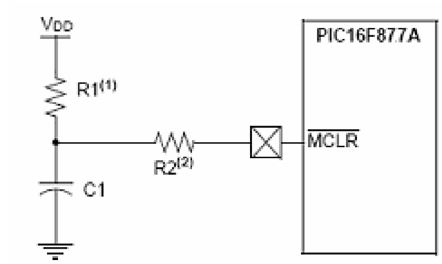

Figure 3-6: Recommended Master Clear pin Circuit.

Assembly programs can be loaded onto the microprocessor via the MPLAB ICD2 or the PICSTART Plus development application.

3.3.2 Component Implementation

As shown in Figure 3-4, the PIC16F877A microprocessor contains two sets of voltage supply pins, V_{DD} and V_{SS}. V_{DD} is connected to +5V DC, and V_{SS} is connected to ground for proper operation. As illustrated in Figure 3-5, a crystal oscillator is connected to pin 13 and 14, which are the oscillator input and output, respectively. A recommended circuit is connected to pin 1 which is the master clear, as illustrated in Figure 3-6. The remaining I/O pins are used for I/O interfacing with the SPI interface, ADC I/O, and UART I/O, which are all program controlled. At the current stage, we have not fully specified which pins will be responsible for which particular interface. However, a high level implementation block diagram is shown in Figure 3-7.

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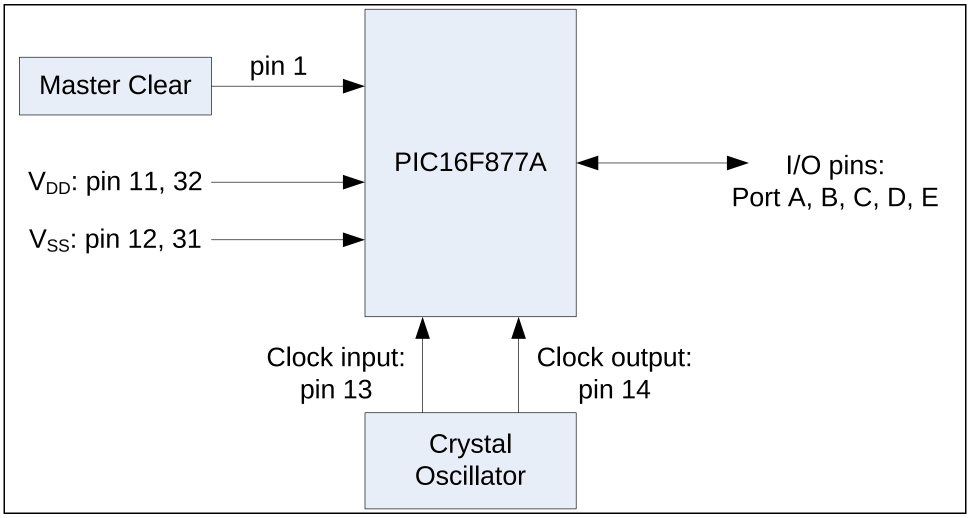

Figure 3-7: Basic Microcontroller Implementation Diagram.

3.4 Digital-to-Analog Converter

3.4.1 Component Description

The selected component, AD7533, is a low cost, high performance, 10-bit accurate, four-quadrant multiplying DAC in a 16 pin PDIP. The pin diagram of the DAC is shown in Figure 3-8 and the corresponding pin description is shown in Table 3-5 [8].

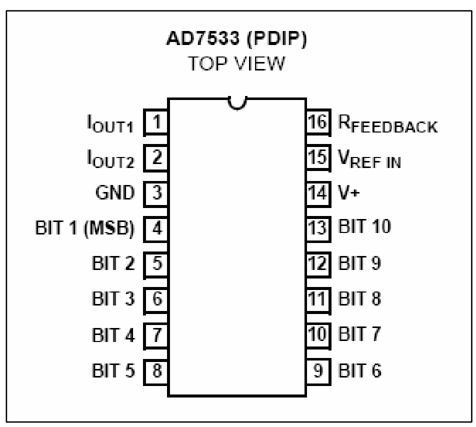

Figure 3-8: Pin Diagram of AD7533 DAC.

Table 3-5: Pin Function Description.

Pin Number		Mnemonic	Description
PDIP, SOIC, CERDIP	LCC, PLCC		
1	2	$I_{OUT}1$	DAC Current Output.
2	3	$I_{OUT}2$	DAC Analog Ground. This pin should normally be tied to the analog ground of the system.
3	4	GND	Ground.
4 to 13	5 to 10, 12 to 15, 17	BIT 1 to BIT 10	MSB to LSB.
14	18	V_{DD}	Positive Power Supply Input. These parts can be operated from a supply of 5 V to 16 V.
15	19	V_{REF}	DAC Reference Voltage Input Terminal.
16	20	R_{FB}	DAC Feedback Resistor Pin. Establish voltage output for the DAC by connecting R_{FB} to external amplifier output.
NA	1, 6, 11, 16	NC	No Connect.

The advantages of the AD7533 are its sufficiency in digital input pins and its high speed settling time (approximately 150ns). The AD7533 operates between 5V to 15V DC and provides proper binary scaling for reference inputs of either positive or negative polarity. The minimum input high voltage (V_{INH}) level is 2.4V, whereas the maximum input low voltage (V_{INL}) level is 0.8V max. The maximum input reference voltage (V_{REF}) is ±25V.

3.4.2 Component Implementation

The AD7533 DAC is a current outputting device; an external operational amplifier is needed to convert the current into voltage. The output voltage stream is then the required analog voice signal. This combination, DAC and op-amp, is called a unipolar binary operation circuit (shown in Figure 3-9 [8]), where the common LM324 is selected as the operational amplifier component.

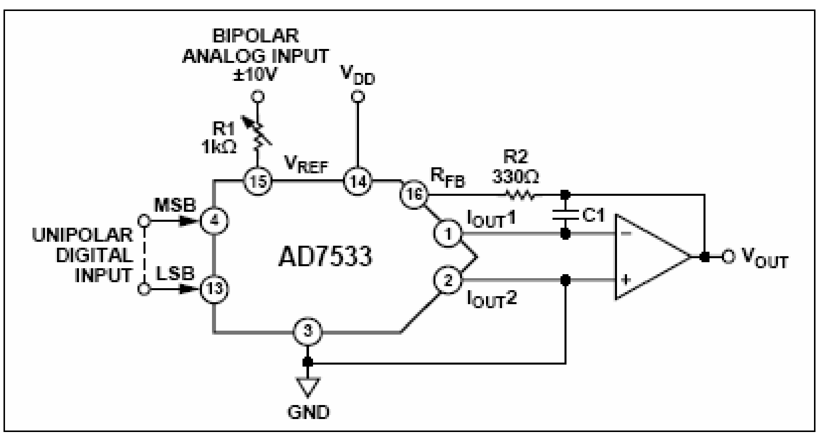

Figure 3-9: Unipolar Binary Operation Circuit.

Table 3-6 shows several examples of the output voltage level corresponding to its respective digital input value. As listed, when all input bits are active high, the output is equal to the negative of the reference voltage [8]. Conversely, when all inputs are active

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low, the output is equal to zero. The output is a negative of the reference voltage because the LM324 is acting as an inverting op-amp in this design.

Table 3-6: Unipolar Binary Operation.

Digital Input MSB LSB	Analog Output (V_{OUT} as shown in Figure 11)
1111111111	$-V_{REF}\left(\frac{1023}{1024}\right)$
1000000001	$-V_{REF}\left(\frac{513}{1024}\right)$
1000000000	$-V_{REF}\left(\frac{512}{1024}\right)=\left(\frac{V_{REF}}{2}\right)$
0111111111	$-V_{REF}\left(\frac{511}{1024}\right)$
0000000001	$-V_{REF}\left(\frac{1}{1024}\right)$
0000000000	$-V_{REF}\left(\frac{0}{1024}\right)=0$

3.5 Signal Amplifier

3.5.1 Component Description

As previously stated, the low-voltage output of a mobile phone data port must initially be amplified to match the dynamic range of the ADC. An LM324 op-amp is chosen for this design because it has low noise and a rail-to-rail output stage. Most importantly, it has a high gain-bandwidth product (GBW), which is approximately 1MHz, allowing for good amplification of the voice signal with very low distortion. Figure 3-10 shows the pin diagram of the PDIP LM324 op-amp [9].

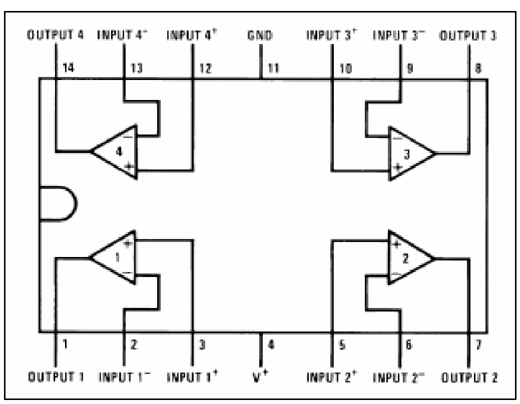

Figure 3-10: Pin Diagram of LM324 Operational Amplifier.

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3.5.2 Component Implementation

The signal amplifier circuit is configured as an inverting amplifier, as shown in Figure 3-11, which means the output is phase-reversed compared to the input [9]. The resistor R1 and the internal impedance of the data port pin form a voltage divider biasing the input voltage. Due to the use of a single voltage supply, the resistor R3 and R4 are necessary to set the common-mode voltage at half the supply voltage. The feedback resistor, R5, and the input resistor R2, determine the voltage gain factor of the amplifier. The capacitor C1 blocks the DC level, used for biasing the input, from the inverting input of the op-amp.

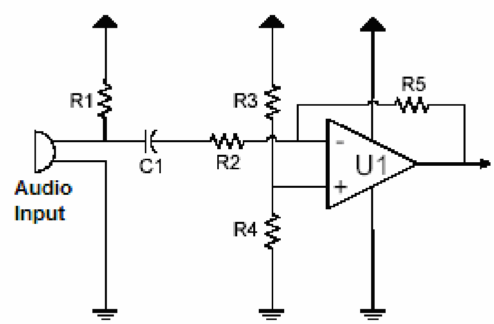

Figure 3-11: Audio Signal Amplifier Circuit.

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4 Wireless Transmission Module

4.1 Architectural Overview

The WTM, although defined as a module, is actually composed of two transceiver units where one unit is embedded into the BSM and the other one is embedded within the RSM. The wireless transceivers serve as the bridging device between the BSM and RSM. Looking at Figure 3-1 and Figure 5-1, we notice both figures have an arrow that starts from the microprocessor and points to the WTM. We fill the gap with the introduction of Figure 4-1.

Figure 4-1: Overview of WTM as bridging device.

4.2 RF Radio Module

4.2.1 Component Description

The wireless communication between the base and the receive station is done through the use of the Chipcon CC1100EM RF transceiver module. Figure 4-2 illustrates the actual CC1100EM unit [10].

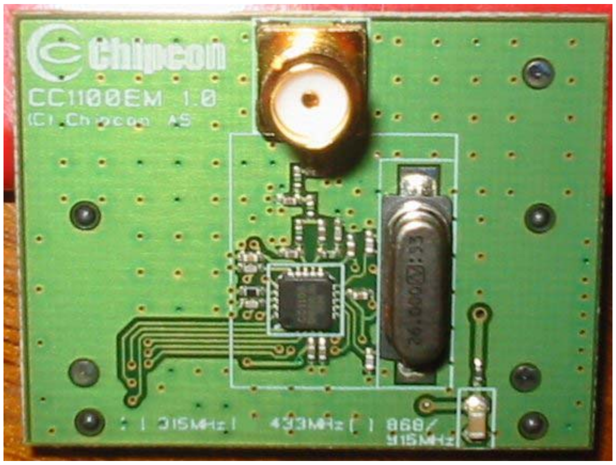

Figure 4-2: CC1100EM Wireless Transceiver

The main reason behind the selection of this transceiver over other models is due to its ease of use and its extensive hardware support for packet handling and clear channel assessment. The built-in hardware packet filtering mechanism reduces the load on the microcontroller and the clear channel assessment ensures data are always sent in the least occupied channel, resulting in a more reliable wireless link between the two stations.

The transceiver is designed to operate within the ISM (Industrial, Scientific and Medical) and SRD (Short Range Device) frequency bands at 868 and 915 MHz [11]. In this application, we decided to choose 915 MHz as the operating frequency because this frequency lies within the 906-928MHz band, which is a legal frequency band in North America [11]. In contrast, the 868MHz band is only legal in Europe thus it cannot be used in North America.

Another reason we chose this radio module is due to its high data rate. In our application, voice is sampled at 8 KHz (because human voice is roughly 4kHz and applying Nyquist's sampling theorem, we result in a sampling frequency of 8kHz [12]) as an 8 bit sample. The PICF16877A provides a 10 bit ADC conversion resolution but only the most 8 significant bits are used for transmission only [13]. This requirement means the radio must be able to transmit at a minimum rate of 64kbps without taking the additional overhead frames before and after the voice data. The CC1100EM provides a user programmable data rate up to 250kbps, which allows for plenty of overhead to include other useful status frames to be sent along with the voice data.

4.2.2 Component Implementation

The Chipcon CC1100 radio module uses Serial Peripheral Interface (SPI) as its main communication channel with the PIC16F877A microcontroller. SPI is a serial bus standard established by Motorola. The SPI employs a master / slave relationship, which utilizes four signals: clock (SCLK); master output, slave input (MOSI); master input, slave output (MISO); and slave select (SS).

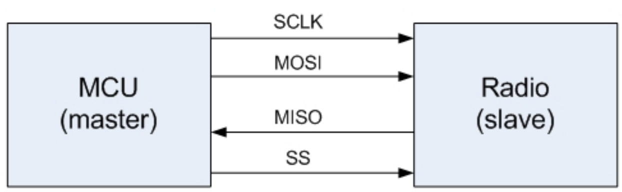

Figure 4-3: Block diagram to illustrate connection between radio and MCU

Figure 4-3 shows a block diagram between the microcontroller and the radio module with the four SPI signals. The microcontroller acts as a master device which will supply the radio (slave) with a clock signal through SCLK. When the radio is selected by the microcontroller thru the slave select (SS) signal, data are transmitted in both directions simultaneously through the MOSI and MISO line.

During a SPI transmission, data are always transferred in both directions between the master and the slave. It is the hardware's responsibility to determine if the data is useful or not. For the SPI interface on the CC1100, all transaction starts with an 8 bit header byte containing a read / write bit, a burst access bit and a 6 bit address. When a header byte is sent on the SPI interface thru the MOSI line, a chip status byte will be sent back to the microcontroller thru the MISO signal path. The chip status byte contains key status signals useful for the microcontroller. Figure 4-4 illustrates registers read and write operation for the CC1100 radio [14].

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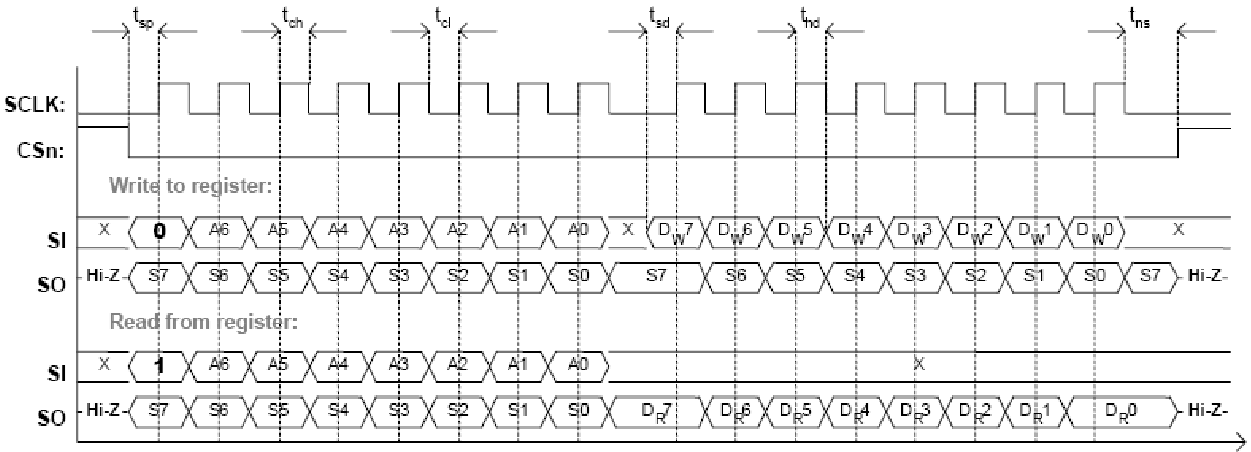

Figure 4-4: CC1100 registers read / write operation

The SPI clock (SCLK) is configured to run at 5 MHz, which is the highest speed allowed for the PIC16F877A microcontroller [7]. The reason behind this speed selection is to reduce the amount of blocking on the whole program flow during the SPI read / write cycle.

For handing packets during transmitting and receiving, full duplex voice transmission is achieved through the use of time division duplex (TDD) mechanism. In order to implement TDD, the radio needs to transmit voice data in packets and the CC1100 radio has built in hardware packet handling support to ensure easy and reliable transmission. Figure 4-5 illustrates the format for each radio packet [14].

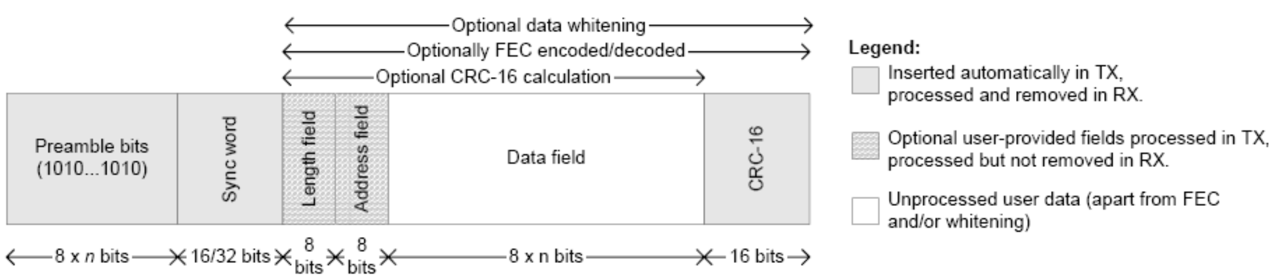

Figure 4-5: Packet format

Before data are being sent out, the CC1100 radio automatically adds the following elements to the outgoing data:

- Preamble bits field where the length is programmable by the user
- A two byte synchronization word which can be extended to 4 bytes
- Optionally compute and add a CRC checksum over the data field.

Depending on the transmission setting, the radio can optionally specify the length of the packet along with an address field that can be use as a sanity check for the receiver.

5 Receiver Station Module

5.1 Architectural Overview

The receiver station module (RSM) shown in Figure 5-1 provides an overview of the design that will meet the functional requirements set out in Websa Technology's Functional Specification [3]. Six individual components make up the RSM, including the microcontroller unit (MCU), the Dual-Tone Multi-Frequency (DTMF) transceiver, the Digital to Analog Converter, the Data Access Arrangement (DAA), the call progress generator, and the ringer voltage generator. Each component is controlled by a single PIC16F877A MCU. The MCU is also connected to (and controls) the part of the WTM that resides in the RSM.

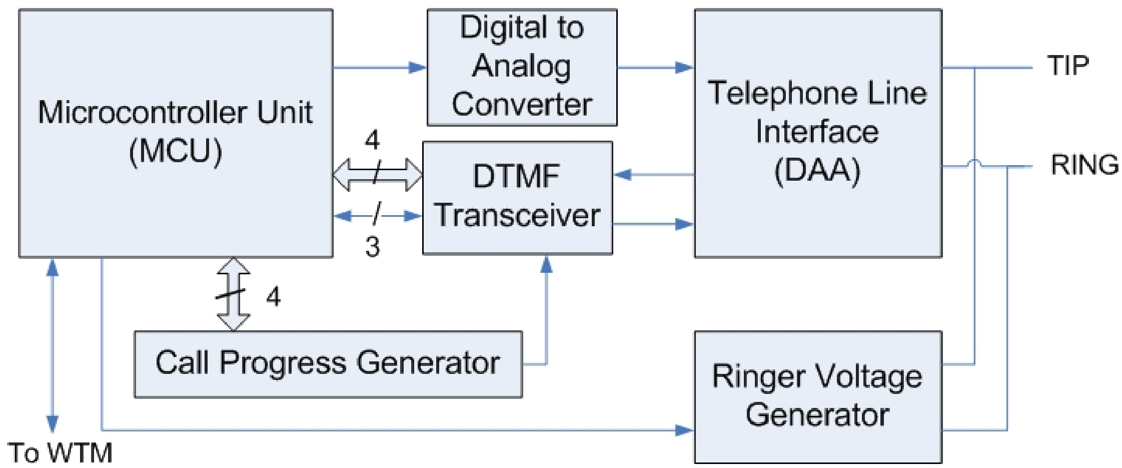

Figure 5-1: Architectural Overview of Receiver Station Module

When a corded or cordless handset is lifted off hook for the purpose of dialing an outgoing call, a dial tone is expected to signal the readiness of the unit. This dial tone can be achieved by the use of a call progress generator (CPG). In addition, the CPG is also capable of generating alert tones, which is used to signal a handset is off hook and no input has been received for a predetermined timeout period. Since the RSM must be able to transmit and receive audio and touch tone signals, the DAA provides the necessary telephone line interface between analog transmission equipment (the DTMF transceiver) and the *TIP* and *RING* lines. Digital voice signals that are received by the MCU from the WTM must be converted into analog signals before they can be understood by the DAA. The DTMF transceiver is used to decode and generate touch-tone audio signals, which are to be heard by the user when the keys on a handset are pressed. Because any corded or cordless handset that is connected to the RSM must ring upon detection of an incoming call, a ringer voltage generator circuit is required. The following sub sections detail the specifications of each component and its implementation.

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5.2 Microcontroller Unit

The MCU (Microchip's PIC16F877A), with its component description already presented in section 3.3, will be the single source of control for the RSM. Its implementation is similar to that of the BSM, with the I/O pins assigned for controlling the DTMF, DAA, DAC, CGP and the ring voltage generator circuit.

5.3 DTMF Transceiver

5.3.1 Component Description

The MT8880CE, shown in Figure 5-2, is a fully integrated DTMF transceiver used within the receiver station module (RSM) and is manufactured by Zarlink Semiconductor [15].

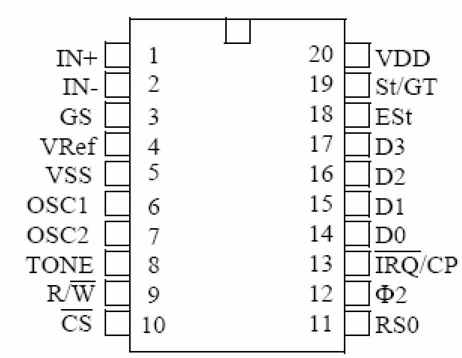

Figure 5-2: MT8880CE Pin Connections.

Table 5-1: Functional Encode/Decode (0 = Logic Low, 1 = Logic High)

F_{LOW}	F_{HIGH}	Digit	D3	D2	D1	D0
697	1209	**1**	0	0	0	1
697	1336	**2**	0	0	1	0
697	1477	**3**	0	0	1	1
770	1209	**4**	0	1	0	0
770	1336	**5**	0	1	0	1
770	1477	**6**	0	1	1	0
852	1209	**7**	0	1	1	1
852	1336	**8**	1	0	0	0
852	1477	**9**	1	0	0	1
941	1336	**0**	1	0	1	0
941	1209	*	1	0	1	1
941	1477	#	1	1	0	0
697	1633	**A**	1	1	0	1
770	1633	**B**	1	1	1	0
852	1633	**C**	1	1	1	1
941	1633	**D**	0	0	0	0

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The MT8880CE communicates through the microprocessor over a 4-bit bus (*D3* to *D0*). It contains an internal counter that provides a burst mode so that the tone burst can be transmitted with precise timing. The chip is capable of generating all 16 pairs of DTMF signals by utilizing a switched-capacitor D/A converter for low distortion and high accuracy DTMF signaling. In addition, the MT8880CE is capable of analyzing call progress tones with it's built in call progress filter. Table 5-1 (above [15]) outlines all the 4-bit bus values with the corresponding digit values and dual frequencies. A list of pin assignments and the description for the 20-pin MT8880CE (shown in Figure 5-2) is provided in Table 5-2 [15].

Table 5-2: MT8880CE Pin Descriptions.

Pin #	Name	Description
1	IN+	Non-inverting op-amp input.
2	IN-	Inverting op-amp input.
3	GS	**Gain Select**. Gives access to output of front end differential amplifier for connection of feedback resistor.
4	V_{Ref}	**Reference Voltage** output, nominally $V_{DD}/2$ is used to bias inputs at mid-rail (see Fig. 13).
5	V_{SS}	Ground input (0 V).
6	OSC1	DTMF clock/oscillator input. Connect a 4.7 MΩ resistor to VSS if crystal oscillator is used.
7	OSC2	Clock output. A 3.579545 MHz crystal connected between OSC1 and OSC2 completes the internal oscillator circuit. Leave open circuit when OSC1 is clock input.
8	TONE	**Tone** output (DTMF or single tone).
9	$R/\overline{W}$	**Read/Write** input. Controls the direction of data transfer to and from the MPU and the transceiver registers. TTL compatible.
10	$\overline{CS}$	**Chip Select**, TTL input ($\overline{CS}$=0 to select the chip).
11	RS0	**Register Select** input. See register decode table. TTL compatible.
12	Φ2	**System Clock** input. TTL compatible. **N.B. Φ2 clock input need not be active when the device is not being accessed.**
13	$\overline{IRQ}$/CP	**Interrupt Request to MPU** (open drain output). Also, when call progress (CP) mode has been selected and interrupt enabled the $\overline{IRQ}$/CP pin will output a rectangular wave signal representative of the input signal applied at the input op-amp. The input signal must be within the bandwidth limits of the call progress filter. See Figure 8.
14-17	D0-D3	Microprocessor Data Bus (TTL compatible). High impedance when $\overline{CS}$ = 1 or Φ2 is low.
18	ESt	**Early Steering** output. Presents a logic high once the digital algorithm has detected a valid tone pair (signal condition). Any momentary loss of signal condition will cause ESt to return to a logic low.
19	St/GT	**Steering Input/Guard Time** output (bidirectional). A voltage greater than V_{TSt} detected at St causes the device to register the detected tone pair and update the output latch. A voltage less than V_{TSt} frees the device to accept a new tone pair. The GT output acts to reset the external steering time-constant; its state is a function of ESt and the voltage on St.
20	V_{DD}	Positive power supply input (+5 V typical).

Finally, the attractiveness of the Zarlink MT8880CE DTMF chip is due to its dual in-line packaging and a maximum dimension of 26.92mm x 7.11mm x 5.33mm. Both its packaging and compact size make this a suitable selection for our RSM.

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5.3.2 Component Implementation

The MT8880CE can be connected to the MCU as per the implementation illustration shown in Figure 5-3 [15].

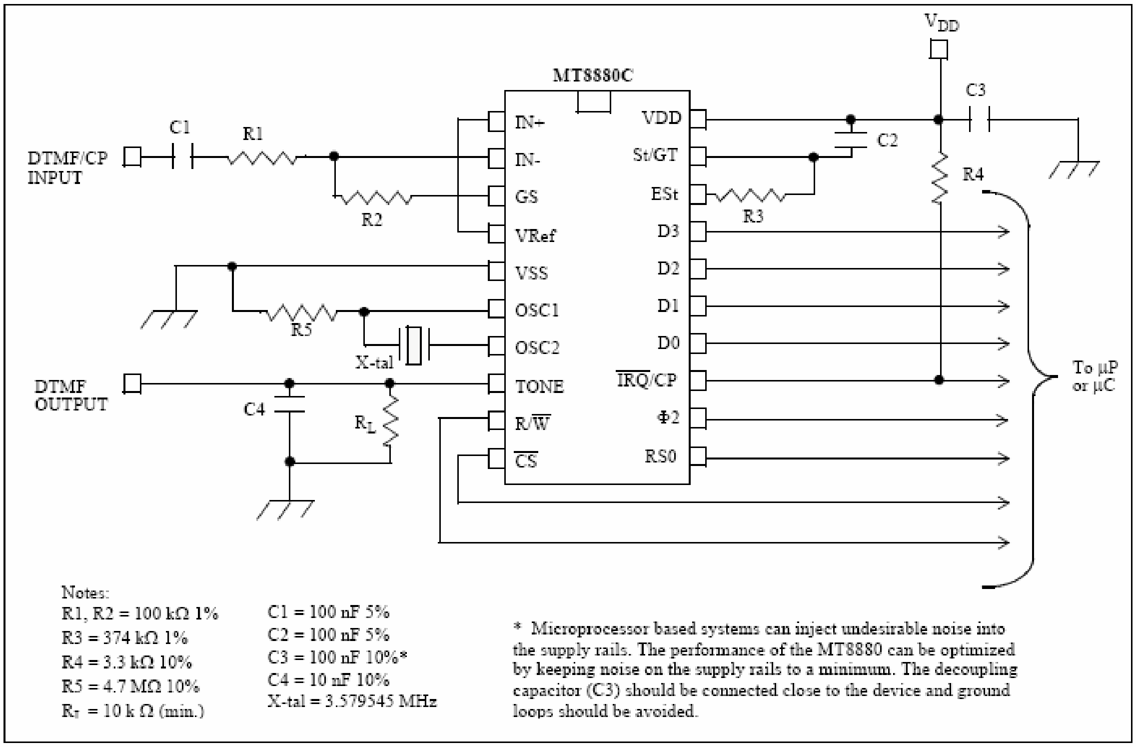

Figure 5-3: MT8880CE Single Ended Input Configuration (µP = Microprocessor, i.e. MCU).

The single-ended input will be the implementation method used with the MT8880CE. Provisions for gain adjustments are made by connecting a feedback resistor between the Gain Select pin, *GS*, and the Inverting Input pin, *IN-*. A check on the validation of the signal duration, with the use of a steering circuit, is required before registration of a decoded tone pair in the internal register can be made. The steering circuit involves the use of an external RC circuit with a time constant driven by the Steering Input pin, *ESt*. The dual tone signals are generated by an internal clock, which requires a standard burst crystal with a resonant frequency of 3.579545MHz. The typical operating supply voltage for the configuration in Figure 5-3 is 5V and is supplied into the V_{DD} pin. The suggested capacitor and the resistor values provided by the data sheet ([15]) will be followed closely to obtain the best results while preventing any damages to the DTMF chip or any other electronic components.

5.4 Digital to Analog Converter

The DAC used in the RSM will be implemented the same way as it is in the BSM. Its detailed component description and implementation was presented in section 3.4. The input to the DAC will come from the MCU and its output (an analog voltage signal) will

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be connected to the DAA before reaching the telephone handset through the *TIP* and *RING* wires.

5.5 Telephone Line Interface

5.5.1 Component Description

The MITEL MH88435AD DAA chip is used for the telephone line interface. Under DTMF operation, the MH88435AD provides a complete interface between audio or data transmission and a telephone line. This chip was chosen due to its low cost and ease of implementation. In addition, this DAA is able to provide isolation and surge protection (for the other electronic components) from the telephone line Figure 5-4 and Figure 5-5 illustrates the pin connection and functional block diagram, respectively [16].

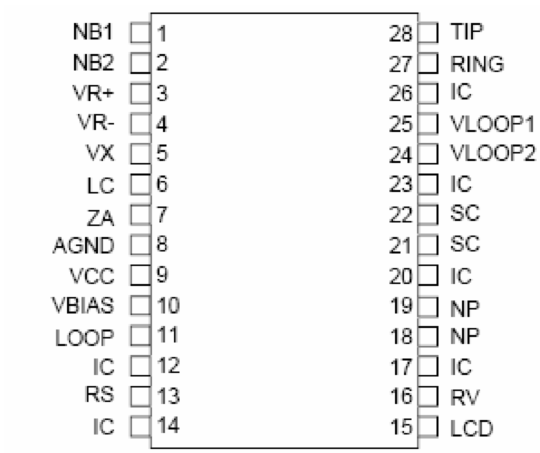

Figure 5-4: MH88435AD Pin Configuration.

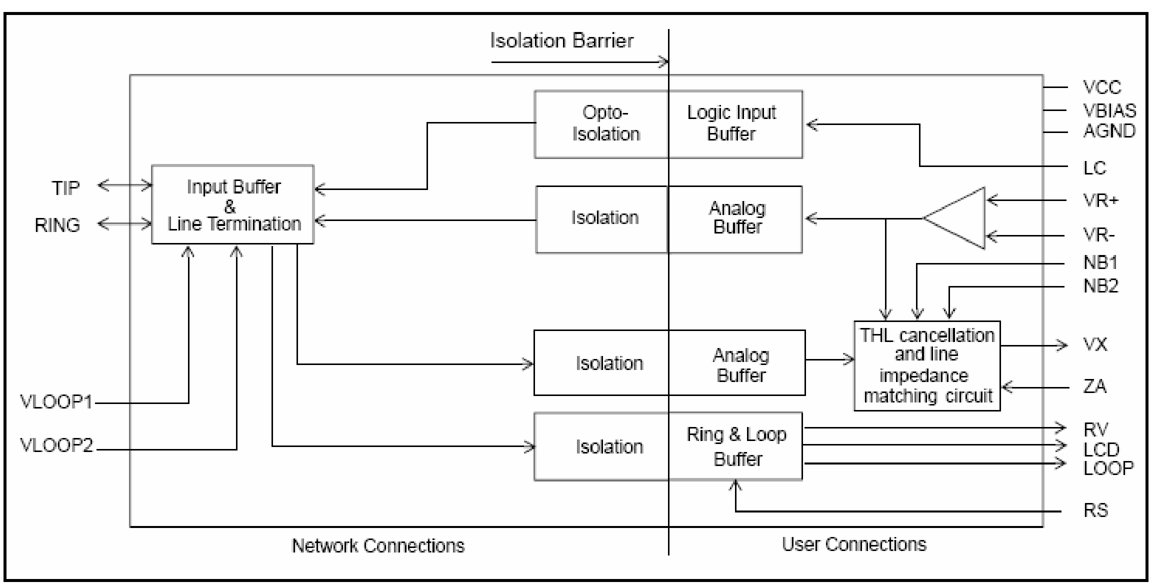

Figure 5-5: MH88435AD Functional Block Diagram.

The features of the MH88435AD include high voltage isolation barrier, 2-to-4 wire conversion, externally programmable line, and network balance impedances. Furthermore, this DAA is capable of variable ring voltage sensitivity. To understand the function of the MH88435AD further, its pin descriptions are presented in the table below [16].

Table 5-3: MH88435 Pin Descriptions.

Pin #	Name	Description
1	NB1	**Network Balance 1.** External passive components must be connected between this pin and NB2.
2	NB2	**Network Balance 2.** External passive components must be connected between this pin and NB1.
3	VR+	**Differential Receive (Input).** Analog input from modem/fax chip set.
4	VR-	**Differential Receive (Input).** Analog input from modem/fax chip set.
5	VX	**Transmit (Output).** Ground referenced (AGND) output to modem/fax chip set, biased at +2.0V.
6	LC	**Loop Control (Input).** A logic 1 applied to this pin activates internal circuitry which provides a DC termination across Tip and Ring. This pin is also used for dial pulse application.
7	ZA	**Line Impedance.** Connect impedance matching components from this pin to Ground (AGND).
8	AGND	**Analog Ground.** 4-Wire ground. Connect to earth.
9	V_{CC}	**Positive Supply Voltage**. +5V.
10	VBIAS	**Internal Reference Voltage.** +2.0V reference voltage. This pin should be decoupled externally to AGND, typically with a 10μF 6.3V capacitor.
11	LOOP	**Loop (Output).** The output voltage on this pin is proportional to the line voltage across Tip - Ring, scaled down by a factor of 50.
12, 14, 17, 20, 23, 26	IC	**Internal Connection.** No connection should be made to this pin externally.
13	RS	**Ringing Sensitivity.** Connecting a link or resistor between this pin and LOOP (pin 11) will vary the ringing detection sensitivity of the module.
15	LCD	**Loop Condition Detect (Output).** Indicates the status of loop current.
16	RV	**Ringing Voltage Detect (Output).** The RV output indicates the presence of a ringing voltage applied across the Tip and Ring leads.
18, 19	NP	**No Pin.** Isolation barrier, no pin fitted in this position.
21, 22	SC	**Short Circuit.** These two pins should be connected to each other via a 0Ω link.
24	VLOOP2	**Loop Voltage Control Node 2.** Used to set DC termination characteristics.
25	VLOOP1	**Loop Voltage Control Node 1.** Used to set DC termination characteristics.
27	RING	**Ring Lead.** Connects to the "Ring" lead of the telephone line.
28	TIP	**Tip Lead.** Connects to the "Tip" lead of the telephone line.

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5.5.2 Component Implementation

Figure 5-6 illustrates a typical application circuit for the MH88435AD DAA chip [16]. Instead of placing a dummy ringer load across the *TIP* and *RING* wires as suggested in the data sheet, however, we've attached a working ringer voltage generator.

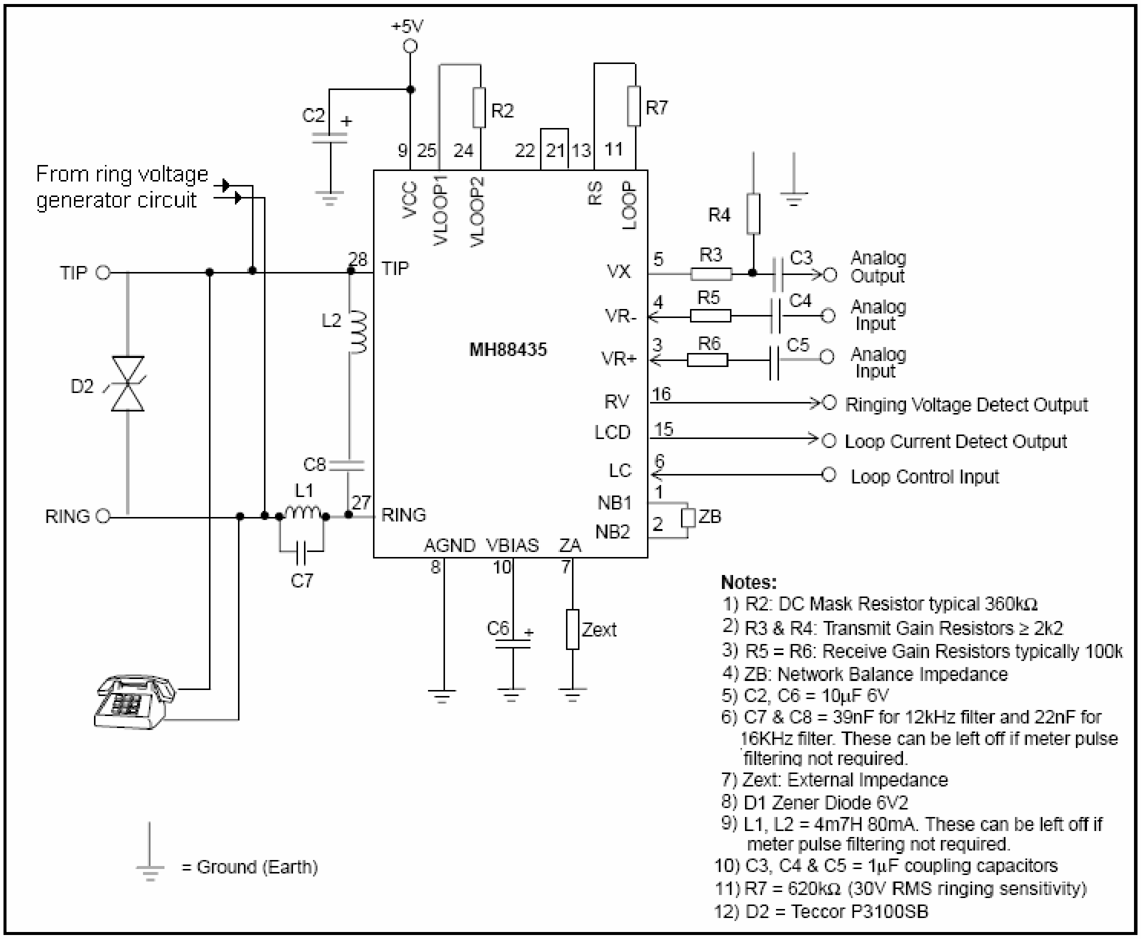

Figure 5-6: MH88435AD Application Circuit.

The MH88435AD's analog output shown in the figure above will be connected to the input of the MT8880CE configuration shown in Figure 5-3. The differential analog input in Figure 5-6 will be supplied by voice signals (during a call) and call progress tones (while dialing). The *TIP* and *RING* wires, shown in Figure 5-6, correspond to the two-wire telephone connection and will be connected to a standard RJ11 telephone jack. Aside from the slight modification stated above, we will follow this circuit configuration closely to prevent damage to the DAA chip or other electronic components.

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5.6 Call Progress Generator

5.6.1 Component Description

The M991 IC call progress generator from CLARE is shown in Figure 5-7 with its block diagram shown in Figure 5-8 [17]. The M991 is chosen because it is a single purpose chip that is small in size [17], which is extremely suitable for use in our system.

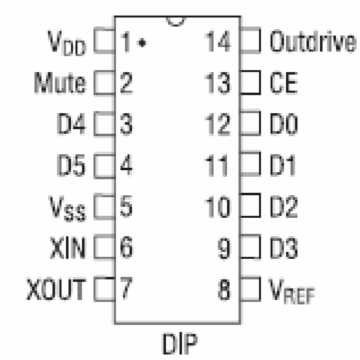

Figure 5-7: M991 Pin Assignments

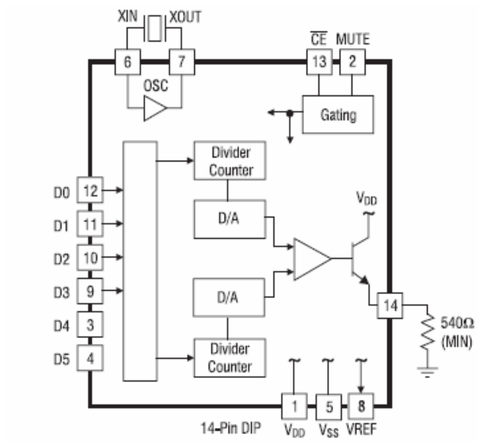

Figure 5-8: M991 Block Diagram

Call progress tones are audible tones normally sent from switching systems to telephones to indicate the status of calls to the calling party. The M991 a 14-pin PDIP circuit that is capable of generating these unique frequencies (singly or in pairs), which are common to call progress signals. In addition, the M991 is digitally controlled and highly linear due to the use of low-power CMOS techniques. Table 5-4 shows the pin functions of the M991 [17].

Table 5-4: M991 Pin Functions

Pin	Function
CE	Latches data and enables output (active low input).
D0 - D3	Data input pins. (See Data/Tone Selection.)
D4-D5	Leave open.
MUTE	Output indicates that a signal is being generated at OUTDRIVE.
OUTDRIVE	Linear buffered tone output.
V_{DD}	Most positive power supply input pin.
V_{REF}	Internally generated mid-power supply voltage (output).
V_{SS}	Most negative power supply input pin.
X_{IN}	Crystal oscillator or digital clock input.
X_{OUT}	Crystal oscillator output.

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Table 5-5 shows the commonly used call progress tones and the proper bit values for duration and frequency selections of that particular tone [17]. A reference table for standard call progress tones is shown in Table 5-6, which allows us to compare the actual behavior of the chip to this chart for verification purposes [17].

Table 5-5: M991 Data/Tone Selection

D0	D1	D2	D3	Frequency (Hz)		Use
				1	2	
0	0	0	0	350	440	Dial Tone
0	0	0	1	400	off	Special
0	0	1	0	440	off	Alert Tone
0	0	1	1	440	480	Audible Ring
0	1	0	0	440	620	Pre-empt
0	1	0	1	480	off	Bell high tone
0	1	1	0	480	620	Reorder(Bell low)
0	1	1	1	350	off	Special
1	0	0	0	620	off	Special
1	0	0	1	941	1209	DTMF "*"

Table 5-6: Standard Call Progress Tones.

Tone Name	Frequency (Hz)		Interruption Rate
	1	2	
Dial	350	440	Steady
Reorder	480	620	Repeat, tones on and off 250 ms ± 25 ms each.
Busy	480	620	Repeat, tones on and off 500 ms ± 50 ms each.
Audible Ring	440	480	Reat, tones on 2 ± 0.2 s, tones off 4 ± 0.4 s
Recall Dial	350	440	Three bursts tones on and off 100 ms ± 20 ms each followed by dial tone.
Special AR	440	480	Tones on 1 ± 0.2s, followed by single 440 Hz on for 0.2s on, and silence for 3 ± 0.3 s, repeat.
Intercept	440	620	Repeat alternating tones, each on for 230 ms ± 70 ms with total cycle of 500 ± 50 ms.
Call Waiting	440	Off	One burst 200 ± 100 ms
Busy Verification	440	Off	One burst of tone on 1.75 ± 0.25 s before attendant intrudes, followed by burst of tone 0.65 ± 0.15 s on, 8 to 20 s apart for as long as the call lasts
Executive Override	440	Off	One burst of tone for 3 ± 1 s before overriding station intrudes
Confirmation	350	440	Three bursts on and off 100 ms each or 100 ms on, 100 ms off, 300 ms on

5.6.2 Component Implementation

Referring to Figure 5-8, the MCU will be connected to the call progress generator via pins *D0* to *D3* to trigger the generation of the desired tones. Pins D4 and D5 must be left open, however. The output of the M991 will be connected to the input of the DTMF chip (Figure 5-3). This chip will be powered by +5V DC.

5.7 Ring Voltage Generator

5.7.1 Component Description

The ring voltage generator is a low cost switching supply circuit designed using common electronic components that is responsible for producing the necessary high voltage to trigger the ringer in a normal telephone. This voltage is usually between 40 and 150 Volts DC at a frequency of 20 – 40Hz [18]. We have designed our circuit to produce 75V DC at the output at 30Hz when it is connected (i.e. when there is an incoming call). Figure 5-9 shows the circuit schematic (adopted from Bill Bowden's design [19]), with value modified from the original drawing to yield different ringer interval values and input voltage.

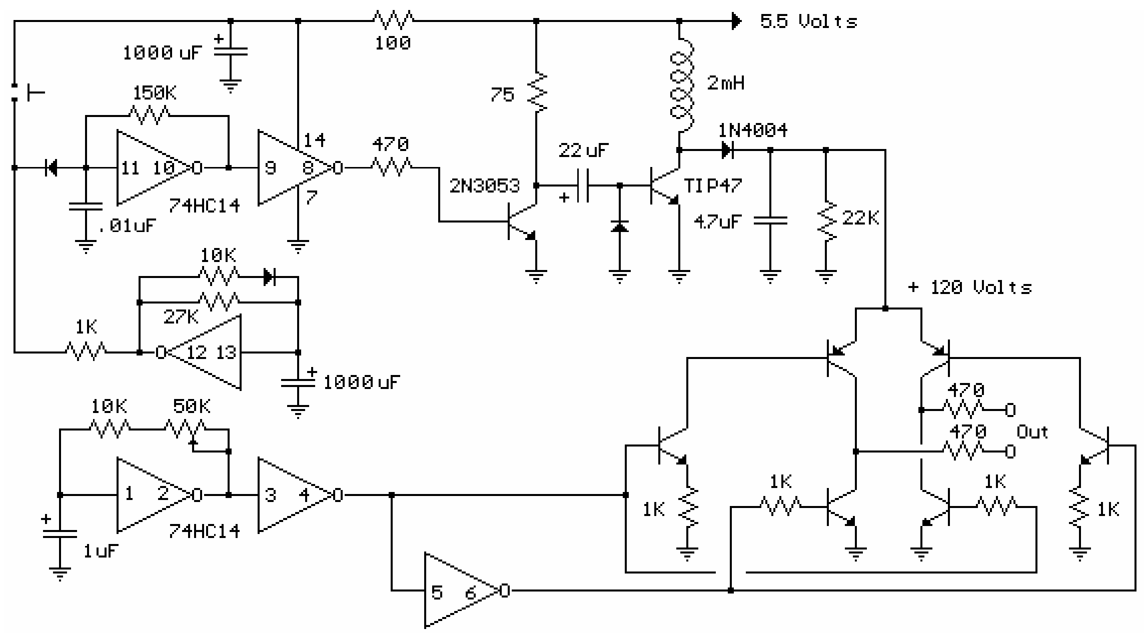

Figure 5-9: Circuit diagram for ringer voltage generator using switching supply.

5.7.2 Component Implementation

The circuit will be activated by the MCU. The MCU output pins are capable of generating 5V [7]. Since we require 5.5V, as shown in the above figure, an operation amplifier (LM324) will be used to amplify the voltage. The outputs shown in the above figure will be connected directly to the *TIP* and *RING* wires shown in Figure 5-6.

6 System Event Flow

The subsections below detail the various high level system event flows of each module and the event interactions between the modules. These designs will help with the low level assembly programming that will commence shortly after this design specification is completed. The flow diagrams will also provide the testing procedures when the microprocessor programming is complete.

6.1 Base Station Module

6.1.1 Idle State

A cell phone status flag is kept inside the MCU to monitor the existence of the mobile phone. When the system is first powered, the BSM assumes that no cell phone is plugged in and the BSM state is set to *NO_MOBILE_PHONE*. The microcontroller then tries to send a command to the mobile phone to request its power status. If the cell phone replies with an *OK* status bit, it means that there is a cell phone connected to the BSM and it is powered on. Upon receiving the positive acknowledgement from the cell phone, the BSM enters into the IDLE state. The IDLE state denotes that a cell phone has been connected and the BSM is awaiting the next instruction.

6.1.2 Incoming Call Connection

This scenario shows the procedures of receiving and answering incoming cell phone calls. The successful establishment of a communication channel between mobile phone and home phone depends on the status of the RSM and the WTM. It requires both modules to be in *IDLE* state ready for receiving commands.

Data exchange between the MCU and the cell phone is done by the built-in *Universal Synchronous Asynchronous Receiver Transmitter* (USART). The baud rate of the USART is set to 9600 Hz to agree with the GSM modem clock rate inside the mobile phone. Data is transmitted in unit of packets. Each packet has 10 bits with the first and last bit serving as the starting bit and stopping bit, respectively. The middle 8 bits are the information bits. Whenever the MCU has finished receiving one packet, an interrupt flag bit is set by the hardware, which causes the interrupt routine to run. Inside the interrupt routine, the received packet is compared with pre-defined masks and the associated interrupt handler is called.

When the mobile phone receives an incoming call request, its GSM modem will send a *RING* notification to the BSM. At this stage, the mobile phone is assumed to be connected with the MCU and the BSM is in its *IDLE* state. Upon receiving the *RING* indication, the BSM switches from *IDLE* state to *INCOMING_CALL* state. A packet is also sent to the RSM indicating an incoming call request along with the BSM state. The RSM is expected to return the proper acknowledgment (*on-hook* or *off-hook*). If the reply from the RSM shows that the landline phone is *off-hook* (i.e. the handset has been picked up before the incoming call), the BSM will continue to send the incoming call request to

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the RSM until the caller stops calling or until the user places the handset back to the *on-hook* position. If the RSM returns with the acknowledgement that the home phone is *on-hook*, then the RSM is ready to receive calls.

If the first acknowledgement indicates that the landline communication device is *on-hook*, the BSM changes the state from *INCOMING_CALL* to *AWAITING_ANSWER* state. When the RSM replies again after another state inquiry which indicates the telephone is still *on-hook*, the BSM, in the *AWAITING_ANSWER* state, will send an AT command to the cell phone to indicate that the Docking Station is ready for voice communication. The ADC is now turned on and the voice communication channel is now created. Both BSM and RSM then enter the *ACTIVE* state (the state where a communication is established).

If the caller disconnects the call before the user picks up, a *TERMINATE* packet is sent to the MCU via the USART. Upon receiving this indication, the BSM will forward this message to the RSM and change its own state from *AWAITING_ANSWER* to *IDLE*.

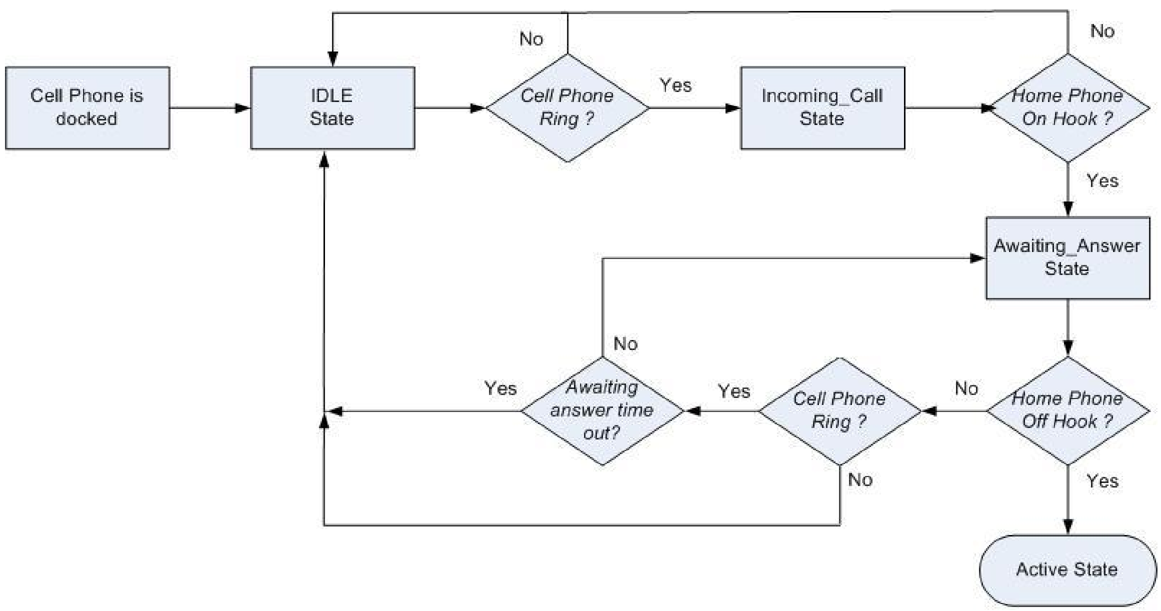

Figure 6-1: Flow Chart of Incoming Call Algorithm

6.1.3 Outgoing Call Connection

The outgoing call procedures are similar to that of the incoming call except the direction is reversed. Whenever there is a data packet from the RSM, the *DATA_READY* flag is set by the WTM. The BSM periodically monitors this flag to catch any data that is available. Pre-defined bit-map masks are adopted to compare the different statuses. If an outgoing call request is received, the BSM immediately answers with a packet indicating the connection status of the mobile phone given that the BSM is in *IDLE* state. The BSM then waits for the dialing strings (DS) from the RSM. The DS is a string of numbers that the user inputs to make an outgoing call (i.e. 6042914400).

The BSM state is not changed until the DS is provided by the RSM. A packet indicating the starting of string transfer is expected by the BSM. The packets that follow are treated as strings and forwarded to the mobile device until the ending of string transfer packet is obtained. The ADC is then turned on and the BSM is changed into the *DIALING* state. If the call is accepted by the intended recipient, a *CONNECT* packet is transmitted from the cell phone GSM modem to the MCU. The BSM is then switched from *DIALING* to *ACTIVE* state. The *ACTIVE* state indicates that a conversation is currently in progress.

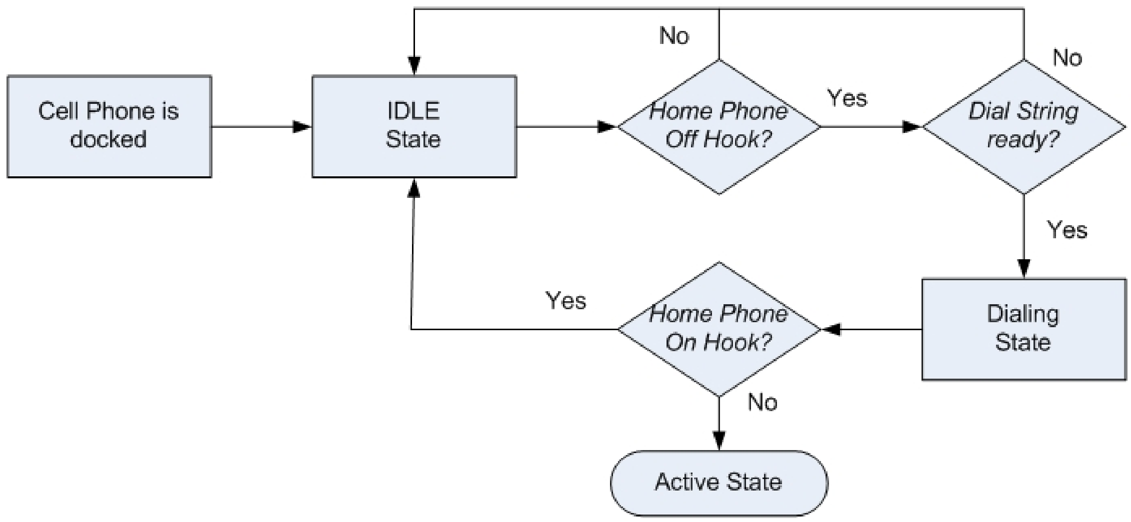

Figure 6-2: Flow Chart of Outgoing Call Algorithm

6.1.4 Voice Communication

The PICF16877A microcontroller's pin 2 (bit 0 of PORT A) is reserved as an analog input pin. During communication, voice signals are extracted from the voice output pin of the mobile phone and fed into pin 2 of the MCU. The built in ADC is sampling the analog signal at a specified conversion rate (mentioned in section 4.2.1).

The voice data from the RSM is stored in the SPI receive buffer and its contents are also updated every 1/8K sec. Voice data is written to PORT D of the MCU and then converted back to analog signal with the aid of external digital-to-analog converter. The output of the DAC is connected with the voice input of the mobile phone.

The communication cycle repeats continuously until a *TERMINATE* packet is received from either the RSM or the mobile phone. If the packet is sent from the cell phone, BSM will forward the message to the RSM and switches back to IDLE mode. On the other hand, if the packet sent is from the RSM, an AT command is issued to the mobile device to terminate the voice channel and the BSM is set to the *IDLE* state again.

Table 9-1 to Table 9-4 in the Appendix describes in detail the binary bit values assigned for the states and flags mentioned above.

6.2 Wireless Transmission Module

6.2.1 Time Division Multiplexing

The objective of time division duplex (TDD) is to utilize the half duplex wireless link in a way that will appear to the user as a full duplex link. The half duplex link, which can only transmit or receive at a given time, must switch between transmit and receive mode fast enough and often enough to appear to the user as a full duplex link. The figure below shows the timing in which the base and receiver RF modules switches between transmit and receive modes to achieve TDD. As shown, the wireless link switches between the two states – transmitting from base to receiver station and transmitting from receiver to base station. A short period of guard time is also added in between the two states when wireless transmission is in opposite directions. This guard time is represented by the overlap when both base and receiver stations are in receive mode. It has been tested and verified that safe operation is possible when both wireless units are in receive mode. However, it has not been tested when both wireless units are in transmit mode. However, we can intelligently conclude that if both units are transmitting, there will be packet loss because no transmitted packets are being received. In addition, jamming may occur if both units are transmitting at the same time without any receivers.

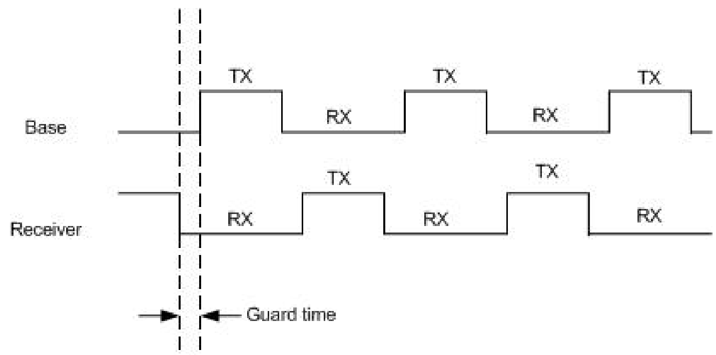

Figure 6-3: Base and receiver TDD receive and transmit mode switching scheme. Horizontal axis represents the time axis.

Another form of illustration would help understand the concept of TDD further. The figure below shows the TDD event flow during a voice call, with the voice packet payload being 10 ms of voice data for example purposes. The base station and receiver station alternates in sending voice data back and forth, switching the send direction every 5 ms or half a packetization time. We start off the example with the base station being the transmitting side. The restriction is that the time to transmit the voice packet must be less than half the packetization time (or 5 ms in this case). The transmitting side (base station) will then switch to receive mode after packet transmission is complete, forming the guard time as indicated by the green shaded portions in the diagram below. The receiving side (receiver station) will receive the voice data into a buffer during the guard time, and then switch to transmit mode after it accumulates 10 ms of voice data, ending the guard time. Now, the wireless link transmit direction has switched fully, from base sending to receiver station to receiver sending to base station. This cycle of voice alternation wireless link transmit direction continues as long as there is a call in progress.

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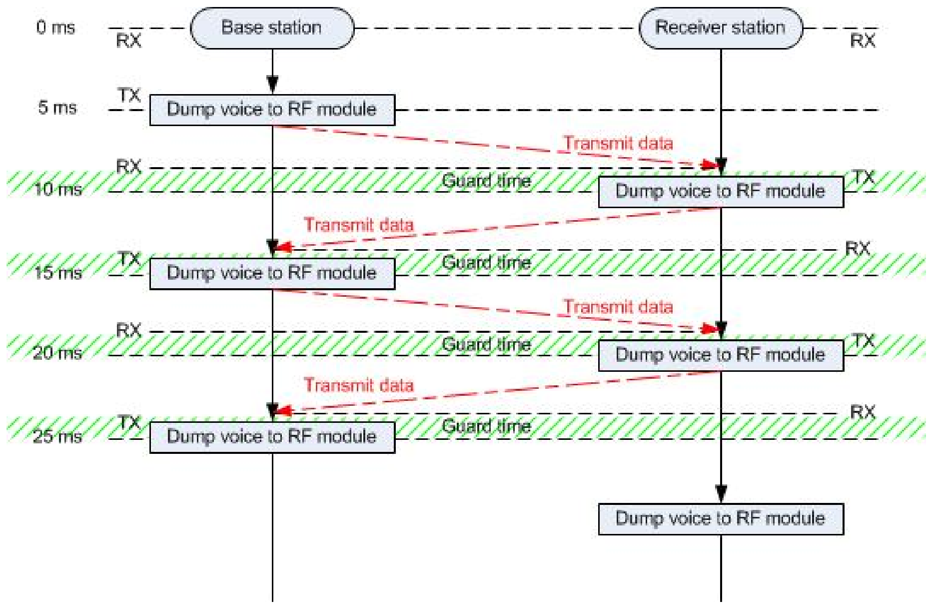

Figure 6-4: Base and receiver TDD event flow during a voice call. Vertical axis represents the time axis. Voice packet payload is 10ms of voice data in this case for example purposes.

6.2.2 Program Flow

Figure 6-5 is the program flow chart of the wireless module. Upon power up of the system, the RF module settings and SPI interface settings are initialized along with other miscellaneous initialization steps.

The wireless module will first check to see if 10 ms of voice data has been accumulated yet. Once the 10 ms voice data packet payload has been accumulated, the wireless module performs the tasks as represented by the “Dump voice into RF module” block in Figure 6-4. The tasks include setting the RF modules to transmit mode, dumping the packet payload into the RF module via SPI interface, and clearing the data buffer in the MCU. All these are done with interrupts disabled to ensure the packet payload transfer does not get interrupted. Whether a voice packet or an instruction packet is transmitted depends which payload type is available, but instruction payload type has higher priority than voice payload type. If both payload types are available, instruction payload will be transmitted and the voice payload will be ignored.

Now that the packet is transmitted, the wireless module will check to see when the packet transmission is finished, or in other words checking when the TX FIFO of the RF module is empty. When packet transmission is finished, the RF module is switched to receive mode, again with the interrupts disabled. This initiates the beginning of the guard time until the other side of the wireless link switches to transmit mode.

Next, the wireless module will check when the RF module has received the packet. When the packet is received, the payload data will be dumped to the MCU buffer via SPI interface, again with the interrupts disabled. After receiving from the other side of the

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wireless link, the wireless module is now ready to loop back to beginning of the flow chart and check for when it should transmit another packet.

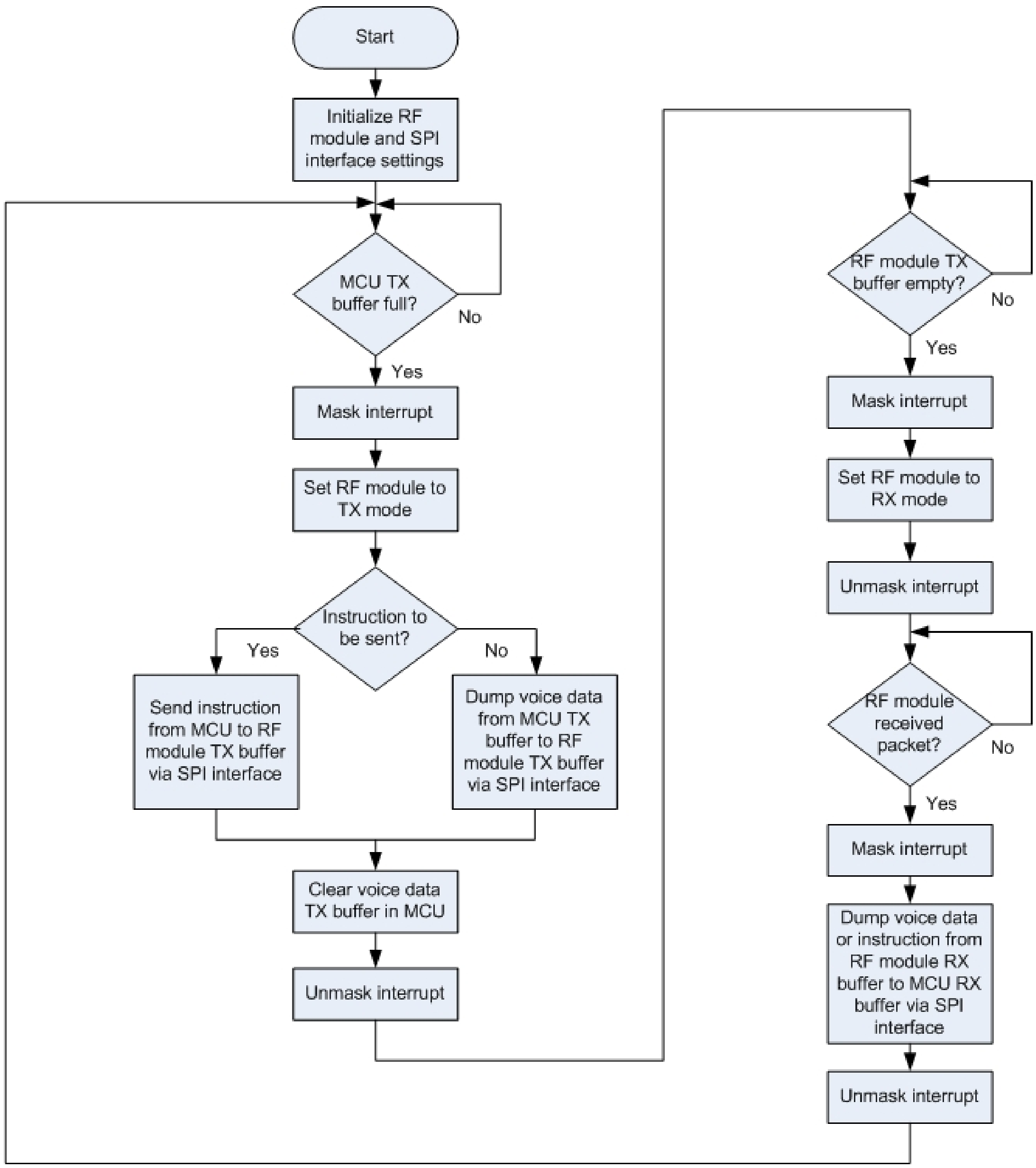

Figure 6-5: Flow chart of the wireless module program flow.

6.3 Receiver Station Module

6.3.1 Incoming Call Event Flow

The incoming call event flow is illustrated in Figure 6-6. Assuming the home phone is *on- hook*, the RSM will be in IDLE state and will continuously monitor for incoming calls. When a call arrives, the ringer will be turned on and the microprocessor awaits the user to pick up the telephone. When the handset is lifted, the ringer shuts off and the RSM enters the *TALKING* state, allowing the user to engage in a normal conversation. Two factors will cause the RSM to return to the *IDLE* state; if the user hangs up and places the handset *on-hook* or if the cell phone disconnects either through lost of reception or if the opposing party hangs up.

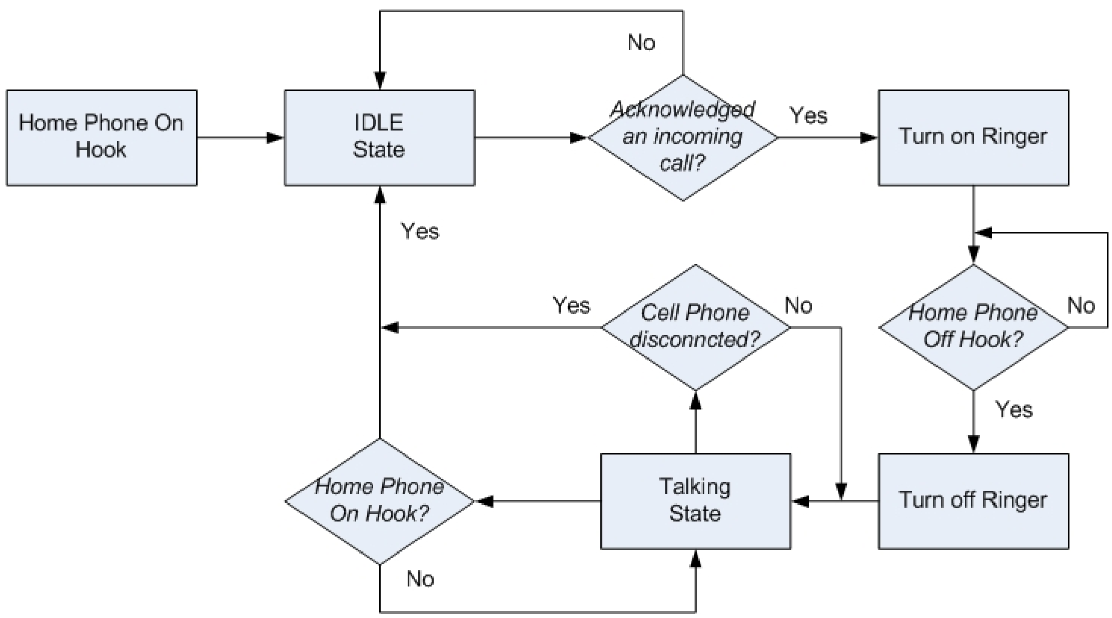

Figure 6-6: RSM incoming call event flow.

6.3.2 Outgoing Call Event Flow

The outgoing call event flow is illustrated in Figure 6-7. This event occurs if a user wishes to dial an outgoing call. The phone first needs to be *on-hook* to ensure it is in the *IDLE* state. When an *off-hook* is detected and there are no incoming calls at the same time, the dial tone will be generated for the user. A *dial_tone_timeout* counter is automatically started. Once this timer reaches 0, the dial tone switches over to an *off-hook warning tone*. This is analogous to if a user accidentally knocks off their telephone accidentally. Before the *dial_tone_timeout* is reached, if a key is pressed, the telephone switches to normal operation and continually accepts the key presses, generating the corresponding DTMF tones for the user.

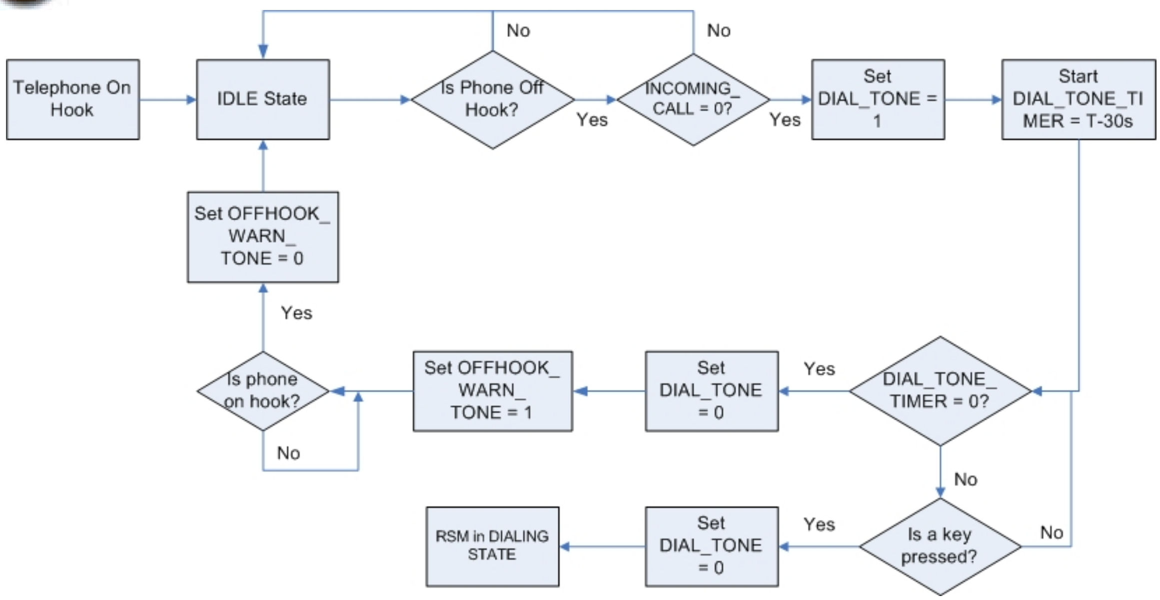

Figure 6-7: RSM outgoing call event flow.

6.3.3 Voice Communication

During communication, voice signals are extracted from the analog output pin of the DAA and fed into pin 2 (the reserved analog input pin for the PICF16877A) of the MCU. The built in ADC is again sampling the analog signal at a specified conversion rate (mentioned in section 4.2.1).

The voice data from the BSM is stored in the SPI receive buffer and its contents are also updated every 1/8K sec. Voice data is written to PORT D of the MCU and then converted back to analog signal with the aid of external DAC. The output of the DAC is connected to the DAA and then transferred to the *TIP* and *RING* lines of the telephone.

A *TERMINATE* packet received from the BSM, the mobile phone, or an *on-hook* flag will cause the system to return to *IDLE* state.

7 Conclusion

This design specification of Websa Technology's Wireless Cell Phone Docking Station provides an idea of how our developing team hopes to implement and construct each module: Base Station Module, Wireless Transmission Unit, and Receiver Station Module. This document also outlined the architectural layout, selected components, and implementation procedures to provide a more comprehensive insight of the low-level design. All designs were carefully thought out and specifically chosen to meet the requirements set by the Functional Specifications. Upon completion of this project in April 2006, our developing team is confident that our prototype, the *Wireless Mobile-Dock*, will satisfy the minimum functional requirements using the designs presented in this document.

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9 Appendix

Table 9-1: **Base Station Module Data Packet Format**

7	6	5-3	2-0
MIB	--	CS	BSMS

Table 9-2: Bit Representation of Status and States

MIB: Module Indication Bit	CS: Cell-Phone Status	BSMS: Base Station Module State
0: Cell Phone Module	0: OK	000: NO_MOBILE_PHONE
1: Landline Phone Module	1: CONNECT	001: IDLE
	2: RING	010: INCOMMING_CALL
	3: NO CARRIER	011: WAITING_ANSWER
	4: ERROR	100: DIALING
	5: NO DIALTONE	101: ACTIVE
	6: BUSY	
	7: NO ANSWER	

Table 9-3: 8-bits Binary Representation of Cell Phone Status

Cell Phone Status	Binary	Mask
OK	00000000	11111111
CONNECT	00000001	11111110
RING	00000010	11111101
NO CARRIER	00000011	11111100
ERROR	00000100	11111011
NO DAILTONE	00000101	11111010
BUSY	00000110	11111001
NO ANSWER	00000111	11111000

Table 9-4: 8-bits Binary Representation of Base Station Module State

BSM State	Binary	Mask
NO_MOBILE_PHONE	00000001	11111110
IDLE	00000010	11111101
INCOMMING_CALL	00000100	11111011
DIALING	00001000	11110111
WAITING_ANSWER	00010000	11101111
ACTIVE	00100000	11011111

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