AN064
Two Way Audio Communications using the CC2510

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Keywords

- *CC2510*
- *Two Way Audio*
- *Frequency Hopping*

Introduction

This note describes a two-way audio communications link between a 'Master' and a 'Slave', based on the Texas Instruments CC2510 System-on-Chip Low Power RF Transceiver. It operates in the 2.4 GHz ISM/SRD band. It has been optimized for low cost, using a PCB (Printed Circuit Board) folded dipole antenna and a minimum of external components. Several key features of the CC2510 are employed, including the internal ADC, the DSM (Delta-Sigma-Modulator), and (optionally) ulaw compression/expansion algorithm implemented in hardware. These features eliminate the need for an external audio codec.

While the card described herein is based on the CC2510 and operates in the 2.4 GHZ band, both the card and the software can be easily modified for use with a CC1110 operating in the 433 MHz, 868 MHz, or 915 MHz bands. Project collateral discussed in this application note can be downloaded from the following URL: http://www.ti.com/lit/zip/SWRA225.

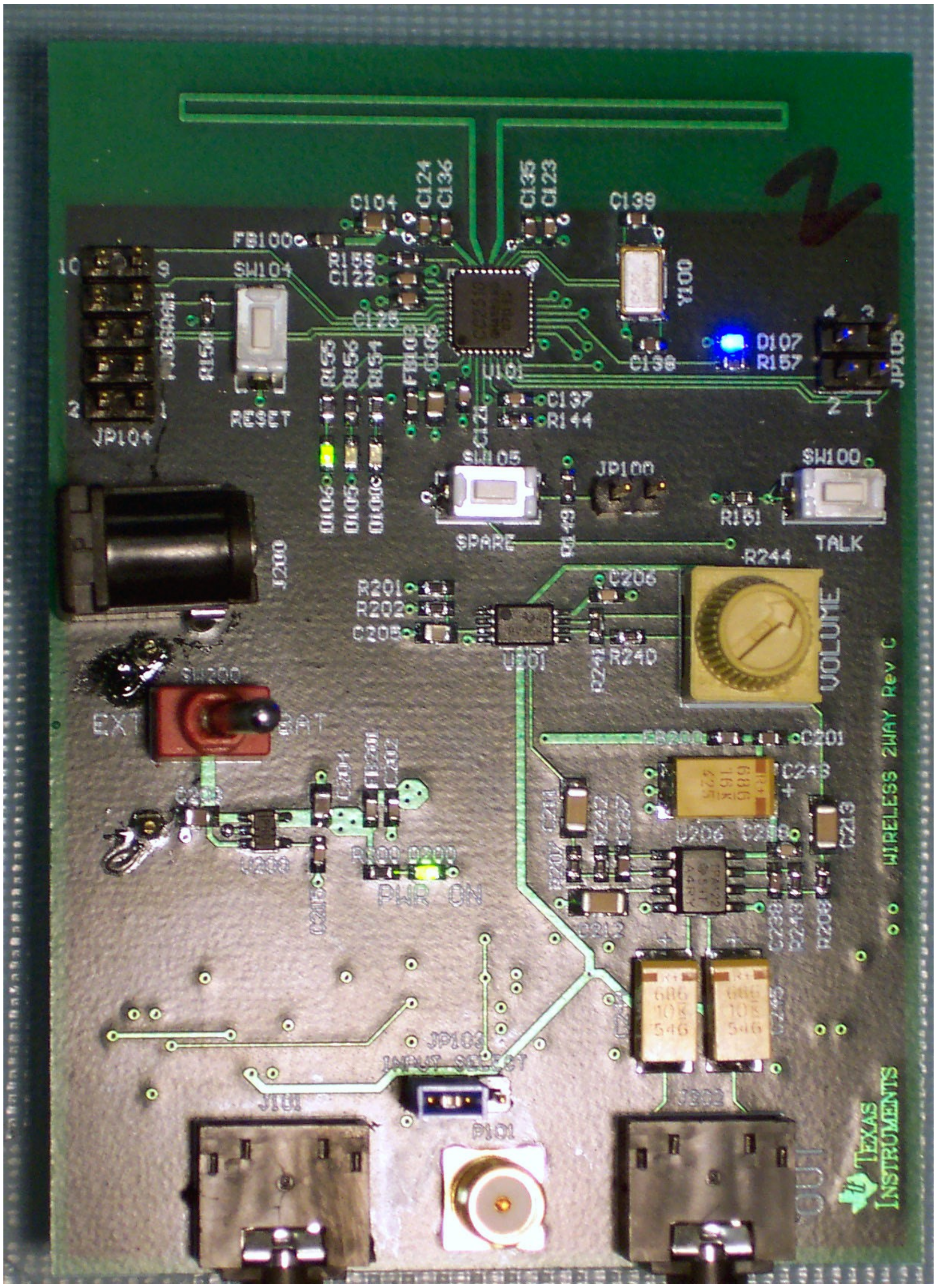


Figure 1 – Two Way AudioTransceiver

Brief Description

The CC2510 can be configured to perform μ-law compression and expansion. Using μ-law compression, the ADC samples are compressed from 11 to 8 bits, thereby reducing the required number of bytes in a 'packet' and the transmitted data rate, as compared to a system not using μ-law encoding. However, the SNR (Signal to Noise Ratio) and quality of the audio will not be as good when compression is used. Software is available for both compressed (μ-law enabled) and uncompressed (μ-law disabled) modes. The table below shows parameters with compression both enabled and disabled. In 'μ-law disabled' mode two 11 bit ADC samples are packed into two bytes.

Parameter	μ-law Enabled	μ-law Disabled
ADC Samples per Packet	54	60
Packet Length (bytes)	61	92
RF Data Rate (kbps)	250	300
Period Between Packets (msec)	6.75	7.5

Note: Throughout the remaining document, the descriptions given pertain to the 'ulaw enabled' code.

At both the Master and the Slave, audio is sampled at an 8 kHz rate (i.e., every 125 usec), using the CC2510's ADC set for a resolution of 11 bits. Every 54 samples (6.75 msec), the data is transferred to the TX Buffer, and the transmitter enabled. Similarly, a received sample must be entered into the CC2510s Timer 1. The Master controls system timing: it will blindly transmit a packet containing the last 54 audio samples every 6.75 msec. Following each transmission, the Master will listen for a response from the Slave, which should respond with a packet containing the last 54 audio samples that it has collected. The Slave listens for a transmission from the Master; should it 'hear' the transmission, it will transfer the data received from the Master into its 'playback' buffer and subsequently transmit its ADC samples back to the Master.

The Slave has two modes: 'Waiting for Beacon' and 'Paired'. In 'Waiting for Beacon', the Slave continuously listens on channel 0 (see 'Frequency Hopping', below) for a transmission from the Master. While in 'Paired' mode, the Slave expects to receive a packet of audio data every 6.75 msec.

The card has been designed for use with a headset that contains both a condenser microphone and a headphone, such as the Sennheiser PC131.

Detailed Description - Hardware

Refer to the block diagram, Figure 2. The Texas Instruments CC2510 System-on-Chip Low Power RF Transceiver is used with a half wave dipole antenna etched onto the printed circuit board. Audio from the microphone is amplified to a maximum peak-to-peak voltage of 3 volts using one section of a LMV324 operational amplifier. The gain of this amplifier is controlled by the CC2510 via analogue switches (TSA3157), and can be set to a value of 20, 40, 60, or 80 (20, 26, 32, and 38 dB). After amplification, the audio is filtered by a 6 pole Chebychev low pass anti-aliasing filter having a cut-off frequency of 3 kHz. This filter connects to the CC2510's ADC.

Audio is obtained from the CC2510's Timer 1, used in Delta-Sigma Modulation (DSM) mode. The audio output is further filtered using a 6 pole Chebychev low pass filter identical to that used in the audio input path. A TPA122 150 mW Stereo Audio Power Amplifier is used to drive the headphone.

The card may be powered from either the three AAA batteries attached to the back of the card or from an external 5 volt DC power source. A TPS73033 Low-Noise, High PSRR, RF 200-mA Low-Dropout Linear Regulator is used to obtain the required 3.3 volts.

Included on the card are five LEDs (Light Emitting Diodes), two push button switches, and a Reset switch. Currently, the two push buttons are not used. The five LEDs have the following functions:

Blue: Heartbeat, Flashes twice per second as long as the CC2510 program is running.
Green (D200): Power On
Red: Timeout Error. No data was received from a Slave (Master) or from the Master (Slave).
Orange (Slave only): Waiting to 'hear' the Master's Beacon signal
Green (D106) (Slave only): Paired with a Master

Detailed Description - Software

Due to the time critical nature of streaming audio, the supporting software code used for this project is application specific; that is, it is not based on any standard protocol.

Figure 3 is a flow chart of the Master's main program loop. Note that the 'Frame Ready' flag is set by the audio handler, an interrupt driven routine (executed once every 125 microseconds) that both reads ADC samples and places them in an available buffer and transfers audio samples received from the Slave into the Timer 1 'count' register. 'Frame Ready' is set to True every 54 samples (6.75 msec).

Figure 4 is a flow chart of the Slave's main program loop. Its timing is based on a 'Frame Timer' (T2). This is a 'count down' timer, which is initialized to a value of 229 tics, corresponding to a period of 6.75 msec, after a packet is received from the Master. Two variations of a 'Receive A Packet' subroutine are used. Subroutine 'ListenforMaster' is used while in 'waiting for beacon' mode, and will return immediately after receiving a packet, or after the specified 'timeout' period (27 milliseconds) if no packet is received. Subroutine 'rfReceivepacket' is used in 'paired' mode, and will return after the specified time period (1438 usec) if a packet is not received. If, after the specified time period a packet is being received (SYNC word was detected), the subroutine will not return until the entire packet has been received.

Frequency Hopping

A set of four channels (frequencies) is used, on a rotating basis (every transmission is on a different channel). Currently the four channels used are 0, 13, 26, and 39. The base frequency is set to 2406.0 MHz and the channel spacing to 250 kHz, so channels 0, 13, 26, and 39 correspond to frequencies of 2406.000, 2409.250, 2412.500, and 2415.750 MHz, respectively. When first turned on and after 4 consecutive packets have been lost, the Slave goes into a 'waiting for beacon' mode, continuously listening on channel 0 (2406.000 MHz).

Lost Packets\Latency

There are several ways in which a 'lost packet' can be handled, including both simple (mute the audio during the duration of the lost packet) and complex (e.g., interleaving data across three consecutive packets). The idea behind the interleaved data approach is that should a packet be lost, the missing data can be re-created via interpolation, based on correctly received samples in other packets.

Experimental results suggest that a simple muting approach is the least audibly objectionable alternative. If a packet is lost, the audio is simply muted for the duration of the packet. Interpolation of missing data using any practical algorithm invariably leads to audible ‘clicks’ and ‘pops’. A further advantage of this simple approach is that latency (the time delay between when audio is sampled by the ADC and when that sample is ‘played back’) is minimized. For this implementation, latency is 13.5 msec, short enough that it is not audibly detectable. Note that local audio feedback is provided; e.g., each ADC audio sample is mixed with a sample received from the external card. This allows a user to hear him of her self in the headphones as well as the other user.

System Timing\Other Data Rates

Sampling rates other than 8 kHz are possible, within the limitations of the CC2510s maximum data rate (500 kbps), maximum packet length (255 bytes), and ADC conversion time.

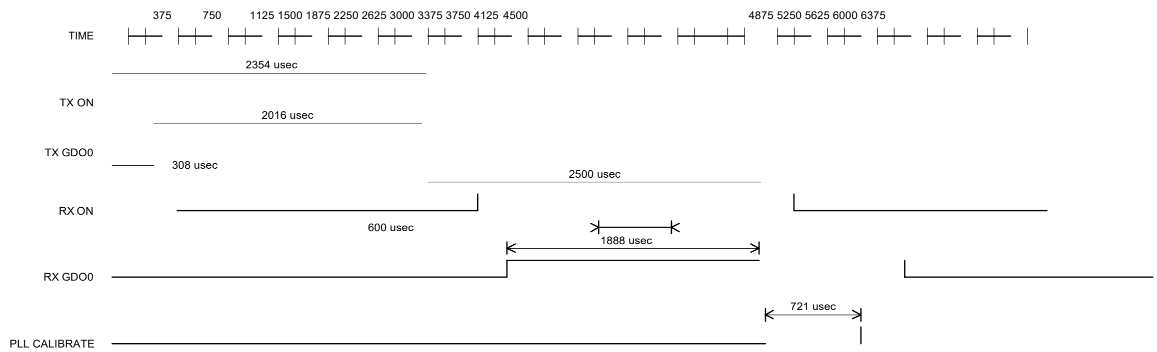
The ADCs conversion time and, by implication, maximum sample rate is a function of the selected resolution.

ADC resolution (bits)	Conversion Time (usec)	Maximum Sample Rate (kHz)
7	18.4	54.3
9	33.12	30.2
10	62.56	16.0
12	121.44	8.2

Note: The listed ADC resolutions apply to differential input applications, where the ADC value represents the voltage difference between the input signal and Vref. For example, with a Vref of 1.25 volts, ADC values can range from -1.25 to +1.25 volts. For single ended applications, the ADC value represents the voltage difference between ground and Vref. One bit of resolution is lost in single ended applications.

The ADC data must be sent in ‘packets’. Every packet contains some fixed overhead, including a preamble field (4 to 8 bytes, usually 4 bytes as in this application), sync field (2 to 4 bytes, usually 4 bytes as in this application), CRC field (2 bytes), a packet length specifier (1 byte), and a MAC (Machine) Address (1 byte). In addition, the CC2510 requires 88.4 usec to transition from the Idle state to either the Receiver On (Rx On) on Transmitter On (Tx On) state, and 721 usec to calibrate the PLL.

Packet overhead can be minimized by maximizing packet length. However, ‘long’ packets are more likely to be corrupted during transmission than ‘short’ packets. Additionally, audio latency and the length of the ‘blank out’ period (if a packet is lost) increase with increasing packet length. A timing diagram for the Master card is shown below.



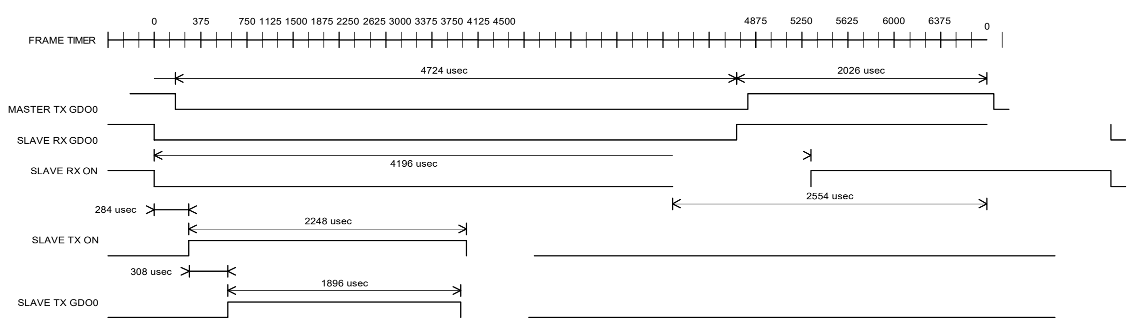
Master Timing

Where

Audio samples per packet: 54
Audio Sampling Rate: 8 kHz
Audio data rate: 64 kbps in each direction (voice quality)
Radio data rate: 250 kbps
Packet length (excluding preamble, sync, and CRC fields): 61 bytes
Total packet length (including preamble, sync, and CRC fields): 71 bytes
Idle to TX ON and Idle to RX ON: 89 usec

In the diagram, "TX ON" and "RX ON" include the time required to change the radio from the IDLE state to the RX or TX ON state. "GDO0" refers to a signal available on pin 34 of the CC2510 (P1_5). It asserts when the sync word has been sent\received, and de-asserts at the end of the packet. 1605 usec after enabling the receiver, bit 3 ("SFD") of the PKTSTATUS register is checked to see if it is asserted. This bit is set when the SYNC word is found and reset after a packet is received. If the bit is not set, it is assumed that the expected packet has been lost and the receiver is shut off.

A timing diagram for the Slave card is shown below.



Slave Timing

Note that the Frame Timer is reset immediately after a packet is received from the Master, and that the receiver is turned on approximately 2570 usec before the timer 'times out'.

Additional Specifications:

Resolution: 9 bits, compressed to 8 bits using ulaw algorithm
Spread spectrum technique: Frequency hopping (4 channels)
Hop rate: 148 hops/sec

Modulation: MSK
RF Output Power: 0 dBm (1 mW)

Features:

- Frequency Hopping (4 channels)
- Automatic Gain Control (Microphone): A software controlled AGC is implemented. The microphone preamplifier has four discrete gain settings of approximately 10, 20, 40, and 80 (20, 26, 32, and 38 dB), selectable via two control lines. The VGA gain is increased by 6 dB (2 times) whenever the peak ADC voltage is less than approximately 618 millivolts. The VGA gain is decreased by 6 dB (.5 times) whenever the CC2510s peak ADC voltage is more than approximately 1440 millivolts.
- Manual Volume Control (Speaker): The card uses a Texas Instruments TPA122 Mono power amplifier. Volume is controlled via a one-turn potentiometer.

Card Layout

In a ‘two way’, full duplex design, the placement of components on the card is critical. The RF signal from the antenna during transmission can be rectified by the microphone preamplifier and filter components, producing a noticeable ‘buzz’. To minimize this effect, the preamplifier, filter, and headphone driver components have been placed as far away as possible from the antenna and (with the exception of the TPA122 headphone driver and associated components) on the back side of the card. The CC2510 and its associated components have been placed on the front side of the card.

Conclusion

This document describes a set of cards and software that demonstrate telephone quality audio over a 2400 MHz two way, full duplex link, using the CC2510 transceiver.

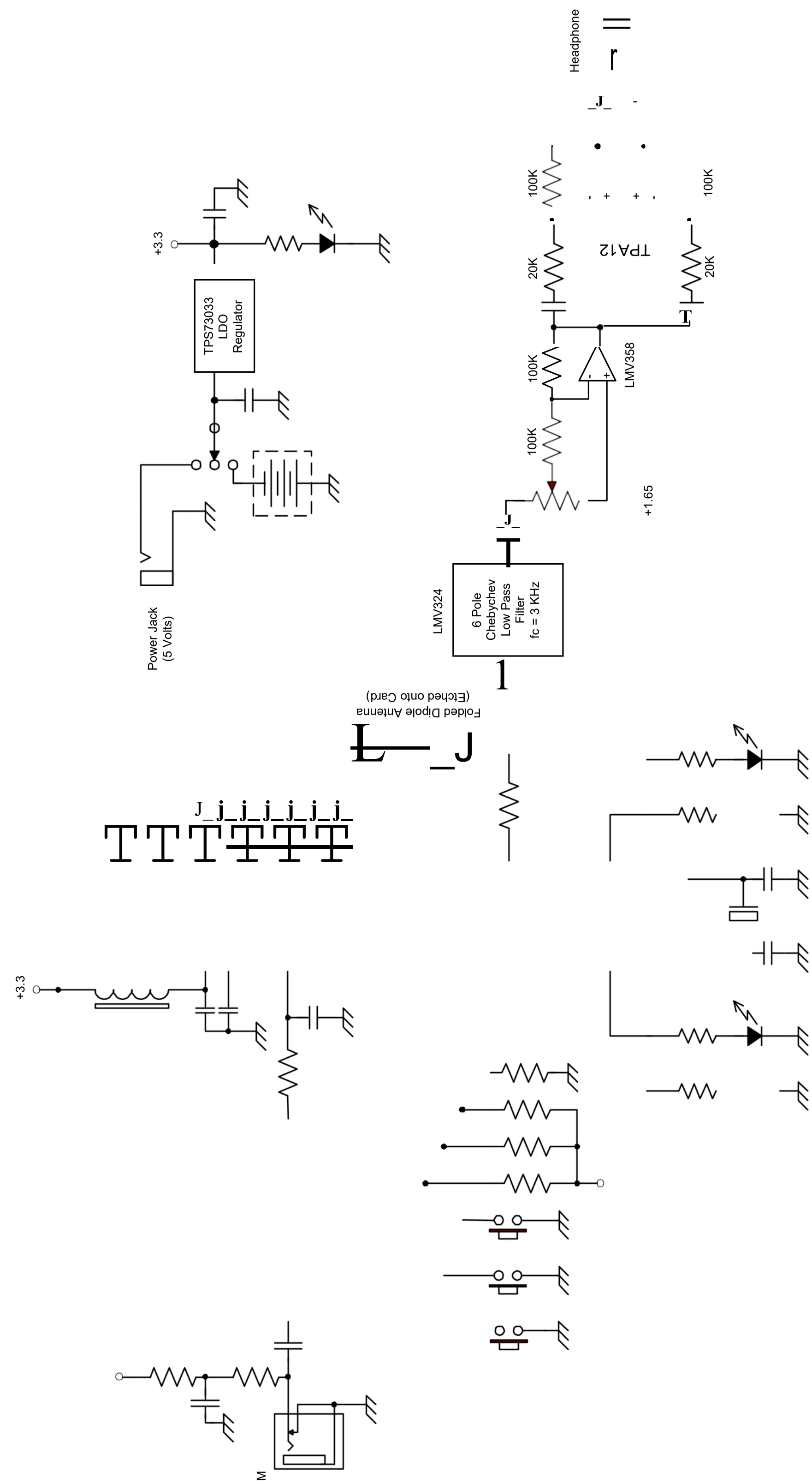


Figure 2 – Transceiver Block Diagram

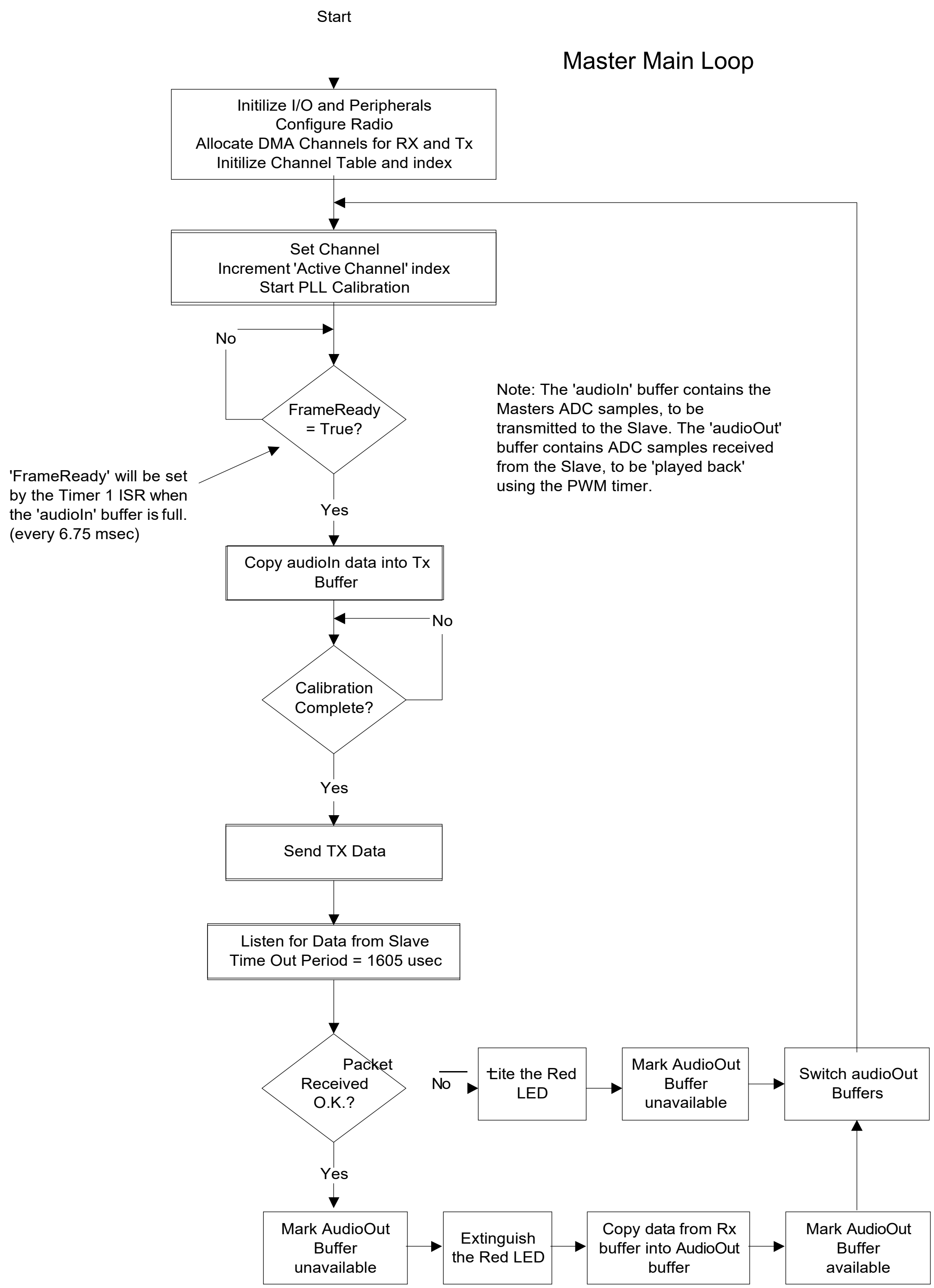


Figure 3 – Master Main Program Loop

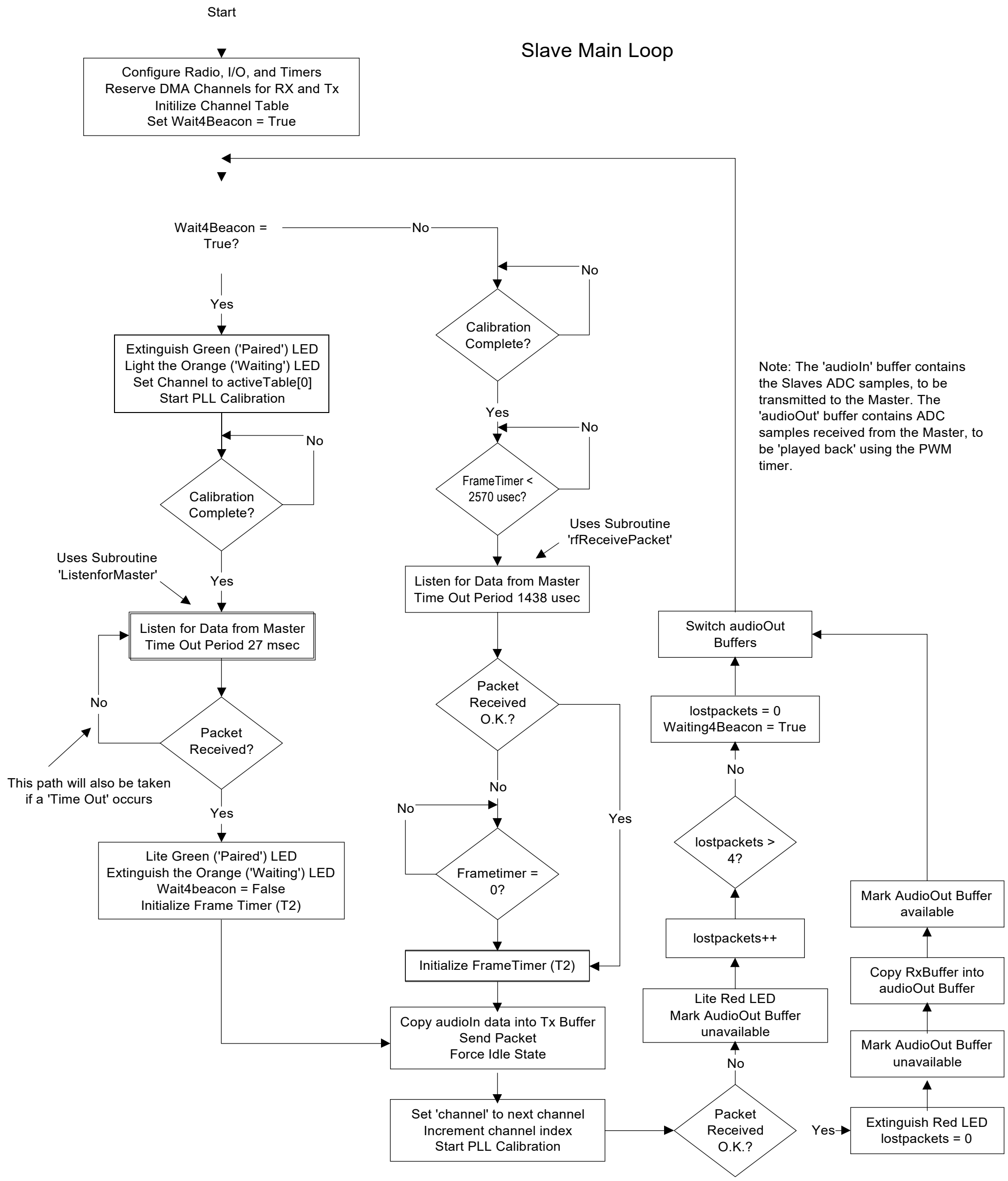


Figure 4 – Slave Main Program Loop

References

General references

[1] CC2510 Data Sheet (CC2510Fx)
[2] CC2510 Reference Design (http://www.ti.com/litv/zip/swrr035a)
[3] CC2510 Folded Dipole Antenna for CC25xx (swra118)
[4] Simple Audio Loopback Using CC251X (DN402)
[5] TPA122 Data Sheet (TPA122)
[6] TPS73033 Low Noise High PSR RF Low Drop Out Regulator (TPS73033)

Document History

Revision	Date	Description/Changes
	18 Sept 2008	Initial release.
A	14 April 2010	
B	30 June 2010	Added reference to optional μ-law compression

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