Wireless hands-free using nRF24E1

INTRODUCTION

This document presents a wireless hands-free concept based on Nordic VLSI device nRF24E1, 2.4 GHz transceiver with embedded 8051 u-controller and A/D converter. Due to the unique features of the nRF24E1 this application can be realized with a minimum of external components.

Key features:
1. From single chip RF transceiver solution to 'Single chip wireless hands-free solution'
2. Voice quality audio (as defined by land line phone systems)
3. No need for codecs to compress audio information.
4. Secure communication, on chip-protocol generation and unique addressing of each system.
5. Emulated full duplex RF link
6. 2.4 GHz, world wide license free RF band, operation
7. Multiple channels available (up to 125 depending on local frequency regulations)
8. Frequency jumping schemes can be implemented.
9. Battery monitoring with reporting to phone.
10. Extremely low current consumption, only 8.9 mA* average during operation.
11. Long battery lifetime, up to 13 hours talking and 1200 hours (50 days) standby on a 120 mAh cell
12. Very small physical outline
13. 7 free GPIO line enables embedding of additional features such as remote pick-up, volume control, LED indicators and other.

- Current consumption in pre and post filtering / gain stages are not included.

BASICS

Hands-free systems are based on two way analog audio so a wireless system must hence emulate this with adequate quality.
Voice audio have a frequency band between 300 – 3400 kHz, similar to landline telephone. When digitizing phone quality audio, one need a minimum 8 bit resolution with a sampling frequency of 6.8 kHz. The sampling frequency is generally set to 8 kHz requiring payload data rate of 64 kbit/s each way on the link.
To improve audio quality and/or reduce the demands to the pre filtering, higher sampling rates can be used, requiring higher data rates or compression accordingly.

On the air a robust RF protocol, dividing the audio information into packets with address and check sum to ensure data integrity, is utilized to give a secure link both in terms of stability and protection against eavesdropping.

The simplest way to re-create the audio signal from the digital information is to utilize pulse width modulation (PWM) with post filtering. To re-create an audio signal with 8 bit resolution a 8 bit PWM modulator is also needed. The carrier frequency of the modulator should be as high as possible to reduce the requirements to the post filtering.

The I/O format on both sides of the wireless hands-free is audio. Two identical modules are hence required. One module must however be given the role of master in the system, controlling the communication timing.

nRF24E1

The nRF24E1 has all the key features needed for a wireless hands-free system:

1. Embedded (up to) 12 bit A/D converter for audio sampling and battery monitoring.
2. 2.4 GHz RF transceiver, featuring the unique ShockBurst™ communication mode.
3. 8 bit PWM output for D/A conversion.

Pre and post filtering and amplification of the audio must be done off chip.
This concept presents a basic system utilizing 8bit resolution at 8 kHz sampling rate resulting in two way 64 kbit/s. The block schematic can be seen in Figure 1.

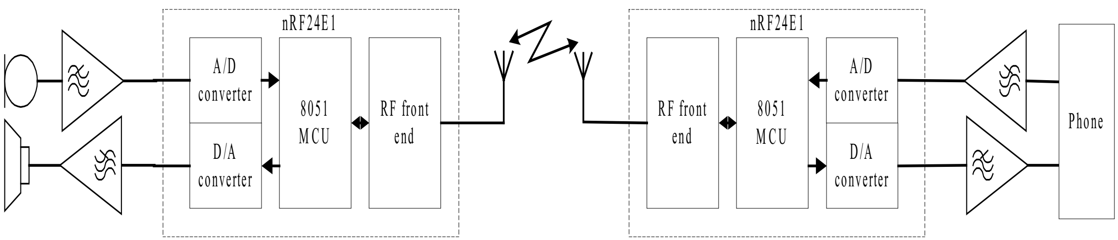

Figure 1: Wireless hands free block schematic

A/D conversion

The 8051 store samples from the nRF24E1 on-chip ADC until sufficient data for a RF packet is collected. The data, along with the address of the receiver, is then transferred to the RF front end, while the 8051 continue storing data for a new package. By utilizing the full length of the ShockBurst™ RF packages, 24 bytes or 3.0 ms of audio samples can be transferred pr. package.

By muting the output and only send short status messages on the air when no audio input is present one can also reduce the data transfer on air beyond this and hence reduce system current consumption. To achieve this, the muting function must be implemented on the TX side of the link.

RF communication

The link must emulate a full duplex system giving 64 kbit/s true data rate (i.e. 64 kbit/s payload) in each direction.

Due to the unique ShockBurst™ feature of the nRF24xx devices where all protocol related issues are handled by a hardware engine, a cheap and relatively slow u-controller like the 8051 can handle and take advantage of the 1 Mbit/s on the air data rate offered. With this feature the 64 kbit/s true data rate is achieved, even with the overhead needed for the RF protocol.

After configuration, the RF front is seen by the u-controller like any other peripheral unit with its I/O registers and interrupt-features.

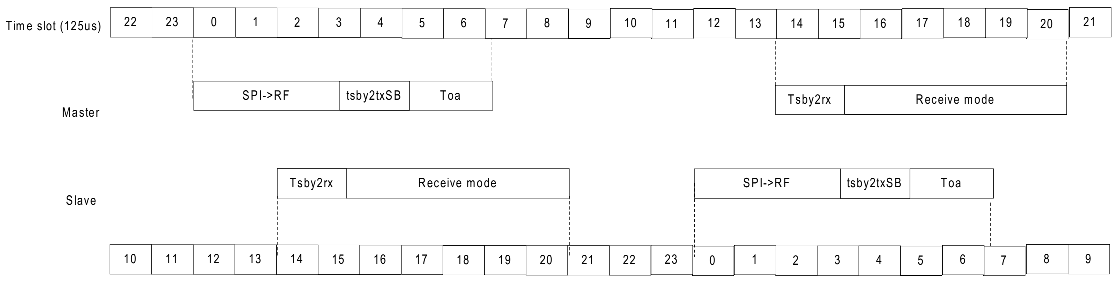

Figure 2 Full duplex handshaking concept

The fundamental timing is based on standard length time slots dictated by the sampling frequency (8 kHz = 125 us).

In each timeslot the A/D must be read and the PWM updated. This activity takes ~15 % of each time slot leaving the rest to handle the RF communication. Synchronization of the master and slave is achieved by coupling the activities on air to a counter. As can be seen in Figure 2, the master always transmits in time slot 5 – 7, and once the slave has received a valid package, the RF front end is turned off and the slave counter is reset to 19. The time spent in receive mode in the figure, is hence a worst case scenario, when in lock the modules only keep the RF front end active until a valid package has arrived.

The RF packages utilized can be 248 bits long (8-bit preamble+16-bit CRC+32-bit address+24 byte payload). Each package hence contains 24 samples, which results in a transmission every 3 ms.

Thanks to the 1 Mbit/s on the air data rate the system can duty cycle the RF front end in normal operation, hence reducing current consumption while emulating the full duplex link.

D/A conversion

Upon receiving a valid packet on the RF front end, the u-controller is interrupted and the payload section of the incoming package can be downloaded from the RF front-end I/O register. This data is then stored for transfer to the PWM output.

The PWM output is driven by an 8 bit hardware PWM engine and hence doesn’t load the u-controller core with its operation. The maximum carrier frequency of the nRF24E1 PWM modulator with 8 bit resolution is 64 kHz and this is utilized to ease the post filtering.

PRE AND POST FILTERING

Both on the input from the microphone and on the output to the loud speaker gain and filtering is needed.

Pre filtering

A/D theory states that an A/D converter sampling a spectrum 3.4 kHz wide at a rate of 8 kHz and 8 bit resolution needs 50 dB damping of any noise at the repeated spectrum (which starts at 8 kHz – 3.4 kHz = 4.6 kHz). In speech processing it is however usual to take advantage of the fact that the energy is mainly situated below 3 kHz and hence the lack of energy above 4 kHz realizes the main part of the filtering. In addition to this, one of course also gains from any bandwidth limitations in the microphone itself. Due to these facts it is common to implement only a 1. or 2. order filter on an ADC input handling voice quality audio.

As the output signal from the microphone must be amplified (improved SNR) to fit the dynamic range of the A/D converter, an external amplifier is needed.

Post filtering

The PWM output has a large frequency component on its carrier frequency (in this case 64 kHz). It also has a repeated spectrums around 8 kHz due to the update rate of the audio signal. To re-generate the analog audio signal both these noise signals must be filtered out adequately.
The bandwidth of the selected loudspeaker itself, typically 3.5 kHz for applications like this, contributes to the filtering of the output signal. The stop band characteristic of the loud speaker then dictates the complexity needed in the post filter.

CURRENT CONSUMPTION

The total current consumption of the application is of course depending on the pre and post stages. Depending on the complexity one want in the design these stages can be turned of by use of GPIO ports on the nRF24E1, but case using a low end approach with only op-amps the current consumption should be in the uA range.

As this current consumption is entirely depending on the key design features of the end application the current consumption of the nRF24E1 alone is treated below. The same formulas can be utilized on the pre and post stages if they are not continuously on.

Operation

The nRF24E1 current consumption when active is given by the following (please refer to Figure 2):

8051, A/D and D/A has a continuos current consumption of 4 mA.
RF front-end current consumption in the different time intervals:

$T_{SPI\ ->RF}$:	438us @ 0.5 mA
T_a =Tsby2txSB + T_{oa}:	444 us @ 8 mA (-20 dBm RF output power)*
T_b = $T_{sby2rxSB}$ + $T_{receive\ mode}$:	660 us @ 19 mA

*-20 dBm output power is utilized because only short range is needed in the system (<5 m) This figure also depends heavily on preferred antenna solution.

One can now find the nRF24E1 peak and average current consumption.

Peak:

4 mA + 19 mA = 23 mA

Average:

$$I_{av} = \frac{T_{SPI->RF} \cdot 4.5mA + T_a * 12mA + T_b \cdot 23mA + \left(3ms - T_{SPI->RF} - T_a - T_b\right) \cdot 4ma}{3ms}$$

Which gives I_{av} = 8.9 mA + the current drawn in pre and post stages.

The average current consumption will be the same for the master and slave unit.

Stand-by

The current consumption in stand-by is depending on how fast one want/need the hands-free to respond when a call is initiated.

To get a seemingly transparent wireless hands-free, the following concept can be used. The concept depends on the phone unit being activated when the phone itself is activated i.e. before the actual audio is to be transferred.

The phone module is designated master in the system and sleeps until an incoming call is registered or a number is being punched to initiate a call on the phone. The master then goes active, sending packages every 3 ms as described in Figure 2.

The slave (head-set) is also in power down (~2uA current consumption) and must hence wake up and listen on the air to detect any packets sent from the master. Let's look at the different scenarios to establish a reasonable slave wake-up time.

Incoming calls:
1. If the phone is used for pick-up, the delay from the signal is heard by the user until the pick-up button is pressed is generally several seconds.
2. In the case of auto pick up, the delay is in the seconds range as well (pick-up after five signals for instance).
3. A pick-up switch on the ear piece (slave) can wake it up instantaneously establishing a link with the master.

Outgoing calls:
When a call out is to take place, the delay from a number is punched or selected from the phone book until the phone rings on the other end is also in the range of several seconds.

So one may assume that the wireless hands-free system has in the range of seconds to establish a link between the master and slave unit.

Following is an example where the slave is listening for incoming packages once every second.

If a link needs to be established, the master is already transmitting packages every 3 ms so the minimum on the air listening time needed is 3ms + 1 package length + margins depending on TX crystal accuracy.

The nRF24E1 needs 1 ms to start from power down upon RTC wake-up and in this interval the current consumption is 2 mA.
An on-board RC oscillator drives the timer used for this 1 second cycle, and the clock accuracy is only +/-20 %. The maximum stand-by current is drawn when the interval is 1 sec – 20% or 800 ms.

The stand-by current consumption in the slave will be:

$$I_{av} = \frac{T_{startup} \cdot 2mA + T_R * 22mA + \left(800ms - T_{startup} - T_R\right) \cdot 2uA}{800ms}$$

Where:

$T_{startup}$: 1ms

$T_R = T_{sby2rxSB} + T_{receive}$ = 3.45ms (no crystal margins)

Average stand-by current consumption: <u>99.3 uA</u>

The master will draw only 2 uA in stand-by since it will go directly to active mode upon input from the phone.

This approach is as mentioned depending on the hands-free master being activated at the same time as the phone itself.

Battery

The battery for the application must be able to handle the short intervals of peak current when the RF front end is active. Batteries with lower peak current capability can be utilized if the battery is aided by reservoir capacitors.

Battery life time

If a 120 mAh battery is used the expected 'active battery lifetime' (or talk time) is:

$$T_{act} = \frac{Battery\ capacity}{Average\ current} = \frac{120mAh}{8.9mA} = 13\ hours$$

NOTE: current consumption in pre and post stages will increase average current consumption and bring T_{act} down.

The expected stand-by time will be:

$$T_{st-by} = \frac{120mAh}{99{,}3uA} = 1208\ hours = 50\ days$$

CONCLUSION

The nRF24E1 is suitable for simple wireless hands-free applications. The device offers a low-cost and low power solution.

An 8bit x 8 kHz solution is fairly straight forward both in terms of hardware and software. The audio quality in an 8 bit 8 kHz solution will facilitate phone quality, but both bandwidth and dynamic range is limited.

Several issues can be pursued to improve audio quality and reduce current consumption. Below is short overview over both the feasible and not feasible options.

Main factors on audio quality:

Pre and post stages: Here one of course has a lot of options depending on the cost and physical space available. From simple RC as the cheapest to 5-6 order active filters, off-the shelf switch-cap and dedicated phone line filters can be used.

Microphone and loudspeaker: These are of course vital in terms of audio quality in themselves. They will also reduce requirements to the pre and post stages if they are chosen to fit the audio bandwidth needed aiding in the filtering.

Increased sampling rate: Improving audio quality through higher sampling rates is not an option utilizing nRF24E1 and internal ADC/DAC. Due to the processing capacity 8bit x 8kHz uncompressed is close to what the nRF24E1 can handle. 8 bit x 10 kHz should be possible but nothing far beyond that.

Higher dynamic range: This is limited in the nRF24E1 both by PWM resolution and transfer/processing capacity in the 8051 processor. It is hence not a way to go either. Combining higher sampling rates with simple digital signal processing (decimation filter) to maintain the 8 kHz transfer rate on the air is however an alternative.

Main factors reducing current consumption:

Transfer rate: Keeping the volume of data to be transferred down is the main current saving technique when the system is active. Implementing a mute region enabling you to cut the communication rate and/or packet length when no input from the microphone is present can be a powerful current saver. The muting in it self is fairly simple to make, how the user will perceive it is another thing. A raw mute of the output will leave a complete silence in the loudspeaker this might not 'sound naturally' to the user.

Response time: Letting the devices sleep as much as possible is saving current in stand-by. A wake-up routine with a response time of ~1 sec. is presented in this white paper, if a longer response time is acceptable lower stand-by current follows.