AN2394
Application note

STR7/STR9 audio generation with PWM

Introduction

The purpose of this application note is to provide a hardware and firmware solution to STR7 and STR9 microcontroller users for audio playback of a *.WAV* file.

The approach is optimized in that it uses a minimal number of components external to the microcontroller, and offers a high degree of flexibility to the end-user for use with their own *.WAV* files. There are two *.WAV* file parameters that can be controlled by the user; the sample rate and the file size which depends on the application requirements. The actual content of the *.WAV* file is irrelevant and may consist of speech, music, etc., and the only limitation is the audio format. In fact, this application assumes that the *.WAV* file format must be: PCM (no compression), 8000/11025/22050/44100 Hz sample rate, 8-bit and mono.

This document is structured as follows: a brief description of the *.WAV* file format in *Section 1*. *Section 2* provides a detailed description of the basics of audio playback. Finally, *Section 3* presents in detail an example of an application built around an STR711F microcontroller and that can be easily tailored to any other STR7/STR9 microcontroller.

Contents

1 .*WAV* file format

The .*WAV* file format is a subset of the Resource Interchange File Format (RIFF) specification used for the storage of multimedia files. A RIFF file starts with a file header followed by a sequence of data chunks. A .*WAV* file is often just a RIFF file with a single "WAVE" chunk consisting of two sub-chunks:

1. a **fmt** chunk specifying the data format
2. a **data** chunk containing the actual sample data.

The numerical format used is Little-Endian (least significant byte first) and Big-Endian (most significant byte first) order.

The WAVE file format starts with the RIFF header:

```
Endian    Offset    Length     Contents
 big       0         4 bytes    'RIFF'              // 0x52494646
 little    4         4 bytes    <file length - 8>
 big       8         4 bytes    'WAVE'              // 0x57415645
```

Next, the fmt chunk describes the sample format:

```
 big       12        4 bytes    'fmt '          // 0x666D7420
 little    16        4 bytes    0x00000010      // Length of the fmt data (16 bytes)
 little    20        2 bytes    0x0001          // Format tag: 1 = PCM
 little    22        2 bytes    <channels>      // Channels: 1 = mono, 2 = stereo
 little    24        4 bytes    <sample rate>   // Samples per second: e.g., 22050
 little    28        4 bytes    <bytes/second>  // sample rate * block align
 little    32        2 bytes    <block align>   // channels * bits/sample / 8
 little    34        2 bytes    <bits/sample>   // 8 or 16
```

Finally, the data chunk contains the sample data:

```
 big       36        4 bytes    'data'          // 0x64617461
 little    40        4 bytes    <length of the data block>
 little    44        *          <sample data>
```

This application assumes that the .*WAV* file to be played has the following format:

- Audio Format: PCM (an uncompressed wave data format in which each value represents the amplitude of the signal at the time of sampling.)
- Sample rate: may be 8000, 11025, 22050 or 44100 Hz.
- Bits Per Sample: 8-bit (Audio sample data values are in the range [0-255]).
- Number Of Channels: 1 (Mono)

2 Audio playback

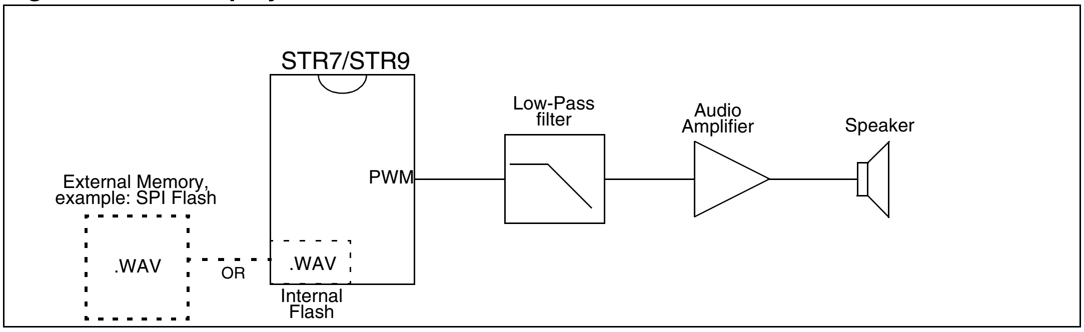
The audio playback requires two dedicated timers:

- Timer used as system timer, SysTimer: generates an interrupt at a programmable rate (fixed by the *.WAV* file sample rate value).
- Timer used in PWM mode. This timer should be able to produce high frequency PWM signal.

The *.WAV* file data can be stored in a compatible SPI-based Flash (requiring an SPI interface), in internal Flash memory (if the chosen STR7/SRT9 MCU has sufficient capacity) or on any other external memory type supported by the microcontroller .

The audio playback flow is presented in the following figure:

Figure 1. Audio playback flow



To playback the audio, the microcontroller reads each 8-bit sample and then outputs it to a digital-to-analog converter (DAC) at the desired sample rate. The analog wave produced by the DAC is amplified and fed to a speaker to produce the sound.

As the STR7/STR9 MCU doesn't have an on-chip DAC, an alternative method is used to implement 1 channel DAC. Such a method is the use of the built-in Pulse Width Modulation (PWM) module to generate a signal whose pulse width is proportional to the amplitude of the sample data. The PWM output signal is then integrated by a low-pass filter to remove high frequency components, leaving only the low-frequency content. The output of the low-pass filter provides a reasonable reproduction of the original analog signal.

2.1 STR7/STR9 PWM description

This section introduces the basic concepts of the PWM mode of the STR7/STR9 TIMER (TIM).

Pulse Width Modulation (PWM) consists of a signal with a fixed period whose duty cycle is variable. The duty cycle represent the ratio of ON period to the period of the signal.

For the TIM, the PWM period and duty cycle are determined by the value of the OCAR and OCBR registers.

The PWM output signal period is fixed by the OCBR which determines the maximum value that the counter can count before starting a new period:

$$T = OCBR * Prescaler * T_{Timer}$$

Where:

OCBR: Output Compare B register value

Prescaler: Timer clock prescaler

T_{Timer} : period of the Timer clock that drives the Prescaler.

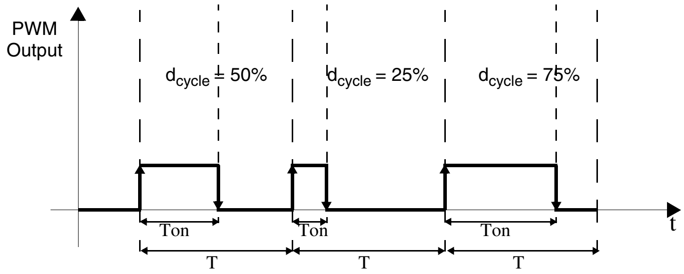
The value set in the OCAR determines the duty-cycle:

$$T_{on} = OCAR * Prescaler * T_{Timer}$$
$$d_{cycle} = (Ton/ T)*100 = (OCAR / OCBR) *100 \%$$

Where:

OCAR: Output Compare A register value

Figure 2. Typical PWM signal waveform



2.2 Audio playback

Before the audio data can be played, the header must be parsed so that the sampling rate of the data and its length can be determined.

The task of reproducing audio is achieved by using sampled data (data contained in the *.WAV* file) to vary the duty cycle of the PWM signal who's period is held at a constant value. The duty cycle represents the ratio of the pulse width to the pulse period, i.e. it represents the ratio of the OCAR value to the OCBR one.

As the duty cycle varies according to the values of the sampled data and this data is coded in 8 bits (with values from 0 to 255), the OCBR value must be set to 255 and the OCAR value fills with the sampled data values (the OCAR value varies from 0 to 255). In this manner the duty cycle varies from 0% to 100%.

The OCAR value is updated at intervals specified by the SysTimer period which is equal to the sample rate of the *.WAV* file.

The PWM output signal is then fed to a simple RC low-pass filter to generate an output voltage directly proportional to the average time spent in the HIGH state (i.e., 50% duty cycle is equal to 1.65V when VDD = 3.3V). When this is averaged over time, we get a reasonable reproduction of the original analog signal.

There are several design considerations involved in the selection of the cut-off frequency of the low-pass filter. Primarily, the filter cut-off frequency must be much lower than the PWM frequency to reduce the noise generated by PWM switching. A PWM frequency of 10 times the cut-off frequency is a generally sufficient. However, the cut-off frequency of the filter is critical to the bandwidth of the output audio.

In addition, it is recommended, on the basis of the Nyquist Criterion, that the cut-off frequency of any filter used be set at half the frequency of the sample playback rate. Since in this application different sample rates are supported (8000/11025/22050/44100 Hz), we must take into account the bandwidth limitation imposed by the lower sample rate.

From a hardware standpoint, a higher PWM frequency is easier to filter, but it corresponds to a smaller reload value and therefore results in lower resolution. By contrast, a smaller PWM frequency is difficult to filter; besides, the lower limit is dictated by the data input rate.

Incorporating this guideline, the filter cut-off frequency is set to 4 kHz. This ensures that almost the entire audio spectrum is passed through, but the PWM carrier is removed.

The values for R and C are computed with this following formula:

$$f = \frac{1}{2\pi RC}$$

where f is the cut-off frequency of the filter.

3 Implementation example with STR711F

This section describes a hardware and firmware solution of *.WAV* file playback implemented for the STR711F microcontroller and that can be easily tailored to any other STR7/STR9 microcontroller.

The following STR711F peripherals are used:

- TIM0 in Output Compare timing mode to act as SysTimer. The TIM0 global interrupt can be linked to IRQ (Interrupt Request) channel0 or to FIQ (Fast Interrupt Request) channel0. For fast, low latency interrupt handling, FIQ channel0 is used to handle TIM0 interrupts.
- TIM3 in PWM mode.
- BSPI0 to communicate with the SPI Flash.

3.1 Hardware description

The electrical schematic of the circuit, which the proposed application refers to is shown in *Figure 3*.

An M25P64 SPI-based Flash memory is used to store and send back the *.WAV* file data to the ICU. The M25P64 is a 64-Mbit (8M x 8) Serial Flash memory from STMicroelectronics, accessed by a high speed SPI-compatible bus.

The connection scheme between the microcontroller and the SPI Flash is as the following: pin S0.SCLK(P0.2) is connected to the Serial Clock pin (C) for data synchronization, pin S0.MOSI(P0.1) and pin S0.MISO(P0.0) are connected, respectively, to Serial Data Input pin (D) and Serial Data Output pin (Q) for data transmission. To select and deselect the Flash, its Chip Select pin ($\overline{S}$) is dynamically driven by P1.1 pin.

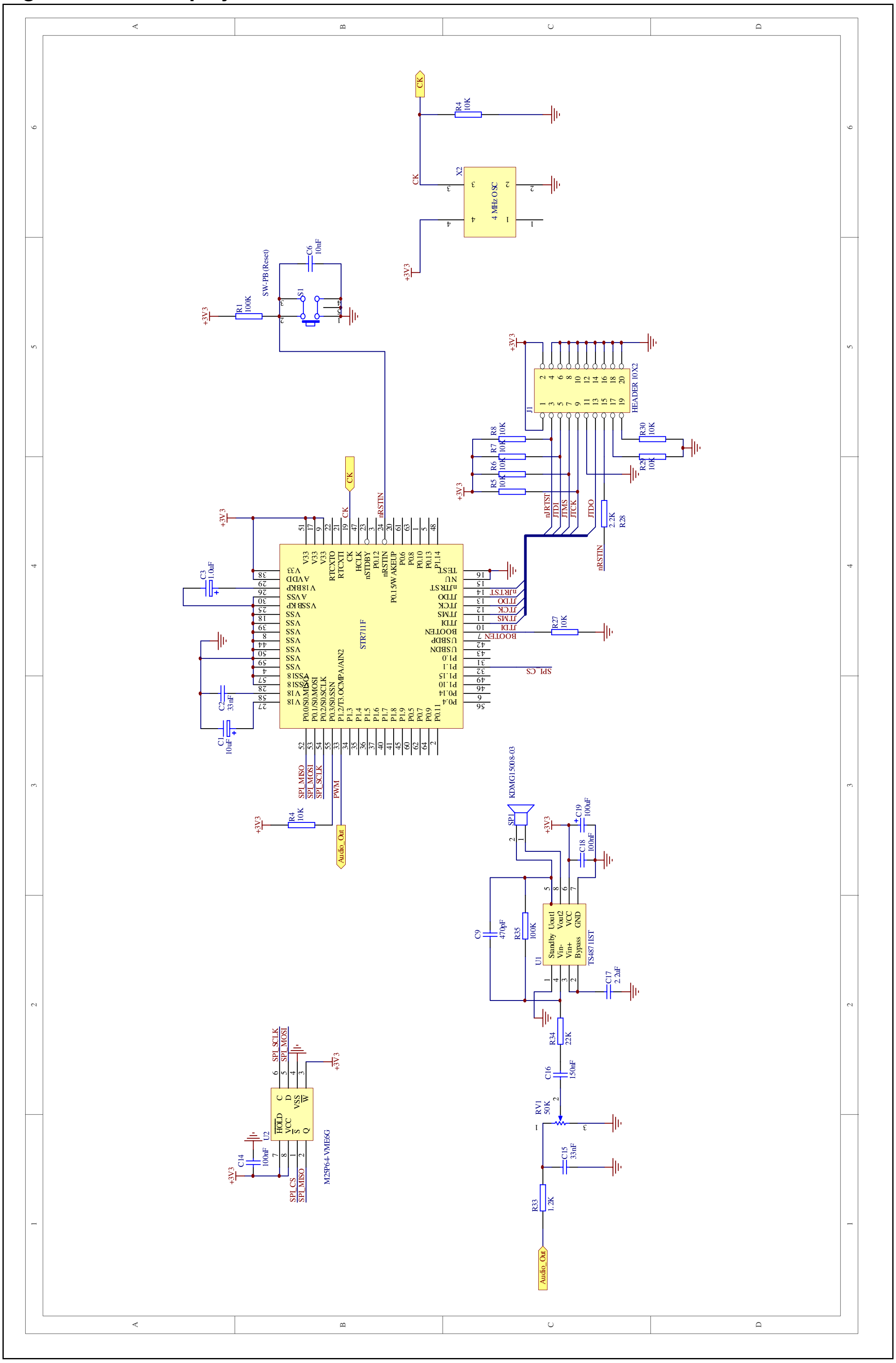
The T3.OCMPA pin (P1.2) is used to output the PWM signal. This signal is passed through a simple low-pass RC (R33 and C15) filter. The cut-off frequency of this filter is set at 4 kHz. This ensures that almost the entire audio spectrum is passed through, but the PWM carrier is cut off.

Since the cut-off frequency of the low-pass filter is set at 4kHz, the R and C component values are calculated as: R = 1.2 kΩ, C = 33 nF.

The filter output is fed to the input of a common TS4871 audio amplifier through a potentiometer to allow continuous control of the audio volume.

The TS4871 is a monolithic integrated audio power amplifier chip from STMicroelectronics capable of delivering 0.5W of output power into an 8ohm load at 3V3. The amplifier output is used to drive an 8ohm speaker.

Figure 3. Audio playback with STR711F



3.2 Firmware description

The software associated with this application is developed in C using the STR71x Firmware Library from STMicroelectronics. The STR71x Firmware Library provides a set of functions covering the functionality of the STR71x peripheral. As a result, users can develop applications for any STR71x device without an in-depth study of each peripheral specification.

Note: The firmware assumes that the .WAV file is stored at address 0x00 of the Flash.

The first part of the software performs the configuration of the different system clocks. The main clock (MCLK), PCLK1 (including serial communication peripherals, such as BSPI0) and PCLK2 (including system peripherals, such as TIM0 and TIM3) are set to 32MHz using an external 4MHz oscillator.

Once the BSPI0 peripheral and its associated pin are configured, the baudrate is set to 5.3Mb/s.

The task of audio playback is achieved by calling the *WAVE_PlayBack* function. This function starts by a call to the *WAVE_Parsing* function. This checks the format of the *.WAV* file and gets information about the audio format. This is done by reading the value of a number of parameters stored in the file header and comparing these to the values expected for the format of a standard *.WAV* file. If a valid *.WAV* file format, it continues reading the header to determine the audio format in terms of the sample rate and the sampled data size. If the audio format is supported by this application, it retrieves the audio format in the WAVE_FormatTypeDef structure and returns a zero value. Otherwise the function fails and the return value is nonzero. In this case, the return value specifies the cause of the function fails.

The *WAVE_PlayBack* function tests the value returned by *WAVE_Parsing* function, if it is a nonzero value the audio playback fails. Otherwise, it configures the TIM0 and TIM3 timers. TIM3 is programmed to generate 125.5kHz PWM signal and TIM0 is programmed to generate an interrupt at frequency equal to the *.WAV* file sample rate value.

Note: As TIM3 OCBR value is fixed to 255, the maximum PWM frequency that can be reached is 125.5kHz.

On each TIM0 interrupt, a byte is read from the SPI Flash to update the TIM3 Output Compare A register (OCAR) value. A counter loop operates by counting down from the sampled data size value and when zero is reached the playback is stopped.

Usually, to read a byte from the SPI Flash several steps are performed:

1. Drive the $\overline{S}$ line low to select the device
2. Send read data instruction
3. Send a 3 byte address
4. Send a dummy byte. Then the memory content, at that address, is shifted out on Serial Data Output (Q)
5. Drive the $\overline{S}$ line high to deselect the device

Repeating this procedure for each byte read from the SPI Flash increases the processing time of the TIM0 interrupt service routine. This can affect the CPU load and limits other tasks that the application may demand.

To avoid this and decrease the CPU load, a solution based on the capability to read the whole SPI Flash memory with a single read data instruction is implemented:

– the first time a data byte is needed, a call to *Start_Read_Sequence* function is made. This initiates a read data byte sequence from the SPI Flash. This is done by driving the $\overline{S}$ line low to select the device, then the read data instruction is transmitted followed by 24-bit address. This function exits and keeps the $\overline{S}$ line low, with the SPI Flash still being selected.

The next routine which needs to read a byte from the SPI Flash has only to generate a dummy byte on the Serial Data Input line of the SPI Flash and get the data on the Serial Data Output line. So only step 4 is performed each time a data is read from the SPI Flash which significantly reduces the CPU overhead.

When the playback is finished, the read data byte sequence is terminated by driving Chip Select high.

4 Conclusion

This document proposes a straightforward and flexible solution to use a member of the STR family, STR711F, for playback of a *.WAV* file stored in an SPI Flash memory. This can be used in many applications where high-performance audio playback is not required such as: talk back units, man-machine interfacing and low-cost music players.

5 Revision history

Table 1. Document revision history

Date	Revision	Changes
06-Dec-2007	1	Initial release.

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