

www.robotshop.com Adafruit Wave Shield User Guide RB-Ada-04

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Introduction

The shield comes with an Arduino library for easy use; simply drag uncompressed wave files onto the SD card and plug it in. Then use the library to play audio when buttons are pressed, or when a sensor goes off, or when serial data is received, etc. Audio is played asynchronously as an interrupt, so the Arduino can perform tasks while the audio is playing.

- Can play any uncompressed 22KHz, 16bit, mono Wave (.wav) files of any size. While it isnt CD quality, it is certainly good enough to play music, have spoken word, or audio effects
- Output is mono, into L and R channels, standard 3.5mm headphone jack and a connection for a speaker that is switched on when the headphones are unplugged
- Files are read off of FAT16 formatted SD/MMC card
- Included library makes playing audio easy

While the shield has been tested and works well, here are some points to keep in mind

- The audio playback library uses 10K of flash so if you want to use an NG arduino, you'll need to upgrade to an Atmega168 chip
- About 600 bytes of SRAM are used to buffer the audio and keep track of file data, so RAM-heavy projects may not work well
- The shield can't play MP3, WMA, Ogg or other compressed audio files. It can only play uncompressed PCM/WAV files. <u>Converting audio to WAV format</u> is very easy, and is often the default format for many audio programs.
- Files are stored as 8.3 name format, and can only be placed in the root directory. That means you can only have ~512 files (but they can be any size).

Ideas for what you can use it for.

- Make a portable audio player
- Use the AT&T text-to-speech site to make snippets of speech that you string together for a talking project, like..
- Talking temperature sensor
- Talking clock
- Interfaces for sight-impared people
- Doorbell that plays a cool tune
- Jukebox/music-box that plays a song when its opened, or a coin is inserted
- Security system that warns the intruder
- Audio looper for musical effects and performances
- Synthesizer with different sounds
- Really freaky halloween props that scream
- Display (like a point-of-sale box) that you can plug into to hear the message

FAQs

Can this shield play MP3 files? What about WMA, Ogg, AAC...?

No, compressed audio requires either a specialized chip (which is expensive) or a very powerful chip. The Arduino microcontroller can't uncompress MP3 on the fly and to keep the shield inexpensive, no mp3 decoder chip is included.

What sort of audio can it play?

It can play uncompressed Wave files (.wav format). This is a standard format and pretty much every audio program can convert your music or audio into wave format. Make sure the sample rate is mono, 22KHz (or less) and 16-bit (or less).

What does it sound like?

The best way to determine if the quality is good enough for your project is use Audacity and go thru the steps in the User Manual for converting MP3s (and other files) to 22KHz/16-bit format.

Can this shield record audio?

There is no hook-up for a microphone, so there is no easy way to record audio. There is also not enough program space on the current Arduino chips (atmega168) to support recording audio and saving it to the SD card as well as playback.

What pins are used by the shield?

Pins 13, 12, 11 are always used by the SD card (they are the only pins that have a high speed SPI interface). Then there are 5 other pins used to talk to the DAC and SD card, but they can be set to connect to any arduino pin. However, by default, the library is configured to use pins 10 (for SD card) and pins 2, 3, 4 and 5 for the DAC. To chanage these pins requires modifying the library - the pins are referenced by their 'hardware' pin names (ie PORTD, etc) not by arduino pins. That means pins 6, 7, 8, 9 and the 6 analog in pins (also known as digital i/o pins 14-20) are available

Overview

Here is an explanation of how the wave shield works. We'll go section by section. You'll want to refer to the schematic

Voltage regulator

The easiest thing to understand is the 3.3V voltage regulator. This takes the 5V supply from the Arduino and converts it to a nice 3.3V supply. This is necessary because SD/MMC cards only work on 3.3V. If you give them 5V they'll burn out & die!

The voltage regulator used is the MCP1700-330, which can provide up to 250 mA of current. There are 4 capacitors associated with the regulator. **C1** and **C2** are the input capacitors; they stabilize the 5V input. **C3** and **C4** are the output capacitors, they stabilize the 3.3V output

There is a jumper that allows you to skip the regulator and use the 'built in' 3.3V supply from the Arduino. However, it is not suggested as that supply is not guaranteed to provide the current necessary.

SD/MMC card holder



SD/MMC cards are very popular, small, and inexpensive. The card holder is what allows you to remove and replace the card easily. They can be removed/replaced thousands of times. The top three 'pins' are **CD**, **WP** and **COMMON_SW**. CD stands for "card detect" this is a mechanical switch that closes when the card is inserted. WP stands for "write protect", this is a mechanical switch that closes when the card has the little side tab slid down to 'lock'. COMMON_SW is the common connection for the two switches. We simply connect this to ground. Thus CD and WP will be grounded when active

At the bottom are the power supplies. There are 2 mechanical ground connections and a logic ground. There is also the logic power connection, connected to the 3.3v regulator



In the middle are the data connections. **DAT1** and **DAT2** are for advanced/high-speed SD card interfacing. We don't do this so they are left disconnected. **DATA_OUT** is the serial data out from the card, which is connected to the SPI port of the Arduino. **DATA_IN** is the input and **SCLK** is the clock input. Since they must be 3.3V and the Arduino usually sends 5V data, we use voltage dividers (**R2, R3, R4** and **R5**) to reduce the inputs down.

CS is the select line, used to tell the MMC that we want to send it data. This line is pulled low (to ground) when we want to send data to the card. That means we need to make sure when we don't have anything connected, the pin is pulled high to ~ 3.3 V. We use **R6** as the pull-up and zener diode **D1** to keep the voltage at 3.3V. **R1** allows the diode to bias properly when the Arduino pulls the pin high.

The microcontroller/Arduino

The library contains a bunch of specialized code. The first part is a 'FAT16' library, this is a set of functions that allow the chip to read the SD card, locate files and read their contents. The method it does this by is particularly detailed and you can read the <u>SD/MMC</u> and <u>FAT16 manuals</u> if you're interested



Image stolen from Microsoft. Take that, Bill!

Once it opens a file and is ready to read it, it looks through the first section of the file. If it's a Wave file, there will be all sorts of information stored in this *header* that will indicate the channels (mono/stereo/etc), bits-per-sample (8 to 32), sample rate (ie 16KHz) etc. Basically, the firmware verifies that it is mono channel, 16 or less bits-per-sample and 22KHz or less sample rate. Then it sets up the audio interrupt that will go off sample-rate times a second. For example, if its a 22KHz audio sample, the interrupt will go off 22,000 times a second!



The audio is encoded in <u>PCM format</u>. This means "pulse Code Modulation". Lets say its a 16bit, 22khz wave. The audio waveform is sliced up 22,000 times a second and a corresponding value (up to 16 bits - from 0 to 65,635) is read from the waveform, then that value is stored in the file. Each sample is a unique value. The file is not compressed. This means the files are very large but the quality is very very good.

The SD card can provide 512 bytes at a time. This is buffered inside the Arduino's RAM so that we have smooth playback. (Techinally, its a double-buffer which means we read 256 bytes and play 256 bytes, then swap.) The audio interrupt picks one sample at a time and sends the data to the DAC (digital/analog converter)

DAC

The DAC is a very simple device. When you send it data it will convert that digital information *back* into an analog signal!

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11.1	÷.,	÷2.	÷.,	÷.	- i -	÷.,	÷.	÷.	÷.	÷.,	÷.,	i
1 B	11	÷21	11	12.1	12.1	Ξ.	1	12.1	12.1	12.1	11	
	÷.,		÷	÷.,		÷.,	-i-	÷.			÷.,	

You'll notice it actually doesn't get the orignal waveform perfect. The more bits of digital data, the higher quality of audio reproduction. CDs have 16-bits per sample. While it would have been nice to have a 16-bit DAC, the best option for this design was a 12-bit dac. (That's still quite good.)

The microcontroller/Arduino uses the DAC_CS (chip select), DAC_CLK (data clock), DAC_DI (data), and DAC_LATCH (convert the digital to analog) pins to send the sample data over. The DAC also has a Vref input, this is the reference voltage that it uses to define the maximum analog value it can generate. There is a very low low-pass filter connected to it (C6 and R8) so that any digital noise (there is -a lot-) will not make it into the audio signal.

There is another low-pass filter connected to the output of the DAC (**R7** and **C8**). This is for filtering out the 'square wave' component you see in the recreated-audio wave. Even though the noise is only 1/4096'ths of the signal (about 1.2mV) its still noise and these two components filter out anything above 11KHz. The reason the filter cut-off frequency is 11KHz and not 22KHz is that if you sample at

22KHz you will only be able to reproduce frequencies at half that rate, 11KHz. This is the <u>Nyquist</u> theory. It is sneaky but true. If you try to sample 16KHz waveform at 22KHz it will actually sound much -lower-, it will play at 6KHz (it is 'mirrored' around 11KHz)

Analog output

Finally there is the volume control and output stage. The potentiometer acts as a simple volume control. It simply divides down the analog signal from 5Vpp down to as low as 0Vpp. The pot is 'audio' type which means that the voltage changes logarithmically, which our ears interpret as linearly.

The analog signal then goes into a high-output, rail-to-rail opamp. This op-amp can provide up to 100mA per channel. The two channels are hooked up in parallel for up to 200mA output (at 5V). This means it can provide 1/8 W into an 80hm speaker (or 1/4 W into 40hm speaker). This isn't enough for a boom-box but its good for headphones and small speakers. The output is filtered through a bypass capacitor **C9** which will keep any DC voltage from going to the speaker, which could damage it.

The headphone jack is stereo, which both mono channels connected in parallel. This gives the most power output. There are internal switches in the jack so that when the headphones are removed, the audio flows to the 'speaker connection' next to the jack.

Parts list

Check to make sure your kit comes with the following parts. Sometimes we make mistakes so double check everything!

Image	Name	Description	Part # & Datasheet	Distributor	Qty
	IC1	3.3V linear voltage regulator, 250mA current	MCP1700- 3302E/TO	Digikey Mouser	1
HCP 4921 E/P (007 0 (0795 D (0795	IC2	12-bit DAC	MCP4921	Digikey Mouser	1
	IC3	High-current opamp	TS922IN or TS922AIN	Digikey Mouser	1
		SD/MMC card holder	Tyco 1734234-1	Digikey	1

TM1	10K Audio thumbwheel potentiometer. Includes pot, thumbwheel and tiny screw	311- 1204F- 10K	Mouser	1
X1	Stereo headphone jack with switches	STX- 3100-5N or STX- 3100-5NB	Mouser Digikey	1
R1	1/4W 5% 1.0K resistor Brown Black Red Gold		Digikey Mouser	1
R7	1/4W 5% 1.5K resistor Brown Green Red Gold		Digikey Mouser	1
R3, R5, R6	1/4W 5% 4.7K resistor yellow purple red		Digikey	3
R2, R4	1/4W 5% 10K resistor Brown, Black, Orange, Gold		Digikey Mouser	2
R8	1/4W 5% 100K resistor Brown, Black, Yellow, Gold		Digikey	1

	C8	0.01uF ceramic capacitor (103) May look deceptively like the 0.1uF ceramic capacitors! Lately has been shipped in an 'axial' (not 'radial' package. See instructions for details.			1
	C2, C3, C5, C6, C7	0.1uF ceramic capacitor (104) Looks deceptively like the 0.01uF ceramic capacitor!		Digikey Mouser	5
10v 100µF I3v 1	C1, C4, C9	100uF / 6V capacitor		Digikey Mouser	3
	D1	3.6V Zener diode	1N5227B	Digikey Mouser	1
	RESET	6mm tactile switch	B3F-1000	Digikey Mouser	1

	ICSP	6-pin ICSP header	Digikey Mouser	1
+++++++++++++++++++++++++++++++++++++++		36 pin male header (1x36)	Digikey Mouser	1
	РСВ	Circuit board	Adafruit Industries RobotShop	1

Make it

Get ready by placing the PCB in a vise
We're going to the SD card first. While surface mount parts are a little tougher than thru-hole, this piece has pin spacing of 0.1" so it is quite easy. Doing it first also gives us lots of working room. The holder should 'snap' perfectly into place thanks to two bumps on the bottom.









Next is the 0.01uF ceramic capacitor C8. The tricky part here is that in older kits there are many 0.1uF ceramic capacitors in the kit that look identical to the 0.01uF! The way to tell the difference is look for the 103 printed on it. If it says 104 then it's a 0.1uF. Make sure it says 103! This capacitor forms the output low-pass filter for the audio so its important to have the right value. Lately I have been shipping kits with axial (long- ways) package, not radial (side-ways) package. These are longer (see left) and are easy to bend over for soldering. This way there is less confusion. Either way, try to spot the 103 marking
Place the capacitor right next to R7. Ceramic capacitors are non-polarized and can go in 'either way'
Solder and clip the small capacitor leads



	Next is the DAC (digital-analog converter) IC2. This is what turns the data into music. Make sure you pick the DAC to solder in here; it says MCP4921 on it and has a stylized M. The chip has a notch in one end and that notch must line up with the notch in the silkscreen. In this photo, that's on the right.
	Flip over the board and solder in each pin of the chip. The pins are already quite short so you don't have to clip them
	Next is the operational-amplifier (op-amp) IC3. It is used to buffer and amplify the output, so that it can drive a small speaker or headphones. This is a similar-looking chip to the DAC. Again, check that the notch matches the silkscreen notch. In this photo, that's to the left. Solder it in, just like you did with the DAC
A ROW REAL STORES	Next is the 3.3V regulator IC1 that provides a nice power supply to run the SD card. The regulator comes in a semi-circular package, so make sure it matches up with the silk-screened image.



	Solder in both components. Their leads are pretty short so you don't need to clip them.
A B B C C C AUGIO Shield V1.8	Next are the three electrolytic capacitors C1 C4 and C9. Electrolytic capacitors are polarized so make sure they go in the right way! The long lead is the positive lead, make sure that goes into the hole marked with a + as shown here.

Real Provide Andrew Constraints of the second	If you're planning to stack another shield on top of this one, you may want to bend the capacitors so they lay flat.
Plaidaveu/sásm b han.ebeybe [.uuuu/sigm b han.ebeybe [.uuuu/sigm b	Next is the headphone jack. It snaps into place right at the edge of the PCB.
	Solder the jack in place. You may want to clip the legs a little if you can, so that it will sit better on the Arduino.
	Next is the volume potentiometer TM1. This is an audio-type 10K pot. It will slip into place pretty easily.

Solder all 5 pins of the potentiometer. Use plenty of solder so that it has a lot of mechanical strength
Next, break the 36-pin header strip into smaller sections so that the shield can be placed on the Arduino. You can use pliers or diagonal cutters. Clip off 2-6pin and 2-8pin pieces.





Hurren Louis 1/6 Biologia 1/0 Biologia 1/0	Pins 13, 12 and 11 are used to talk to the SD card and can't be changed. The rest of the pins, however, are more flexible. Still, for all the examples on the site we'll be using this wiring, so it is suggested to just go with this. You can use any sort of wire. Solder the jumper wires in place.
	Hooray you are done! Now onto the user manual

How to Use it

Introduction

The wave shield uses SD/MMC cards. They are extraordinarily popular, sometimes even available in grocery stores! They are used in MP3 players, cameras, audio recorders, etc. You can use any card that can store 32 MB to 1.0 GB. A 1 gigabyte card can hold 380 minutes of uncompressed audio for the shield, and costs \$5



You'll also need a way to read and write from the SD card. Sometimes you can use your camera and MP3 player - when its plugged in you will be able to see it as a disk. Or you may need an SD card reader



Step 1: Format

The wave shield needs the SD card to be formatted in FAT16 format

10 Open Explore Search... CANO emovable Disk (J:) Removable Disk (H:) PartitionMagic 8.0 AutoPlay Sharing and Security... 🔗 CVS ۲ 2 🛃 SVN Checkout... **R**TortoiseSVN ۲ Local Scan with AVG EAdd to archive... Add to "Archive.rar" Ecompress and email... Ecompress to "Archive.rar" and email Format... Eject

To format the card, place it into your card reader, then right click on the disk and select Format...

Make sure that in the File system pulldown menu, that FAT is selected and not FAT32

Format CANON_DC (G:)
Cagacity:
483 MB
<u>F</u> ile system
FAT
<u>A</u> llocation unit size
Default allocation size
Volume label
CANON_DC
Format options
Quick Format
Enable Compression
Create an MS-DOS startup disk
<u>Start</u>

And click Start

If you get the **Properties** of the card you will see it is FAT formatted. This card has some files on it so its not completely empty



Intro

The wave shield is designed to play a very specific type of audio. If your music sample is in MP3 format, or 44KHz wav, you'll want to convert it to the right format. This way you will get the highest quality audio

Step 2: Check the file

If you have a wave file already, you should check to see if its already in a proper format. That way you will save yourself some time! In windows, right-click on the file, and select **Properties** then click on the **Summary** tab

80.	wav Properties		? 🗙
G	eneral Summary		
	Property	Value	
	Audio		
	 Bit Rate Audio sample size Channels Audio sample rate 	256kbps 16 bit 1 (mono) 16 kHz	
	Audio format	PCM	
		0	<< Simple
		OK Cancel	Apply

This file is 16KHz, 16-bit, mono PCM. Since thats below the maximum (22KHz, 16-bit, mono PCM) you are good to go. No need to convert the file

OK lets say the file is an MP3 or 44KHz or stereo wave file. We will need to convert it down.

Option 1. Use iTunes

You can do the conversion easily with iTunes (available for Mac/Windows) if you have your music in iTunes already this will be super fast to convert multiple files!

You'll have to set the preferences first, but you only have to do it once



Go to the Advanced->Importing tab. Make sure it is set to 22KHz (or less), 16bit (or less) and Mono channels. Click OK

	Adv	anced		
Ceneral Podcasts Playback	Sharing Store	Advanced P	tarental Apple TV	Syncing
On CD Insert:	General Imp Ask To Import	orting Bu CD	rning	
Import Using: Setting:	WAV Encoder		¢	
	22.050 kHz, 16 t	it, Mono. Encoder		
	Sample Rate:	22.050 kHz 16 bit		nternet
Default	Channels:	Mono Cano	cel OK	
		ne munes aco	ri w.	
			Cancel	ОК

Next find the files you want to convert. Select Convert Selection to WAV from the menu



Then simply drag the sounds onto an SD card

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1100 400		Name		_	Time A	rtist	AL	
		B Sound1		0	0:06			
1 Music	2	B Sound2		õ	0.06			
Movies	3	B Sound3		õ	0.09			
TV Shows	4	Sound4		0	0:14			-
Podcasts	5	SoundS		0	0:14			
Audiobooks	6	M Sound6		0	0:17			10
Nº Radio								
A Rinstoner								31
a wingtones								N
STORE								
ITunes Store	9							SD CAR
e Purchased								
* PLAYLISTS								
I GB of Podcasts								
 Music Videos 							100	
			Except 1 minute	1.2.40	6			
140			o sonys, 1 minute	1.3 010	l			

Option 2. Use SoX

Option 3. Use Audacity

If you dont have or don't want to use iTunes you can convert files (one at a time) with Audacity

This is pretty easy. You can use the free Audacity software - available for windows, linux or mac



Grab it from the download page and install it on your computer

Step 3. Start up Audacity and open the file

Start up Audacity

😝 Audacity			
<u>File Edit View Project Generate Effect Analyze H</u> elp			
	L R •) 💌	-24 -12 0	-24 -12 0
(4) Īt p jt	v		
	4.0	5.0	6.0
•	_		
Project rate: 44100 Cursor: 0:00.000000 min:sec [Snap-To Off]			

Select **File->Open...** and open the file. In my case its an MP3

Select one or m	ore audio files					? 🗙
Look jn:	🚞 my audio		• <	⊨ 🗈 d	* 🎟 🕇	
My Recent Documents Desktop My Documents	1 Hey Jude.mp	3				
My Computer						
(File <u>n</u> ame:	21 Hey Jude.mp3			-	<u>O</u> pen
My Network Places	Files of <u>type</u> :	All files (*.*)			•	Cancel



Audacity will spend some time uncompressing and opening the file and then present you with something like this

Step 4. Split and Mix a stereo track

Next, if you have a stero track, you'll probably want to turn it into a mixed mono track. That way it will sound most like the original. Click on the title and select **Split Stereo Track**



Next, when you mix a track you'll end up adding both of them together. This means that if both sides are loud, you'll get distortion. Reduce the gain on both tracks to -6dB



Drag the label vertically to change the order of the tracks

Then convert both tracks to Mono by clicking on each title. Make sure you do it for both tracks!



Now to mix! From the menu select **Project -> Quick Mix**

		-						
File	Edit	View	Project	Generate	Effect	Analyze	Help	
I ۶ •	· ₩	 Ø * 	Impor Impor Impor Impor Edit I	t Audio t Labels t MIDI t Raw Data D3 Tags		Ctrl-	+I	
	-1:	:00	Quick	Mix				3:00
× 2 Mc 32 M	1 Hey ono, 44 -bit flo: ute	Jud ▼ 100Hz at Solo	New / New S New I New 1	Audio Track 5tereo Tracł .abel Track Fime Track	<			ana na na ang ang ang ang ang ang ang an
<u> </u>	•	+	Remo	ve Tracks				

A few seconds later, you have converted your stereo track to mono!

Step 5. Convert to 16 bit audio

If your audio rate is higher than 16-bit, you will want to downconvert it. Click on the track title and select **Set Sample Format -> 16-bit**

×	Mix 🔽 1.0	and the second state of the second state of a second state of the
N	Name	A second s
- 1	Move Track Up Move Track Down	
L L	Waveform Waveform (dB) Spectrum Pitch (EAC)	
	Mono Left Channel Right Channel Make Stereo Track Split Stereo Track	
	Set Sample Format 🔸	16-bit
Dre	Set Rate 🕨 🕨	24-bit ✓ 32-bit float tracks.
Ducia	at water 444.00	0:00 000000 IC T- 0/41

Step 6. Convert to 22-KHz or less

Finally, make sure the audio file will be saved as 22KHz. If the the track label says 44KHz you will want to convert it.

At the bottom of the window there is a little button named **Project rate:** Make sure this is 22KHz or less



Step 7. Prepare to export

Check the **Preferences** menu item and select the **File Formats** tab. Make sure the Uncompressed Export Format is WAV (Microsoft 16 bit PCM)

Audacity Preferences
Audio I/O Quality File Formats Spectrograms Directories Interface Keyboard Mouse When importing uncompressed audio files into Audacity C Make a copy of the file before editing (safer) © Read directly from the original file (faster) Uncompressed Export Format WAV (Microsoft 16 bit PCM) WAV (Microsoft), Signed 16 bit PCM
OGG Export Setup OGG Quality: 5 0 10
MP3 Export Setup
MP3 Library Version: MP3 exporting plugin not found Find Library
Bit Rate: 128
Cancel OK

You only have to do this once!

Step 8. Export!

Finally, you're ready to export the file. Select Export as WAV... from the pulldown



It may take a few seconds to convert and save the file

Export				
Exporting the	e entire project as	s a WAV (Microso	oft) file	
	Remainin	g time : 0:00:18		
			Ca	incel

Finally, check the file Properties. It should be 16 bit, mono, 22KHz (or less) and PCM format

21 Hey Jude.wav Pro	perties	? 🗙
General Summary		
Property	Value	
Audio		
 Bit Rate Audio sample size Channels Audio sample rate 	352kbps 16 bit 1 (mono) 22 kHz	
Audio format	PCM	
	< Sim	ple
	OK Cancel	Apply

OK! Now you can go to the next step, which is formatting an SD card and copying files over

Troubleshooting & Extras

Getting Stack overflow errors?

These examples are all tested to work with v13 or higher, so try to use that if possible!

Get more RAM & Flash!

Before you try to play audio, you'll want to free up some Arduino RAM, so that you don't end up with a nasty stack-overflow. Especially if you're running a Atmega168-based Arduino!

Follow these instructions on how to get more RAM by reducing the input Serial library buffer. You dont need to do this if you're using an ATmega328 (although, hey it wont hurt!)

Note that the library is pretty big (about 10K) so if you want to do a lot more, I suggest upgrading to an ATmega328. The shield was designed with the expectation that this part would be available.

Generating speech

If you want a human voice in your project, you can use the free generator at AT&T Textto-Speech demo page. It will create a 16KHz, 16-bit audio file so you can use the audio 'right out of the box'

http://www.research.att.com/~ttsweb/tts/demo.php#top

Sound sample library

There is huge a collection of C.C available online. Attribution licensed sound samples! A lot of it is already mono, 16 or 22KHz http://wiki.laptop.org/go/Sound_samples

Digital audio player

This is the simplest example. It plays every audio file it finds on the SD card in a loop. This sketch is also included in the library

PI party!

This example shows how to use the AT&T text-to-speech website to speak the first 2640 digits of pi. The number is stored in flash, each digit is spoken one at a time.

Sample Code

```
#include <AF_Wave.h>
#include <avr/pgmspace.h>
#include "util.h"
#include "wave.h"
AF_Wave card;
File f;
Wavefile wave;
                  // only one!
#define redled 9
uint16_t samplerate;
void setup() {
                        // set up Serial library at 9600 bps
  Serial.begin(9600);
  Serial.println("Wave test!");
  pinMode(2, OUTPUT);
  pinMode(3, OUTPUT);
  pinMode(4, OUTPUT);
  pinMode(5, OUTPUT);
  pinMode(redled, OUTPUT);
  if (!card.init_card()) {
    putstring_nl("Card init. failed!"); return;
  }
  if (!card.open_partition()) {
   putstring_nl("No partition!"); return;
  }
  if (!card.open_filesys()) {
   putstring_nl("Couldn't open filesys"); return;
  }
 if (!card.open_rootdir()) {
   putstring_nl("Couldn't open dir"); return;
  }
 putstring_nl("Files found:");
  ls();
}
void ls() {
 char name[13];
 int ret;
  card.reset_dir();
  putstring_nl("Files found:");
  while (1) {
   ret = card.get_next_name_in_dir(name);
    if (!ret) {
       card.reset_dir();
       return;
    }
    Serial.println(name);
```

```
}
}
uint8_t tracknum = 0;
void loop() {
  uint8_t i, r;
   char c, name[15];
   card.reset_dir();
   // scroll through the files in the directory
   for (i=0; i<tracknum+1; i++) {</pre>
     r = card.get_next_name_in_dir(name);
     if (!r) {
       // ran out of tracks! start over
       tracknum = 0;
       return;
     }
   }
  putstring("\n\rPlaying "); Serial.print(name);
   // reset the directory so we can find the file
   card.reset_dir();
  playcomplete(name);
  tracknum++;
}
void playcomplete(char *name) {
  uint16_t potval;
  uint32_t newsamplerate;
 playfile(name);
  samplerate = wave.dwSamplesPerSec;
  while (wave.isplaying) {
       // you can do stuff here!
       delay(500);
   }
  card.close_file(f);
}
void playfile(char *name) {
   f = card.open_file(name);
   if (!f) {
      putstring_nl(" Couldn't open file"); return;
   }
   if (!wave.create(f)) {
    putstring_nl(" Not a valid WAV"); return;
   }
   // ok time to play!
   wave.play();
}
```