

Electronics for Model Railways



Chapter 14

Adding sound

By Davy Dick

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In memory of Margaret



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Adding sound

Many layouts are miniature masterpieces, yet still seem to lack something. That something is often the extra authenticity that comes from movement (other than trains), lighting and sound.

Mostly, a model railway is a visual experience, watching trains travelling through scenic layouts. We can add to the effect by animating lineside and scenic features using motors and servos. We can further improve the layout further with the careful use of scenic and train lighting.

This chapter looks at improving a layout by stimulating another of your senses – that of hearing. Noise is all around us, yet most layouts produce only the sound of electric motors and points operating. This can be corrected, as everything we hear in real life can be reproduced on your layout. This is a much wider area than may initially be thought and this chapter is really an introduction to the subject.

Layout sounds can be divided into two broad categories.

Train-related sounds

These are associated with the direct running of the railway and include:

- Hissing and chuffing
- Diesel engines
- Horns and whistles
- Wheels squealing round tight corners
- Station announcements

Some of these sounds are more easily created using DCC sound decoders, while others can be easily created for both DC and DCC layouts.

Scenic-related sounds

These sounds add to the general ‘feel’ of the layout and might include:

- Factories and mills
- Quarries and mines
- Traffic
- Water
- Farms
- Weather

All of these can be created for both DC and DCC layouts.

Sound Types

- General sound categories include:
- Man-made sounds (e.g. guns, machinery)
- Natural sounds (e.g. water, thunder, rain)
- Ambient/Environmental/Atmospheric sounds

This involves the combination of various sounds to create the overall effect.

For example, a rural scene might combine the sound of a brook with bird and/or animal sounds, while an urban scene might mix traffic noise with general population hubbub.

Likewise, a fairground needs not only organ sounds but the sounds of children and laughter while a factory complex will mix a variety of mechanical sounds.

- Incidental (i.e. one-off) sounds, such as gunshots, foghorns, dogs barking, crossing bells, station announcements

Sound in perspective

Before we start, we must bear in mind the importance of 'getting it right'.

The indiscriminate addition of sounds can easily produce a horrendous cacophony which seriously detracts from the experience.

Just because you can reproduce the sound of everything on a layout does not mean that they all have to be played, at the same volume, at the same time.

The coherence principle rests on the maxim that *'less is more'* – i.e. adding too many unnecessary sounds impedes the user's appreciation of the main content. This situation can be avoided by a few simple rules.

Layout sounds should be:

Appropriate

The sounds should add to the layout, not detract from it. A layout may offer a range of possible sources of sound. However, it is likely that balancing some of the most predominant sources would suffice. Even then, common sense should prevail. For example, it is unlikely that you will hear factory sounds on a day that church bells ring. In the words of Joe Meek (1960's record producer)

"If it sounds right, it IS right!"

Atmospheric

The sounds chosen must reflect the nature of the layout. While layouts may have some elements in common (e.g. vehicles, people, stations, etc.), the dominant sounds must be those most sympathetic to the layout's character. Rural, urban and industrial layouts all have different characters, as do different periods in time. Better to choose the sounds that most characterise the layout than trying to include every possible sound. The sounds chosen should be both harmonious and proportionate.

Scaled

Your layout is a miniature of a much larger area. The baseboard may only be a few feet deep but that will represent quite a scaled distance. The addition of low relief building and detailed backdrops add another layer of perceived depth. The smaller the scale you model in, the greater the scale distance between the front of the baseboard and the rear details.

It follows, therefore, that sounds from the rear of the layout should be quieter than those closer to the front. Otherwise, the overall perception is jarred.

The sounds should provide the impression of depth to a layout. This supposes different volume levels for different distances (see later).

Overall though, we are discussing *background* sounds, so a subdued sound level should apply even to sounds from the front of the layout.

Switchable

Above all, layout sound should be under the control of the operator, who can decide what is played and when. Lots of noise might be perfect for the visits from children and grandchildren but be most unwelcome at exhibitions or during late at night/early morning sessions.

Firstly, then, is the ability to disable all sound at the flick of a switch or the click of a mouse. You may also wish to make individual sounds switchable (e.g. disabling diesel sounds and enabling steam sounds).

After that, there are decisions regarding timing of sounds.

For example, running water is expected to make a continuous sound, while incidental sounds such as bird or animal sounds would only occur at irregular intervals.

Other sound effects, such as whistles and wheel squeals, would only be triggered by events such as a loco entering a specific section of track.



This chapter looks at the following areas:

- Deciding which sounds are appropriate and which reproduction system to adopt.
- Finding and using layout audio clips.
- Sound players.
- Speakers and their placement.
- Triggering of sounds.

Sound options

There are different methods available for adding sound to your layout.

There two different categories;

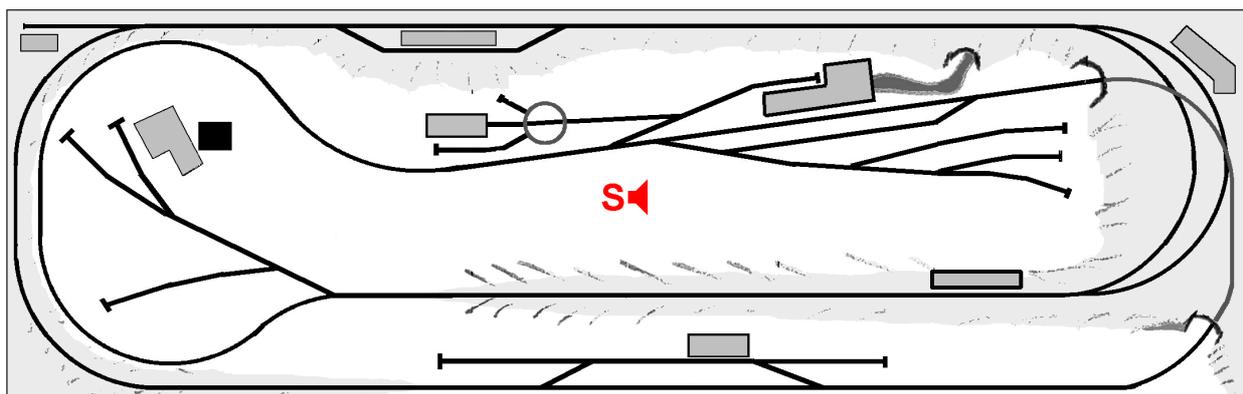
- Trackside sound, on the baseboard
- Sound modules in locos

Trackside layout sound

A number of different approaches have been adopted, both for commercial and home-brew sound systems. The main approaches are shown below, on a typical layout.

Centralised

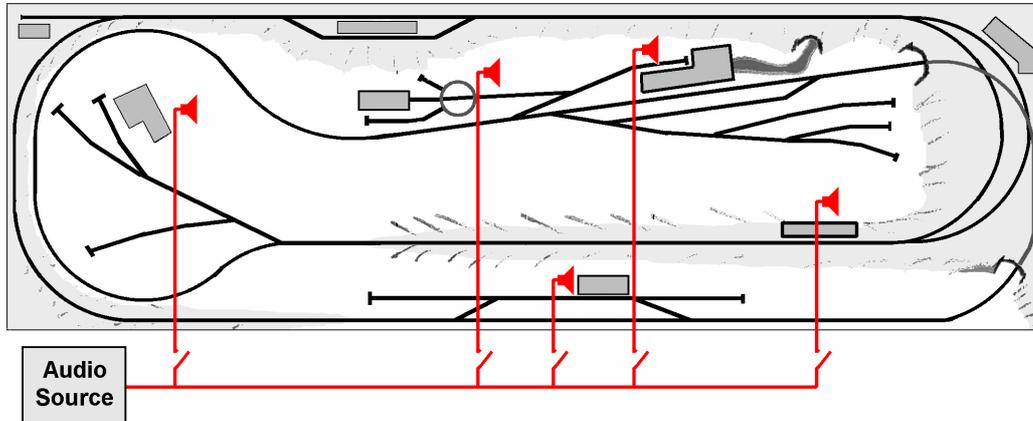
As the name suggests, the layout uses a single audio player and a single loudspeaker placed in the middle of the layout. This has the benefit of simplicity and cheapness (although some expensive commercial systems also use this method).



The downside to this method is that it does not deceive the listener, whose ears detect sounds in stereo. With this system, all sounds are detected as being in the middle, even although the church, factory, etc., may be situated at the extremities of the layout. While this method might work for small layouts, the difference between expectations and reality worsens as the layout becomes larger. It is not very effective in large layouts as the sound does not appear to come from specific objects or buildings.

Distributed

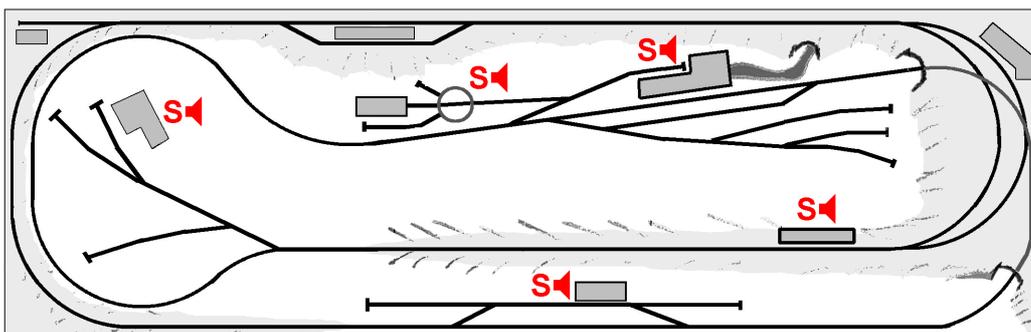
With this system, there is still a single source of all sounds but there are individual loudspeakers situated at each sound location and these are switched on and off to make the sound come from specific areas of the layout. In this way, the sounds are heard from their correct positions on the layout.



The downside in this method is the need to direct the audio from the player to the correct loudspeaker at the correct time, using either manual or electronic switching. Also, only a single sound can be produced at any one time. This reduces the richness of the overall effect. It also prevents continuous sounds (e.g. rivers) from being played.

Local

This system uses a separate sound player and loudspeaker for each location. This allows sounds to be heard from their expected locations and also allows multiple sounds to be played at the same time if necessary. So, a waterfall will have its own sound module, with another module producing sound for the sawmill, and so on. The speakers for these modules are located in, or next to, the supposed source of the original sound.



The modules can be left to play continuously, for background sounds such as with a waterfall or station hubbub. Alternatively, they can be wired to only play when triggered by train-on-track detectors. For example, the sound of a loco whistle/horn or wheel squeals could be produced when a train enters a particular section of track.

They could also be under operator control, switched on/off by MERG CBUS output modules (e.g. CANACC8) or controlled by switches on a control panel.

It can also save on all the extras switch wiring needed with distributed systems.

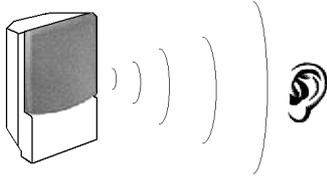
The downside is the extra expense of providing more players, although this not need be prohibitive.

Sound basics

Before moving on, it may be useful to recap on some basic sound concepts.

Sound is produced and carried by the rapid variation of the pressure of the air surrounding the object that is creating the sound.

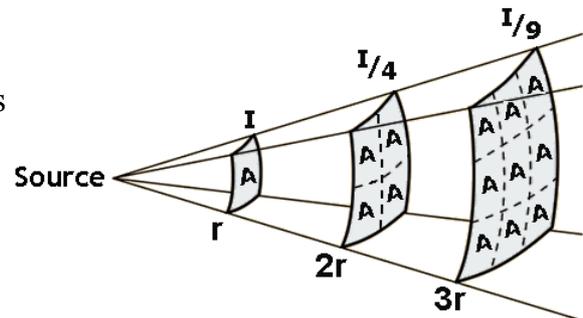
Consider the loudspeaker inside your TV or Hi-Fi. The speaker cone vibrates in sympathy with the audio content. As the cone vibrates, the air next to it is alternately compressed and rarefied. This constant change of air pressure sets up sound pressure waves, which travel through the air until they reach the listener's eardrum.



The air does not actually travel between the speaker and the ear. As the air next to the speaker cone moves back and forth,

it passes on this movement to the air next to it - and so on. It is similar to the water in a pool when a stone is dropped in it. The waves move outwards but the water does not travel outwards.

The knock-on effects of the variations diminish as the sound waves travel outwards, resulting in sound becoming quieter the further away the listener is from the source.



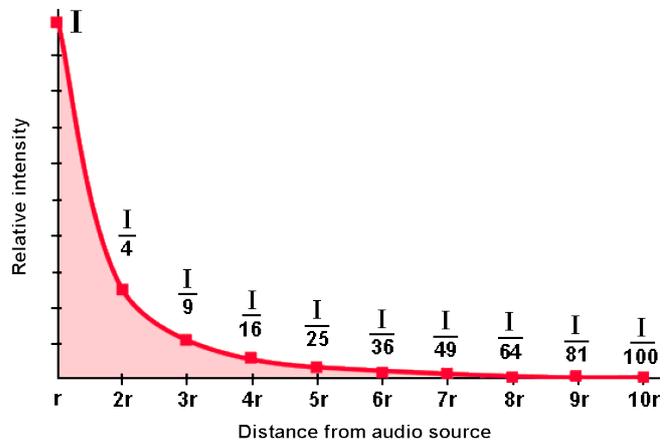
As this illustration shows, the audio source has a sound intensity I at distance r from the sound source.

If we look at twice the distance (i.e. $2r$), the area at r^2 has to influence four times the area as that at distance r .

Similarly, at distance $3r$, it has to influence an area that is nine times larger.

This effect is known as the "Inverse Square Law".

This chart shows just how quickly sounds diminish with distance.



In model railways, this is sometimes considered as 'scale sound'. Although a layout may be physically small, it represents a much larger area (depending on the scale – e.g. 76 times greater in OO and 220 times greater in Z).

Clearly, then, sounds at the rear of a

layout should not be heard at the same amplitude as those at the front of the layout.

Also, if you are standing at one end of a long layout, you should hardly hear sounds from the other end of the layout.

In addition, we have to remember that the dominant impression of a layout should be the movement of the trains, not the sound from various features. Even the loudest sound should not detract from the layout.

In the words of Jim 'Engineers' Wells (of Fantasonics)

"your audience should feel like they want to lean in to hear the sound better"

Commercial sound systems

There are quite a few commercial systems that are specifically designed to produce sounds for model railway layouts. They are generally well specified but expensive.

This image shows a sound module from the GL series produced by ITTP. They are designed for fitting in local situations, with each module producing a different sound. The sounds vary from US loco sounds to lineside sounds such as a construction site, a sawmill, thunder, church bells, and much more.

The module is connected to power, a speaker and a connection for external switching. The sound plays for a maximum of two minutes, although the sound can be looped to play indefinitely.



This image shows the 'Master Blaster' from Model Sounds. It is a combined sound module and loudspeaker in the one case.

It has a built-in battery, making it fully portable – it could be installed in a lineside building or fitted inside a large coach or wagon (probably from 'O' gauge upwards).

It runs for three hours on a charge and can be recharged from a computer's USB port or from a battery charger.

It uses a MicroSD memory card to store the sounds and the different cards can be swapped in to get different sounds (purchased separately).

For details see:

www.modelsounds.co.uk/

An example of a centralised sound module is the Gaugemaster GM780 series. It has a single built-in loudspeaker and can play sounds at random or as DCC functions.

There is an extensive list of sounds as in these examples:

- The GM783 Urban Sounds module can play the sounds of roadworks, an aeroplane flying over, slow moving traffic and general city noise
- The GM780 Lineside Sounds module can play the sounds of point throws, semaphore clinks, signal rodding and trackside sounds
- The GM784 Rural Sounds module can play the sounds of animal noises, church bells, birdsong and general country noise .



Despite the variety of sounds, the sounds cannot emanate next to all the objects that are supposed to be making the sounds. The larger the layout, the more obvious this becomes. A point at the end of a layout operates but the sound comes from the middle of the layout. It is a price you pay for simplicity.

This is an example of a centralised sound module with additional facilities. It is the Pricom Dream Player MK2. At the top of the board is a row of inputs and there is a row of outputs along the bottom.

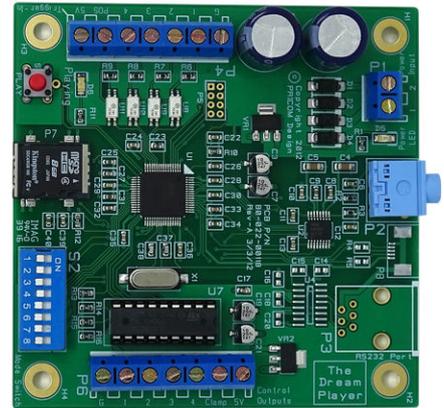
This allows a very flexible set of options. For example, an input can be triggered that plays a sound – and also changes the state of one of the outputs. For example, a change on one input could play a warning alarm while crossing gates close.

The sounds are stored in SD memory cards (see left of image) and therefore are changeable. You do not need to buy samples; you can load any 44.1KHz 16-bit Stereo WAV file that you have.

Although very flexible, its big downside that it has only one audio output (see right of image); this makes it less suitable to cover a large area of a layout.

For details see:

www.pricom.com/audio/players.shtml



An even more versatile, and more expensive, sound module is the RLW 4400 SLITE Animation Board from Republic Locomotive Works. It has two audio outputs that can be used independently or used in stereo. The sounds are stored in a micro SD memory card.

It also has 12 input channels and 16 output channels and can be programmed by a computer through the serial port to carry out sequences of operations.

The program sequence is stored in the SD card.

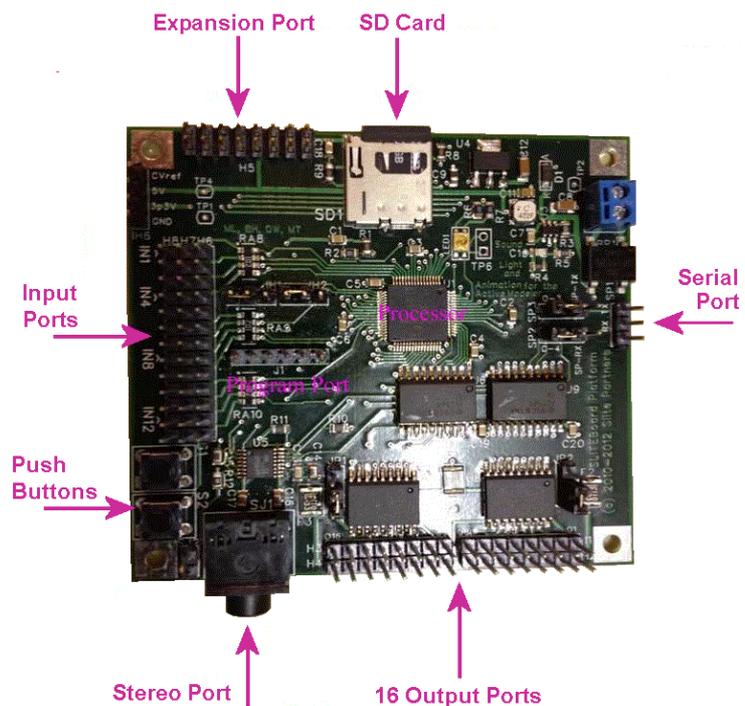
The board has its own microprocessor that can run through a pre-programmed set of sounds and output switching.

The module provides facilities such as PWM (pulse width modulation) for dimming LEDs or setting the speed of motors, A/D converters to read analogue changes such as light levels, and more. It is not currently marketed in the UK.

The top-of-the-price-range trackside sound system is the Surroundtrax DSP-80. It is a distributed sound system, as it places six speakers around the layout.

Each channel can produce sound independently of the others.

Although it is a trackside system, it aims to provide the sounds of locos as they move round the layout.



It requires additional track block sensors, one for each speaker. As a loco enters a track block, the main module is informed and sound is directed to that speaker. Cleverly, as the loco moves from one block to another, the sound in their allocated speakers cross-fade (the volume declines in one speaker while increasing in the other). This provides a smooth stereo-type sound transition.

Unfortunately, six speakers may still not be sufficient for larger layouts.

It uses DCC technology, but without fitting sound decoders in each loco. This may be an advantage for those with large numbers of locos, or limited skills in fitting decoders.

On the other hand, it does not provide the large variety of options that are available in loco sound decoders. Also, it is limited to loco sounds only, with no lineside or other effects.

It is a very expensive system and further details are at:

www.soundtraxx.com/surround/

Limitations of commercial options

Most commercial modules only play built-in sounds with no provision for the user to alter or add their own. The Trax DHM-12 and the Express Models EXX/TTW each have two preset sounds, while the Quasar SG01M module provides 4 preset sounds. The GL series modules produce only a single preset sound.

The Master Blaster module uses memory cards, each storing a different sound. These can be purchased separately, although only one can be installed and used at a time.

The top of the range SoundTrax DS-P80 only works with Digitrax block detectors and transponder equipped locos.

DIY sound systems

Despite their considerable prices, commercial modules have practical limitations.

However, there are a number of devices that are easily obtainable, are considerably cheaper and can provide more flexible and user-alterable sound players.

Here are five different devices, of varying complexities, all requiring little or no alteration.

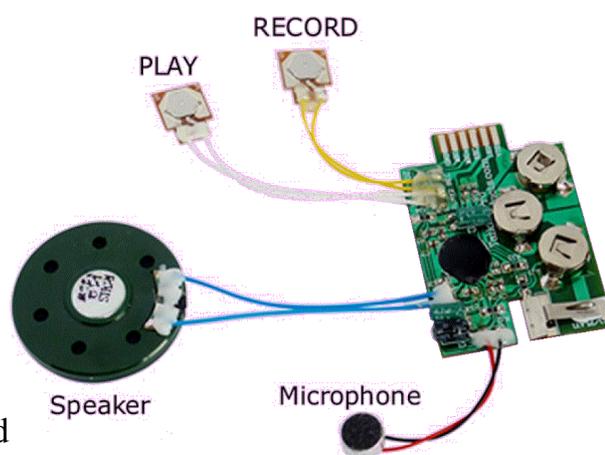
Option 1 - Talking cards

Let's begin with the simplest and cheapest sound player that can be used straight out of the box. Although not of the greatest audio quality it is available for around £2 from eBay. It is intended for incorporating into greetings cards, so that personal recorded messages can be included (*"Merry Xmas Gran, from little Timmy in Australia"*).

It is complete and ready to use, as it has its own built in batteries, microphone, speaker and operating pushbuttons.

It may be supplied bare as shown in this photograph or be enclosed in a plastic case.

The boards are available with slightly differing layouts e.g. some have two batteries and some have three.



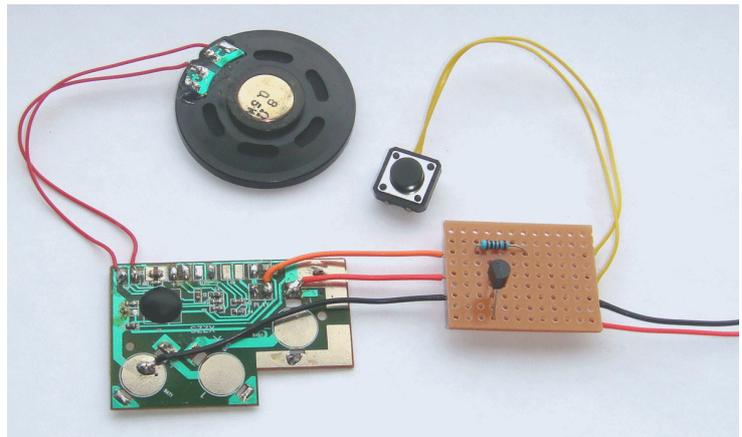


It is extremely simple to use. Just press the record button and hold the mike close to the source of the audio. This may be a sound clip being played through your computer's loudspeaker or be the spoken word (e.g. for a station announcement). The recording limit is 10 seconds.

To play back the recording, simply hold down the play button. Although cheap and simple, this module can be surprisingly effective.

Possible modifications

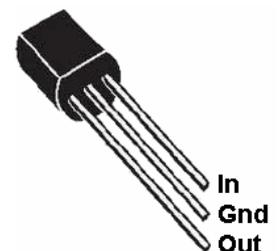
- A likely modification is to remove the on board batteries and replace them with a 78L05 5V regulator so that the module can run from a standard 12V supply.
- I also unsolder the record button after making the recording, to prevent it being accidentally pressed later. It can always be re-connected later if you want to alter the recording.



- It is likely that you will want to operate the module from a distance and this would normally require two wires for the power and two wires for the play switch. This can be reduced to just three connecting wires, as shown in the image. In my module, one of the two 'play' connections has to be taken to +5V to play the sound (the orange wire in the image). This wire can be taken to any device that can provide a switchable +5V (e.g. a switch, relay contacts, output from a train-on-track detector or a CBUS output or an EzyBus output). I included a 1K resistor in series with the orange wire to limit the switching current.
- If you have an old loudspeaker lying around, you will probably find that it is a good replacement for the tiny speaker that comes with the card.
- It should also be possible to remove the microphone and record directly from the computer's 'Audio Out' socket. This is likely to improve the audio quality but I did not pursue the circuit for this.

Wiring a regulator

All five modules operate from either 3.3V or 5V supplies, so these notes on converting to use with a 12V supply apply to all options. Since they all consume relatively little current, they can use the 100mA style regulators (the ones that look like transistors) instead of the larger regulators (the ones with metal tabs). They have three connecting leads, as shown in the illustration.



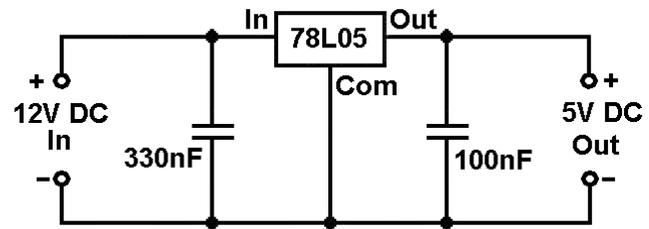
The 12V is wired to the regulator's 'In' pin and the regulated voltage of 3.3V or 5V is available from the 'Out' Pin. The third pin is shown as 'Gnd', also referred to as the COM (common) pin. This is wired to the 0V line of the incoming voltage and also

acts as the 0V of the regulated output.

The 78L33 is used for a +3.3V output and a 78L05 is used for a +5V output. Both have the same size and shape and identified by the printing on them.

Capacitors with values of 330nF and 100nF are fitted across the input and output pins, as shown in the diagram. They are used to smooth the incoming supply and to prevent any possible internal oscillation.

These values are non-critical and some circuits omit them altogether. However, given their small cost, they are best fitted.



Option 2 - MP3 player

Many cheap MP3 players are flooding the market and can be purchased on eBay for around £2. Although designed for playing music on the move, they make great little sound players for a layout.

They store the sounds on a microSD memory card that is inserted into a slot in the player.

These cards can be bought separately for around £2 and can store long sound samples. The sounds can be of higher quality and this player can handle stereo sounds,

The player has a built-in battery that can be recharged from any USB socket (often the one on your computer). The same USB lead is used to copy MP3 sound files from the computer into the player's memory,

The player is intended to be used with headphones so an external audio amplifier needs to be connected to the player's audio out socket.

The steps for this module are:

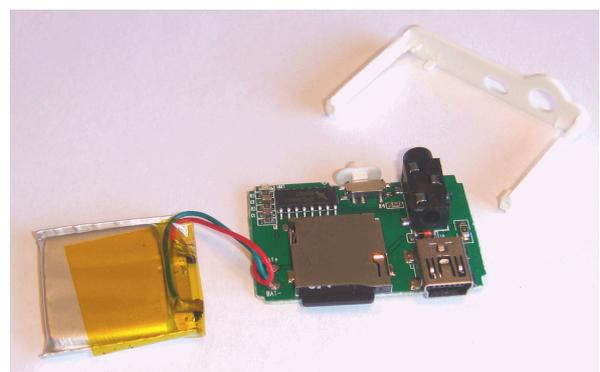
- Find, or create, a suitable sound file in MP3 format.
- Insert the microSD card into the player.
- Connect the player's output to an amplifier.
- Connect the supplied USB cable between the player and the computer's USB socket.
- Copy the sound file into the memory card.
- Remove the cable.
- Operate the power switch to hear the sound being played.

If you are happy to use the internal battery, then the player can be used as is, without any modifications.

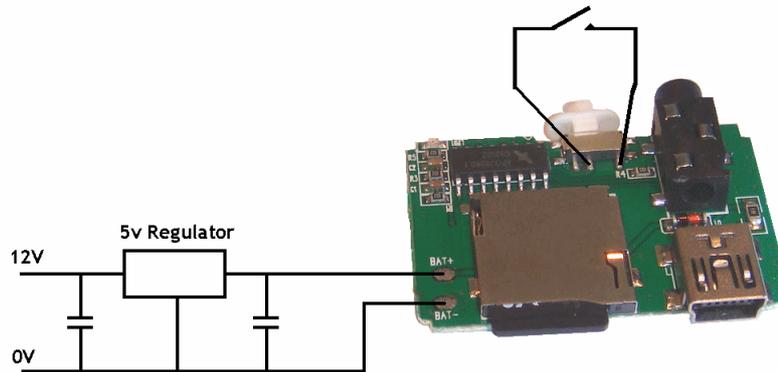
Possible modifications

- Having to regularly recharge the module's internal battery can be a nuisance.

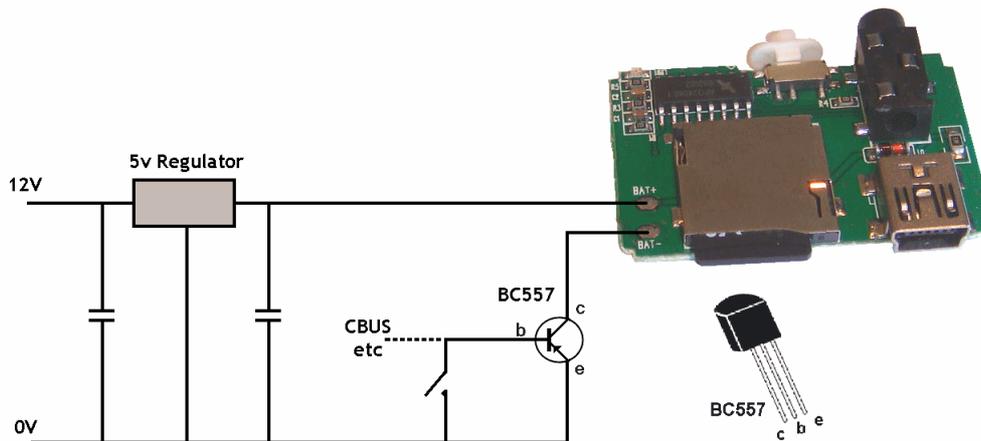
Fortunately, it can be easily replaced with a +5V regulator, allowing the player to be powered from a standard 12V supply. Two small screws secure the player's outer casing. Then, the internal battery can be unsoldered. A 5V regulator circuit can then be wired to the player's power connections as shown below.



- You can wire your own switch in place of the one fitted to the player, to allow you to operate the player from a distance (see the image below).



- You can also keep the switch in the ON position and switch the power to the player. The circuit below shows a transistor, type BC557, being used to act like a switch. When its base connection is taken to 0V, current is allowed to flow between the emitter and collector connections and the MP3 player is powered. The transistor's base can be taken to 0V with a switch, as shown in the diagram. Alternatively, the switch can be replaced with any other device whose output can switch between +5V and 0V (e.g. relay contacts, the output from a CBUS or EzyBus module, the output from a track occupancy detector, etc.).



Option 3 - The ISD1820

This device is readily available on eBay for just over £3. It has a built-in microphone, built-in amplifier and is supplied along with a loudspeaker. The basic version is just 33mm x 42mm.

It needs a power supply of between 3V and 5V, so the 78L05 5V regulator circuit described earlier would allow it to be run from a standard 12V supply.



The module has three buttons. Holding down the red button allows a recording to be made using the microphone. The recording limit is 10 seconds.

The module also has two black buttons, marked as 'single-pass' and 'jog', or 'PLAYE' and 'PLAYL'.

PLAYE stands for edge-activated and even the shortest press of the 'single-pass' button results in the entire recording being played back.

PLAYL stands for level-activated and the 'jog' button only plays back the recording for as long as the button is held down.

Possible modifications

- The black playback buttons are handy for testing the module but it is likely that you will want to operate the module from some external triggering device. Fortunately, there is a row of pins on the board that replicate the job of the buttons. So, for example, taking the REC pin high is the same as pressing the REC button. Raising the PLAYE pin to 5v is the same as pressing the 'single pass' or PLAYE button. Similarly, raising the PLAYL pin to 5v is the same as pressing the 'jog' or PLAYL button. So the PLAYE or PLAYL pins can be wired to switches, relay contacts, the output from a CANACC8, the output from a track occupancy detector, etc.
- The microphone that is on board the module (shown in the top right corner of the module in the photograph) produces reasonable results for the price but there is scope for replacing the mike with a better quality model or with a connection to the audio output from a computer.

Option 4 - The WTV020-SD

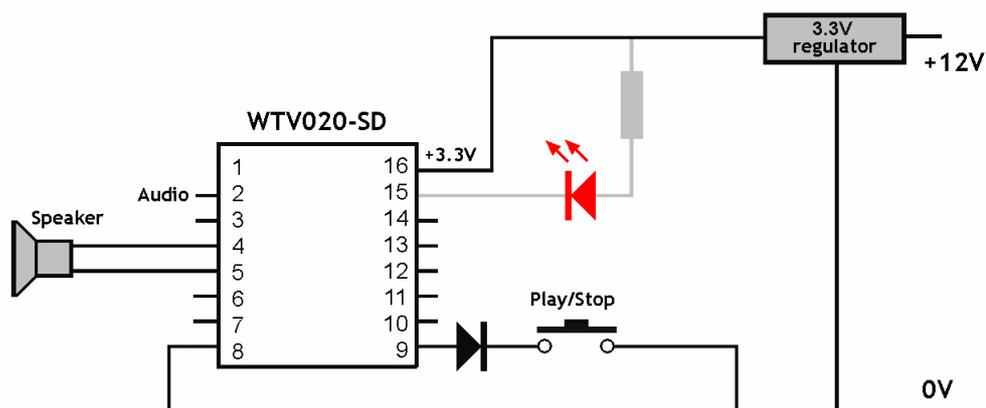
This is another sound player that plays sounds stored on a microSD memory card. It supports up to a 1GB max microSD card so can play long sound files.

It is readily available on eBay for just over £3.

It has a built-in amplifier but needs a loudspeaker. Alternatively, it can be connected to an external amplifier.



It is about the size of a postage stamp and requires few other components, as can be seen from this diagram.

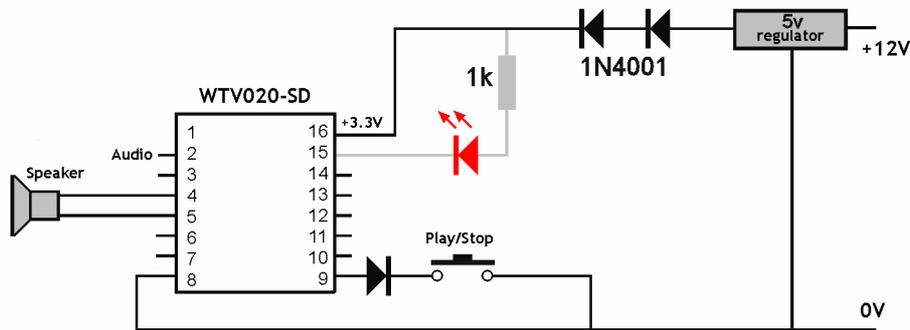


It is a 16 pin module, although most pins are not used in our circuit..

The LED and its resistor are optional. The LED flashes while a sound is playing and would not normally be needed except perhaps for test purposes.

The module would normally be mounted under the baseboard or inside a building, so the LED would not be visible.

The module works from a power supply of between 2.5V and 3.6V, so a 78L33 3.3V regulator circuit described earlier would allow it to be run from a standard 12V supply. Alternatively, you can use a 5V voltage regulator and place two diodes in series with its output.



Preparing the audio files

This and the next module require the preparation of the audio clips prior to them being installed in the module.

With the WTV020, the audio clips have to be stored in the memory card in ad4 format (4-bit adaptive PCM), with a sampling rate of 32KHz. Files should be saved as 0000.ad4, 0001.ad4, etc.

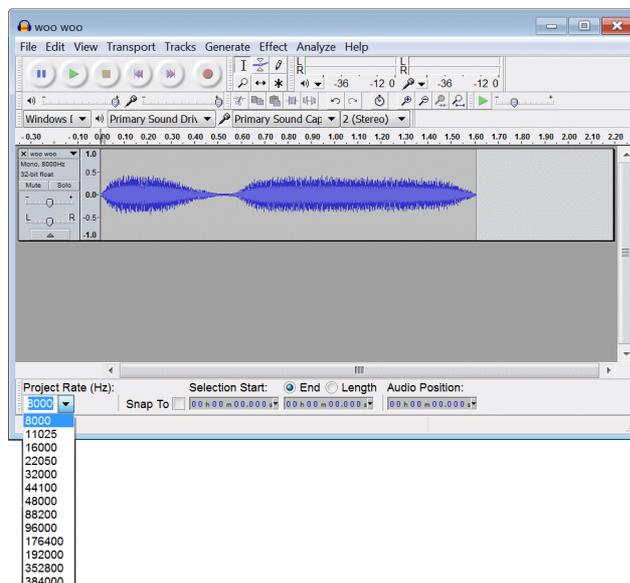
This is easier than it sounds, as there is free software that converts audio files into the required format. The process uses two software packages that are free and easily available. After obtaining, or creating, your required audio file, carry out these two steps.

Step 1 - Convert the file to a sampling rate of 32khz

The most popular software for this is the free open source application called 'Audacity', which is downloadable from

www.audacity.sourceforge.net

When you use the program to open an audio file, the audio track appears in the main window. The current sampling rate of the file is displayed in bottom-right corner. If it needs to be altered (i.e. it is not currently 32000) then 32000 can be selected from the drop down box. The example shows that the file has an 8KHz sampling rate.



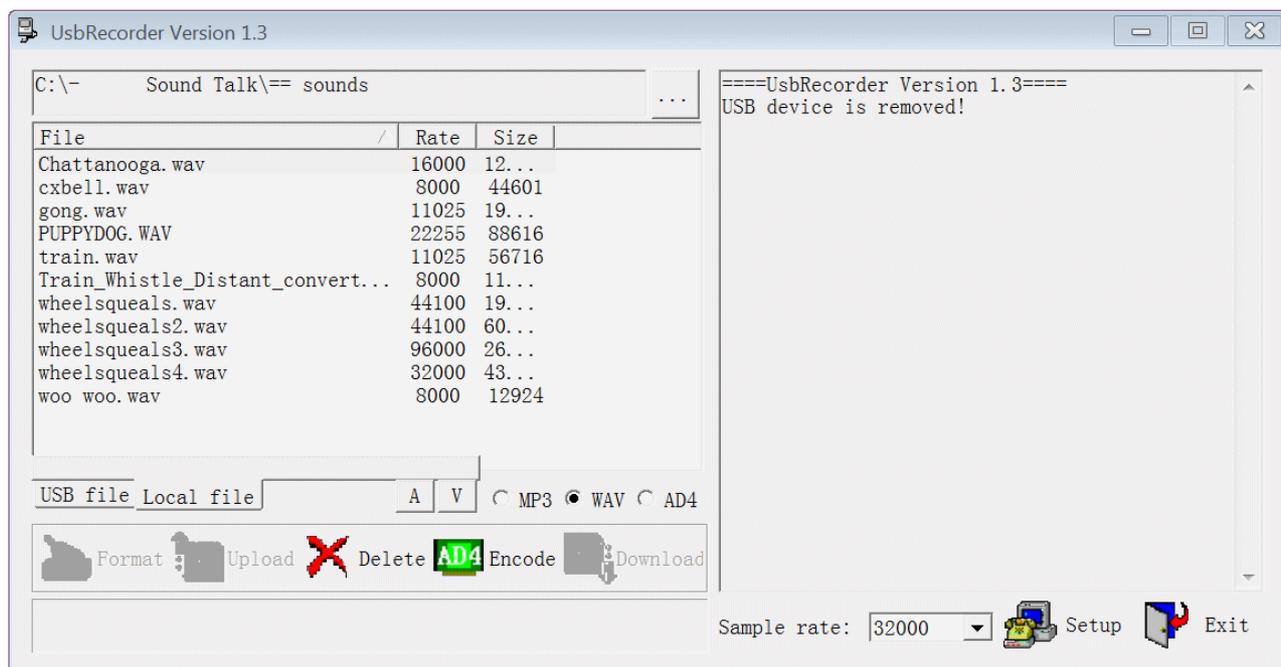
After selecting 32000, the File/Export menu option is chosen to save the file in the new sampling rate.

Although it takes a little effort to learn this technique, it is well worth it, as you can also use this software to edit the audio (convert from MP3 to WAV, mix two or more audio files, remove noise, add echo, introduce fading in/out, etc.).

Step 2 – Convert the file to ad4 format

The conversion software is called USBRecorder and is downloadable from www.waytronic.com/download.asp

The software has a single window that looks like this:



The left-hand pane lets you select a folder to view. From here, you select the file you want to convert.

The example shows that the wheelsqueal4.wav file is already using a 32KHz sampling rate, so an ad4 file version can be created by simply clicking on the 'Encode' button. The newly encoded file would be called whealsqueal4.ad4 and can then be renamed 0000.ad4.

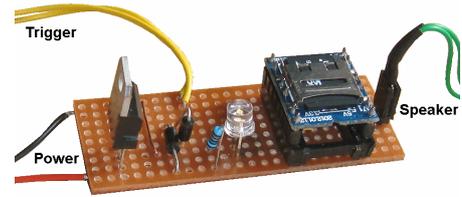
Installing the audio clip

Now that you have the audio file stored in your computer in the correct format, you need a way to copy the file into your memory card. If you already own a digital camera, MP3 player or phone that uses a microSD card, then you are probably familiar with copying files between the device and your PC (probably using the computer's USB port). You simply take out your existing microSD card from the device and insert the card you will use with the sound module. After copying the file to the memory card, you replace the original card in the device.

An alternative is to use a memory card reader (about £1 on eBay) to copy the file into the microSD memory card.

So, the steps are:

- Prepare the audio file as described above
- Copy the audio file onto the microSD memory card.
- Insert the microSD card into the WTV020 player.
- Connect a loudspeaker to the player's pins 4 and 5, or connect to an external amplifier using pins 2 and 8.
- Connect to a 3.3V power supply (see the diagram).
- Press to play, press to stop.

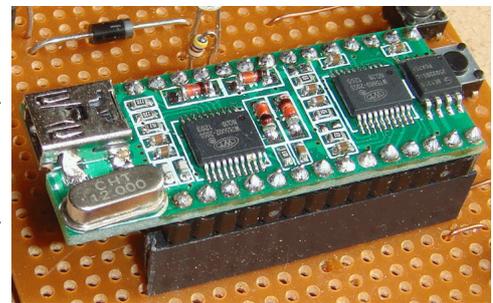


Option 5 - The WT588

This is the most expensive module (around £6 from eBay) but also the most versatile. Multiple configurations are possible, with the example showing four different inputs, allowing it to play four different audio tracks.

It has its own built in memory (no need for a memory card) and built in amplifier (although you have to supply your own speaker).

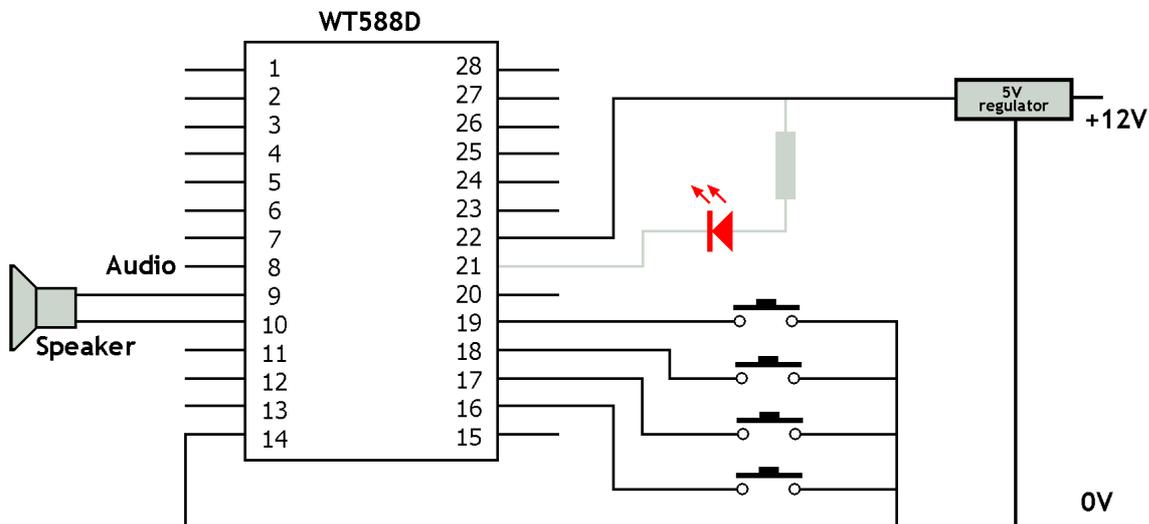
The image shows one of these modules mounted on a piece of stripboard. In the top left of the board, above the crystal, is a miniature USB socket (a Mini-B socket). This is used to transfer the audio clips from your computer into the module.



The pin numbering starts from the pin nearest the crystal. Pins 14 and 22 connect to the 0V and 5V respectively.

Pins 9 and 10 can be connected to an 8 ohm/0.5W

speaker. Alternatively, pins 8 and 14 can be used to connect the module to an external amplifier.



Linking sounds to switches

The module can store multiple sounds and so has to be told which sound(s) to play when a particular key is pressed.

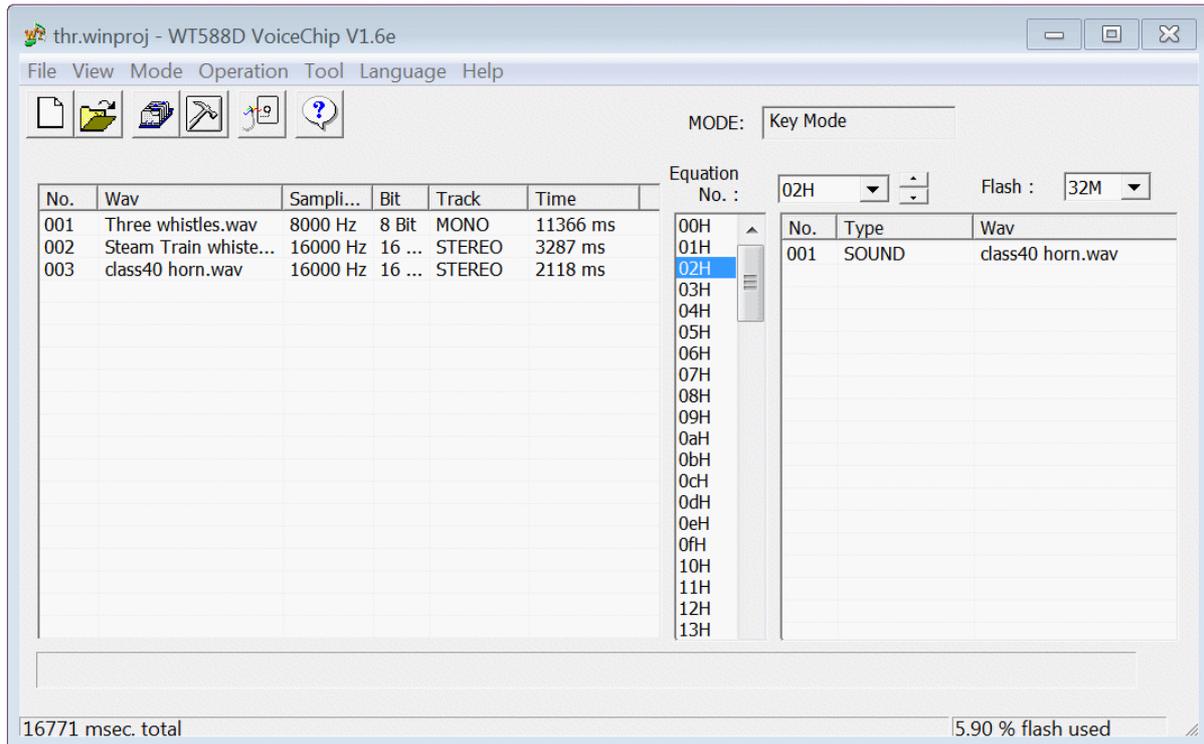
This is achieved with yet another piece of free software and it can be downloaded from:

www.waytronic.com/download.asp

From this menu, choose 'WTD588d VoiceChip Setup'

When run, you may wish to change the default language to English.

The application has a single screen, as shown in the illustration.



When a new project is started, both main panels will be empty.

Right-clicking in the left panel allows you to load the WAV format audio files you need for your project. It will refuse to load an improperly formatted audio file, displaying an appropriate error message

The narrow middle column corresponds to the possible key presses (00H is key 1, 01H is key 2, etc.). In the example, one sound is attached to a key 3. If required, multiple audio files can be attached to a single keypress and they will be played in succession. The program also lets you insert pauses between the playing of a set audio clips, when initiated from a single keypress. Imagine a station making four different announcements, depending on which train arrives (maybe using RFID or CBUS?). When finished, connect your computer to the module, compile the project and write it to the module.

The DFPlayer

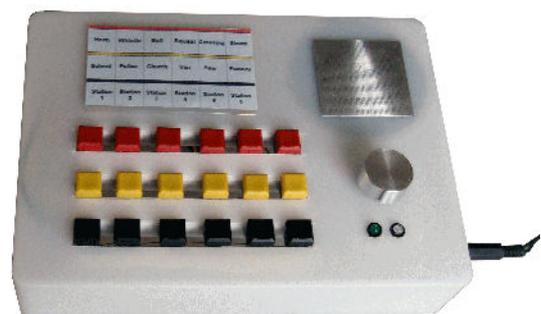
This is a more complicated project, as it requires some programming knowledge and some experience with Arduinos.

It can be set up as a centralised or as a distributed system (see later).

This image is of a centralised sound player, with push buttons (each playing a different sound), a volume control and a power indicator.

A full description of its components and construction can be found in the MERG Journal of March 2017.

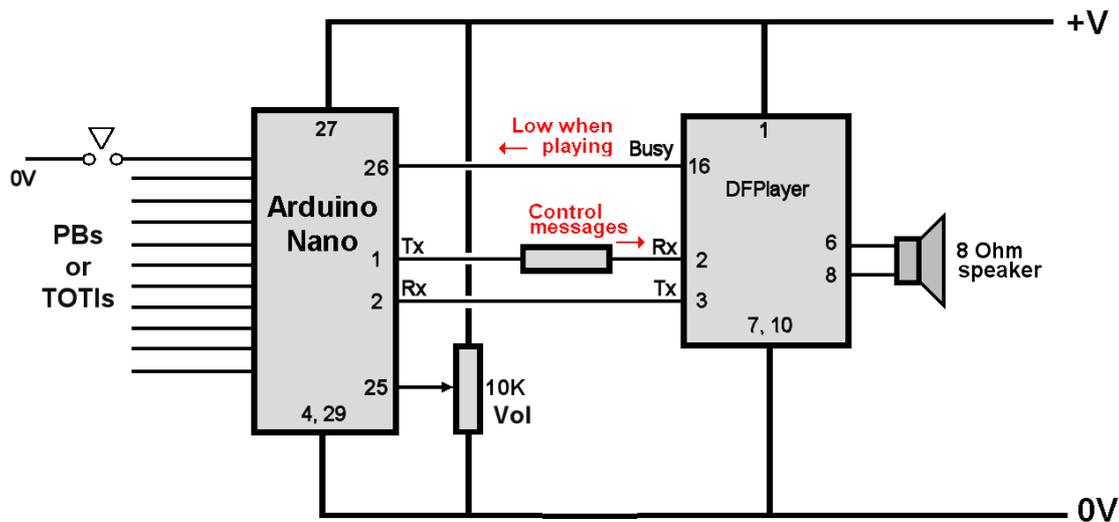
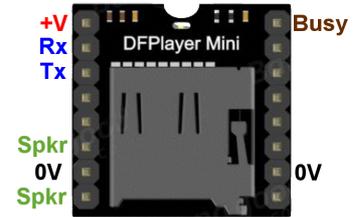
In most cases the module would not be cased and would simply be mounted inside the layout and wired to the appropriate push-buttons and/or sensors.



Sounds are stored on widely available microSD cards and you can download these sounds from the internet or record your own (see later). This allows you to have your own set of sounds that exactly match your own layout's activities.

A 'DFPlayer' module is able to play these sound tracks through its internal mono 3W amplifier is sufficient for most layouts. If you want more volume, or want stereo effects, you can feed the stereo output into a separate amplifier.

The capacity of your microSD card depends on the length and quality of your sound files. Older 128MB cards are cheap and can easily handle 18 normal-sized sound samples. If you want to play long ambient background sounds, then buy a card with a greater capacity.



The DFPlayer module is controlled by an Arduino Nano module.

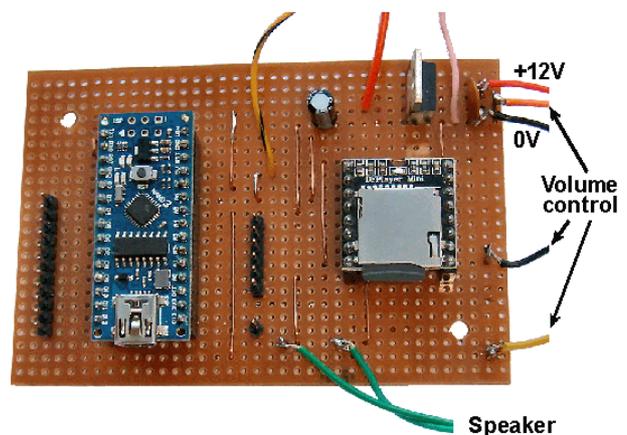
The Nano provides 17 different trigger inputs, some being shown in the diagram.

Twelve of these inputs can be connected to push buttons or other digital sources (see later).

Six other inputs can be connected to either digital or analog sources, which trigger when the voltage falls below 2V.

The Nano detects which input has been triggered (i.e. brought down low) and sends a message to the DFPlayer, telling it which audio track to play. When input 1 is triggered, the module plays track 1, trigger 2 plays track2 .. and so on.

Since tracks are likely to run for different lengths of time, the DFPlayer brings its pin 16 down to 0V while a track is playing. During this time, the volume control can be used to adjust the volume. The Nano waits until the track is played (i.e. pin 16 reverts high again), but triggering another input will play the new track selection instead.



The microSD plugs into the DFPlayer and stores an audio sample for each trigger. If you only want to use nine triggers, for example, you only need to put nine sound samples on the card.

Adding sounds

The next stage is to add sound files to your microSD card.

Where to get them

There are many web sites that offer free sound effects for downloading (see later).

Files should be in the MP3 format and the DFPlayer supports a wide range of commonly-used kHz sampling rates (8,11.025,12,16,22.05,24,32,44.1, and 48). In almost all cases, the files will be one of these formats and nothing needs to be done apart from downloading them.

Even with the large range of available files, you may still have extra requirements. For example, you can record your own station announcements or factory, machinery or lineside sounds. Have a look at the '*Audacity*' program for editing your clips.

How to install them

If you have an old mobile phone lying unused, you can recover the microSD card from that and save yourself the cost of buying one.

Now you have to make a connection between your computer and your microSD card. Your computer may already have a socket for plugging in memory cards. If not, you can buy an SD card adapter. Alternatively, you may be able to plug the card into your digital camera and transfer the files over the camera's link cable.

First, you need to create a folder on the card, called mp3

In the mp3 folder, you add all the audio files you intend to use, to a maximum of eighteen.

The files should be renamed as 0001.mp3, 0002.mp3, 0003.mp3 and so on.

It is permissible to add text after the numbering scheme, separated by an underscore character. Examples may be 0003_buzzsaw.mp3 or 0005_crossing_bell.mp3.

Triggering sounds

Imagine a collection of station announcements, tailored to your own layout. These could be called by pressing a button on your control panel or could even be called automatically.

Or how about producing wheel squeal when trains navigate tight corners, or train whistles/crossing gates bells/factory sounds triggered by sensors. Or anything else that fires your imagination.

As mentioned earlier, the module has two sets of input pins.

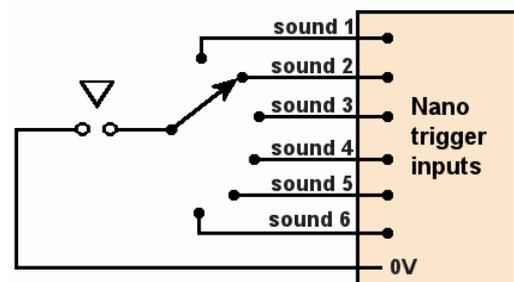
Twelve of these pins, marked in blue on the illustration, can be switched with a change in digital input from high to zero. Six of the pins, marked in green in the illustration, can be switched digitally or with simply a drop in voltage below 2V.

Push-Button control

This is the simplest way to control the sounds. A push-button is wired between one of the input pins and 0V. The push-button could be mounted on your control panel or anywhere round your layout

An alternative to the push-button, in some situations, could be a microswitch.

In most circumstances, it is best to avoid using an on/off switch as the sound will constantly repeat as long as the switch remains thrown. Of course, you may decide to have one audio track as a background



ambient sound and be happy that it repeats itself to avoid having a very large audio file. In that situation, you would have to throw the switch off to allow other sounds to be activated. You may also prefer to use a rotary switch where you select the sound you require then press a push-button.

Train detectors

Their use offers automatic sound triggering. For example, playing the sound of a train whistle, when the train approaches a *'Whistle'* sign or the sound of a conveyor belt when a train enters a factory.

If the train detector's output switches between 5V and 0V, it can be directly connected to the Arduino's input pin. Suitable detectors include reed switches, MERG's laser detector kit, Infrared detector kit and DCC block detector kits.

If you want connect some other train detector module but are unsure of whether its output would be suitable for the Arduino's input, you can fit an opto-isolator between your detector module and the Arduino's input.

Note on opto-isolators

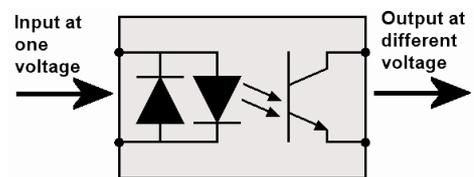
If you are trying to connect two modules together that work on different supply voltages, you have two potential problems:

- Module A does not have a high enough voltage swing to operate module B.
- Module A's output is too high and damages module B.

Opto-isolators are a cheap and very effective solution.

Both modules connect to the opto-isolator but the only link between the two is the light from the opto-isolator's LED hitting the photo-sensitive receiver. When module A passes current through opto-isolator's LED, module B detects the change in voltage across the opto-isolator's photo-detector.

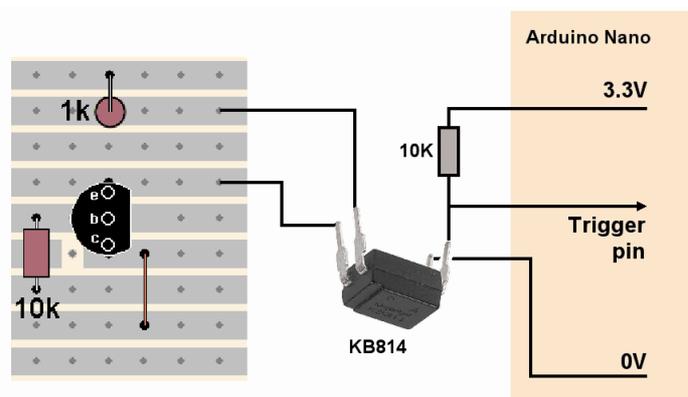
That way, the LED can be switched from a low voltage source and this switches a higher voltage in the other module – or vice versa.



Example

MERG's light detector module works at 12V while the Arduino Nano is on 3.3V or 5V (depending on the model)..

The LDR TOTI can still be used if you removed the LED and replace it with a KB814 opto-isolator, as shown in this illustration. The other side of the opto-isolator is powered from the 3.3V supply of the Nano.



Note

The MERG single DCC detector and the TOTI2 already have an opto-isolators on board.

Other inputs

Apart from switches and train detectors, other inputs can trigger the DFPlayer's sounds.

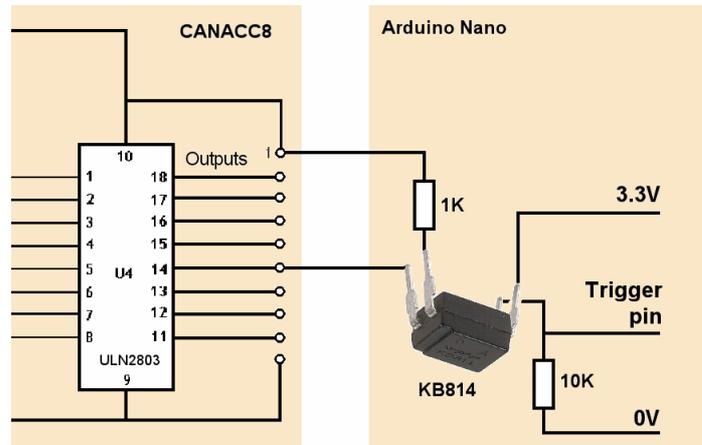
CANACC8 input

Opto-isolators can also be used to connect the sound controller to a CANACC8 module.

This is required since the CANACC8's outputs can be a voltage between 5V and 12V. If a KB824 is used, the CANACC8 can be used to play eight different sound files. Imagine trains arriving at stations with JMRI or RocRail making the appropriate station announcements.

Notes

The CANVout module has 5V logic outputs and is safe to interface to the Arduino.

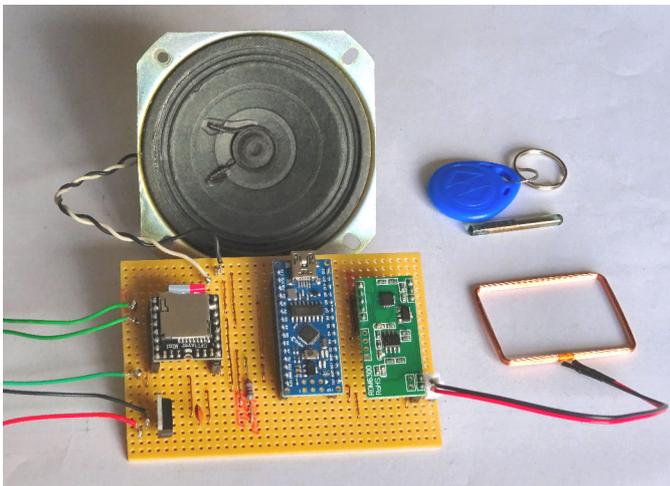


EzyBus input

Like the CANVout, the output from an EzyBus Output Module changes between +5V and 0v and can be directly connected to the input of the Arduino.

RFID input

The MERG Journal of September 2017 provides details of how to adapt the DFPlayer to act as an automatic station announcer.



This module makes station announcements specific to the particular train arriving at the station. The system uses RFID (See the chapter on Radio Frequency Identification). The RFID reader is under £2 on eBay and the tags are also cheap. The reader is placed in the track at the station and each train has its own tag fitted. Since each tag has its own unique code, the Arduino can link a particular code to playing a particular audio clip.

The steps are:

- Find out the unique codes of the tags you will be fitting.
- Fit the tags to the locos/coaches/wagons, noting the code used for each.
- Record your own station announcements.
- Match the tag codes to each announcement.

Extra facilities

Using an Arduino means we can insert extra code so that it not only controls the DFPlayer but also accepts different inputs, controls other outputs, makes decisions, controls the timing of events, controlled fading of sounds, and so on.

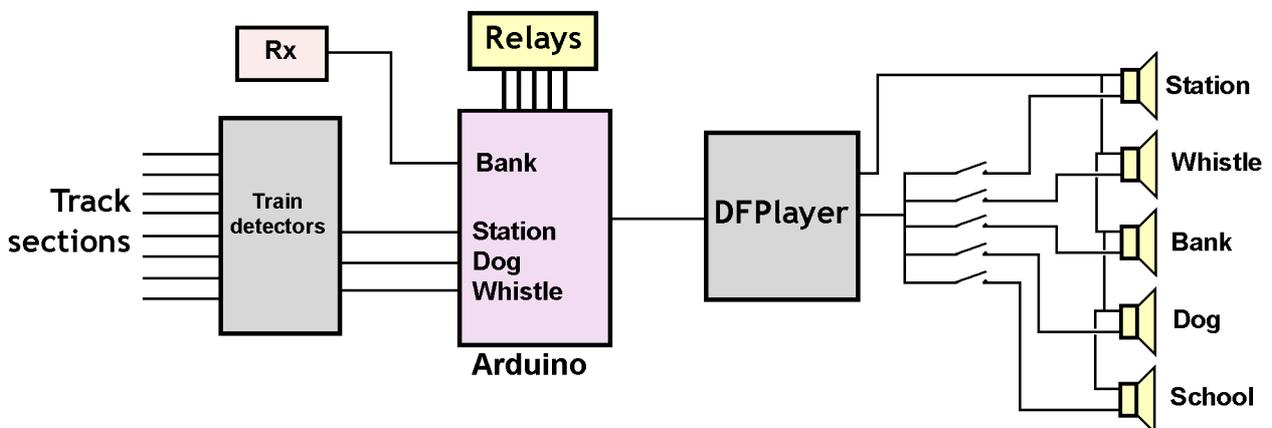
Example

The *'Uppeen Doon'* demonstration layout includes features such as a station, a school, a building site, a sawmill, a *'Whistle'* sign, a dog barking at the passing train, and emergency vehicles round a burning bank building. All of these features are enhanced by adding sound effects (not all at the same time and not at any great volume!).

Some sounds, such as the buzz saw and the jack hammer are triggered when the animation starts. Others, such as the station announcements, the barking dog and the steam whistles are triggered by train detectors.

Yet others, such as the school effects and the fire engine are triggered at random. The school effects include the school bell, the sound of children in the playground and the inevitable ice cream van.

Even though the layout is small, the public can discern the direction of the sound source, so the idea of a single speaker was discarded. Instead, speakers are placed at the appropriate locations round the layout and the Nano switches the DFPlayer's audio output to the required speaker using reed relays.



DIY Comparisons

DIY sound players are of varying complexity, quality, length of audio, price, etc. This table gives a broad comparison of the different types.

As always, it is a matter of choice and ability – but think of the possibilities!

	Greeting card	MP3 player	ISD1820	WTV020	WT588	Arduino
Play time	10 secs	Long	20 secs	Long	Long	Long
Audio source	Mike	microSD	Mike	microSD	Memory	microSD
Type	Mono	Stereo	Mono	Mono	Mono	Stereo
Looping	N	Y	Some models	Y	N	Y
Amp	Y	N	Y	Y	N	Y
Speaker	Y	N	Y	N	N	N

Sound sources

Some commercial sound systems are supplied complete with their own embedded sound(s). Often, their audio content cannot be altered, although some allow additional sounds to be purchased.

Apart from the expense, the available sounds can be limiting.

Others sound players, including DIY versions, need to have suitable sound files installed in them.

In these cases, you have to find your own sound files, usually from the Internet.

The most likely options are for commercial (but royalty-free sites) and public domain and free download sites.

Here are some commercial “pay per sound” sites:

www.pond5.com/royalty-free-music/

www.sounddogs.com

www.soundsnap.com

Here are some free download sites:

www.grsites.com/archive/sounds

www.audiomicro.com/free-sound-effects/free-transport-and-vehicles/free-train

<http://trainweb.org/mdamtrak199/trainsounds.html>

<http://20th-century-fox-sound-effects.freownloadmp3.net/steam-trains-trolleys-etc/>

www.naturesoundsfor.me

Making your own audio files

For those who have the time, the resources and the skills, capturing or creating sound files can provide unique layout sounds.

Capture on site

Visiting specific real world locations (e.g. station announcements at a particular station, sounds from a local factory, etc.) can bring even greater authenticity to a layout.

iPhones has a '*Voice Memos*' app and most Android-based phones have voice recording apps

You can find details of how to use them here:

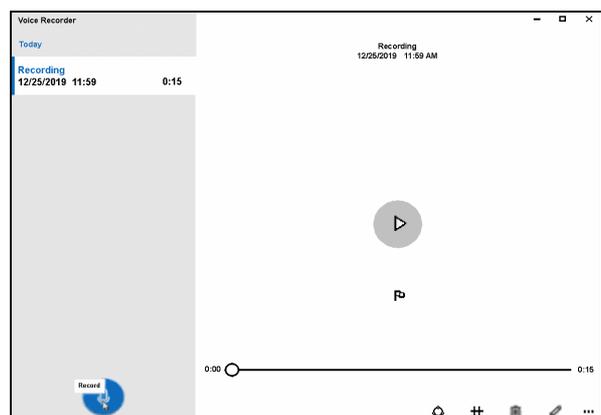
www.wikihow.com/Record-Audio-on-a-Mobile-Phone

Create at home

Windows 7 has a very basic application called '*Sound Recorder*'. It lets you create an audio file from a microphone's input. The microphone may be built into your laptop or you may have to plug an external mike into the sound car of a desktop computer. It saves files in the wma format.

In Windows 10, the application is called '*Voice Recorder*'. It can record, playback, trim, rename and delete recordings.

For some reason, it saves the file in the high-quality but large m4A format, which usually means you have to convert it into a format that other software can use.



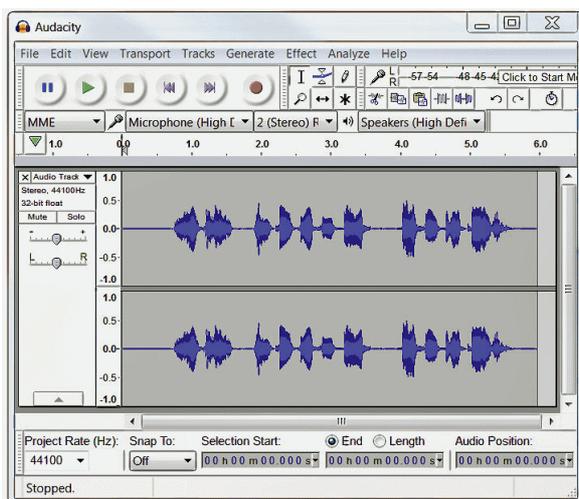
In both cases, clicking on the record icon starts the recording process. This creates a digitised sample from the signals received at the currently active inputs. The time of the sample is shown on screen as the recording is made.

The use of Voice Recorder is explained here:

www.windowscentral.com/how-record-sound-using-voice-recorder-app-windows-10

Neither Sound Recorder or Voice Recorder allow you change the performance settings for a recording.

This free piece of software allows you decide on making a mono or stereo recording, plus settings to control the quality of the recording. It saves file in the well recognised mp3 format.



This image shows the more sophisticated but free 'Audacity' package.

It provides control over the quality settings and can save in WAV, AIFF and MP3 formats.

It provides an on-screen display of the recorded waveform.

If you intend carrying some editing of the audio clip, Audacity is primarily an audio editing package.

The recommended steps for creating an audio clip are:

- Set the dynamic range and the sampling frequency.
- Make tests to adjust the incoming sound levels for optimum recording.
- Make the final recording.
- Replay the clip to check its content and quality.
- Save the clip as a digitised audio file, with a meaningful file name being used.
- Enter the details of the new clip into a database of the sound files used for your project.

The dynamic range and the sampling frequency determine the sound quality and also the size of the file. These are explained in the section at the end of this chapter.

Editing sound files

Even very basic recording techniques can produce useful results, while additional recording and editing skills can further enhance the files. This could include adding echo, compression, file conversion, volume adjustment, noise elimination, panning, cropping, looping, fading up/down.

Once you have your audio files, downloaded or recorded, you have to decide whether the sound is played as a one-off (e.g. a train whistle) or is played as a continuous loop (e.g. running water or city background hubbub).

If you are creating continuous loops, leave breaks in the sound, or have variations in volume; this relieves the listener from a constant barrage of sound.

Alterations to a sound file require the use of a computer and audio editing software.

There are many such applications available and the most commonly used free software is called '*Audacity*' and it can be downloaded for free from:

www.audacityteam.org/

It has an impressive list of features for a piece of free software.

The site even provides manuals and tutorials on how to use the software.

Synthetic sound

For best results, use the actual recordings of the sounds you want to play on your layout. This is specially true for complex sounds comprising multiple sources (e.g. street scenes, football crowds and so on).

Fortunately, some of the more common railway sounds are less complex. That means that we can create an electronic approximation to the real sound – the result sound like the real thing. Such sounds are the hiss of steam, whistles and horns.

For these, you will find a number of circuits available on the Internet. For example:

www.ehow.com/how_12099618_build-electronic-steam-engine-sound-circuit.html

www.circuitstoday.com/dual-tone-train-horn-circuit

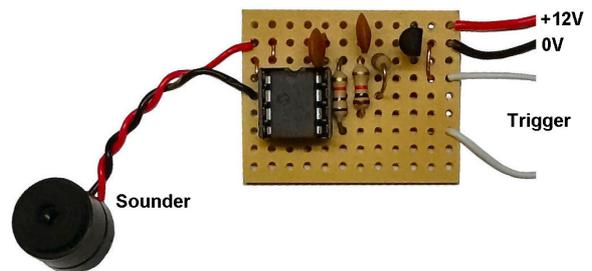
MERG has a steam emulator kit.

Its options are:

- Play the steam sound at random intervals.
- Play when a switch, or other trigger, is operated.
- Play after a five second delay when a switch, or other trigger, is operated.

The first option is useful for unattended operation.

The third option is for use with a train detector. When a loco stops in a section of track, the steam release is heard after a short delay.



Audio players

An audio system comprises three parts:

- The audio file to be played
- An audio player
- A loudspeaker

The player is an important part of the system, in terms of audio reproduction (e.g. quality, mono/stereo), facilities (e.g. looping, maximum length of audio clip supported) and method of operation (e.g. handheld controller, switches).

Most, but not all, players have built-in amplifiers. Some use internal batteries, some use power packs and some get their power from another device (e.g. phone or PC).

Types of speaker

The task of the loudspeaker is to translate audio signals into sound waves. Therefore, larger speakers are able to influence more of the surrounding air than smaller speakers. Of course, large speakers are not always practical on layouts. Fortunately, we don't need high volume levels on our layouts so small speakers are adequate in most situations unless we are going for high-quality reproduction. You may also consider using a larger speaker for reproducing the lower frequency rumbles.

Bear in mind that even a poorish quality clip will sound better on a better speaker, while a high quality clip will suffer when played on a cheap speaker. That said, experience shows that perfectly satisfactory results can be achieved with the many small speakers purchased from eBay. These range from speakers a few inches in diameter to the tiny speakers that are sold as spares for games consoles, mobile phones, laptops and toys.

The picture shows a 38mm speaker, a pair of replacement speakers for a Dell laptop and a replacement speaker for NDS Lite games machine.

For the budget conscious, you can also recover speakers from discarded radios, televisions, etc.



Apart from size, there are two main considerations when buying or reusing a speaker – its impedance and its rated wattage.

The speaker coil commonly has an impedance of 4 ohms or 8 ohms. Impedance is the 'resistance' of a device when used in non-DC circuits; in these cases the inherent inductance and capacitance of the speaker coil has an additional inhibiting effect.

Speakers also have a wattage rating. This is not a measure of the power it normally consumes; it is a measure of the maximum audio power it can handle without producing distortion to the audio or damage to itself.

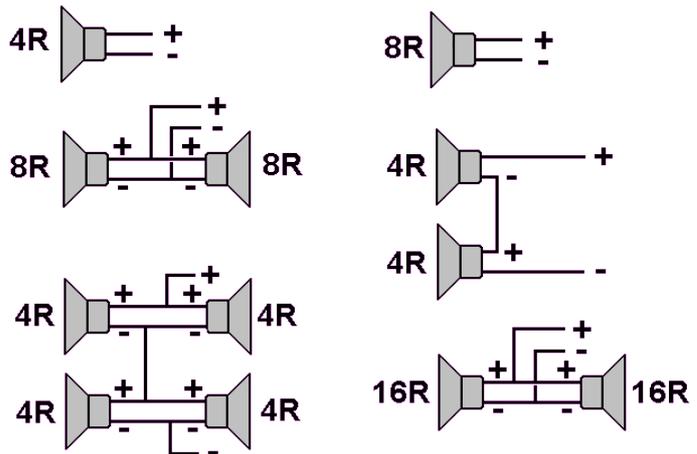
The maximum transfer of power to the speaker occurs when the speaker impedance matches the output impedance of the audio module/amplifier. Using a lower impedance speaker than that specified will usually result in poor audio reproduction and often destruction of the speaker. You can use a speaker with a higher impedance than specified, This will cause no damage but is likely to result in a reduced audio level.

Wiring speakers

Connecting a single speaker to an amplifier or sound module is straightforward, with only two wires to connect them. There may be times when you want to connect extra speakers to a single module (e.g. to spread the sound of thunder or traffic over a large area). Or, you may have some speakers whose impedance do not match the requirements of the module.

This illustration shows various ways to connect speakers.

The images on the left show two 8 ohm speakers wired in parallel to present a 4ohm impedance to the amplifier. If required, you could even wire two sets of 4 ohm speakers as shown. The upper pair and lower pair are both wired in parallel, resulting in 2 ohms each. The pairs are then wired in series, to present a final 4 ohm impedance to the module. Similarly, the images on the right show combinations for 8 ohm systems.



You must not end up with a final impedance that is less than the specified load.

Polarity

The diagrams above show that many, but not all, speakers have positive and negative markings on them, or red and black leads.

A speaker should be connected with its positive lead to the amplifier's positive output terminal and negative lead to the negative terminal.

In practice, the ear is not sensitive to the absolute phase of an audio output. Connecting a conventional speaker in reverse has no impact on how we hear the sound.

However, things change when we connect two or more speakers to the same audio output. If one speaker is wired differently from another, they are said to be '*out of phase*'. This means that at any one given moment, one speaker will be '*pushing*' the air while the other is '*pulling*' the air. If you were positioned mid-way between the two speakers. The sound would be completely cancelled out in theory. In practice, the effect is less less drastic although noticeable, particularly at lower frequencies.

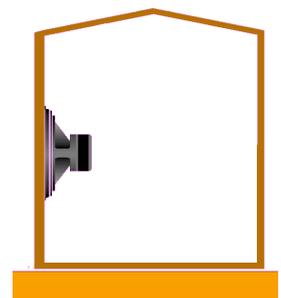
Mounting speakers

You may want to consider these suggestions before placing speakers on your layout.

A loudspeaker works by using a cone shape to vibrate the air. So, you have to ensure that there is no impediment to this free movement.

Firstly, don't just screw the speaker to a flat surface, such as the wall of a building or the underside of the baseboard.

This illustration shows the building being used as the speaker enclosure.

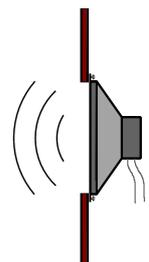


This method will weaken and muffle the sound. We also want to avoid unwanted vibrations – only the speaker should be producing the sounds.

Instead, make a hole in the baseboard/building wall and mount the loudspeaker as shown in this illustration. This way, the material supporting the speaker acts as a baffle board.

A baffle board has two purposes:

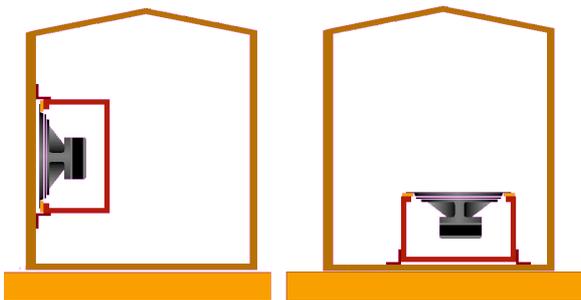
- It increases the volume and clarity. The air is no longer trapped between the cone and the surface it is mounted on and the pressure variations are not absorbed by the material.



- It prevents unwanted interference from the air vibrating at the rear of the cone. Remember that when the cone is 'pushing' air at the front it is 'pulling' air at the rear. We don't want the out-of-phase sound waves from the rear interfering with the wanted sound waves, partially cancelling or distorting them or arriving slightly later to produce unintended echo effects.

The hole in the board should be as large as possible, to allow the maximum effect of the speaker on the air, and should be no less than two thirds of the speaker size.

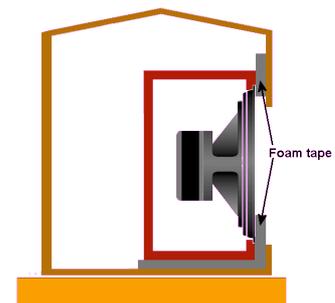
It is best not to screw or bolt the speaker directly to the material, as this can lead to the problems associated with sound transference, such as absorption and unwanted resonance. The speaker can be insulated from the baffle material using doubled-sided foam tape, or a glue gun or silicone sealant.



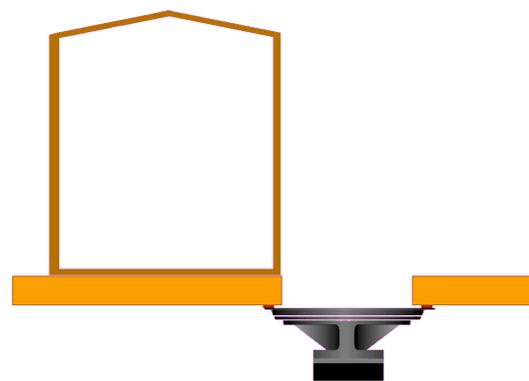
These illustrations shows the speaker mounted in its own enclosure then secured to the side or floor of the building.

Although these may be the only options in some circumstances, they are not the ideal mounting method, as the air is confined in the building and sound is only heard by transference through the building walls and roof.

A better method is shown in this illustration. The speaker is mounted on the rear wall of the building, out of sight. The foam mounting minimises sound transference to the building structure and the hole in the rear wall ensures free movement of air. This is preferable to having speakers pointed directly at the listeners. The sound is being reflected off other buildings or bouncing back off the back scene towards the listener, a situation that often exists in real life. It's a bit like when you first hear a fire engine or the chimes of an ice cream van in the distance. You can hear the sound but can't initially work out where it is coming from. You may identify the general direction but can only pinpoint an exact source when it is closer to you.



An alternative, where the situation allows, is to mount speakers under the baseboard as shown in this illustration. You can position the speaker hole behind objects such as trees, bushes, walls, hills and buildings. It is best to cover the hole with a grill or piece of cloth, to prevent junk from falling into it. If the hole has to be in a less hidden position, it can be covered with a material that allows air to pass through it – such as lichen.



Lastly, it helps to use the largest size of speakers possible, subject to the physical restrictions on a layout. This improves the fidelity of the sound, particularly for low-frequency sounds such as thunder, machine rumble and diesel engines.

Sound switching

A variety of methods are available, depending on the effect that is required.

In addition, some sound modules only play as long as an input is held low/high, while others will play an entire sound clip when an input pulse is detected.

Permanent sounds

Some sound modules can be set up permanently plays sounds on a loop, perfect for running water, country sounds, etc. These modules run continuously when connected to power.

Random sounds

The sound module can be operated from a circuit that sends out randomly spaced pulses to start the module playing. This is useful for the sound of birds, dogs, saws, car horns, etc.

Pulsing circuits are easily found on the Internet and the modules can run unattended.

Timed sounds

The sound module is operated from a timer circuit that runs on its own. This can be useful for any 'scale time' sounds such as a town clock, church bells, etc.

Triggered sounds

This is probably the most useful approach. The sound modules are operated by an event on the layout. For example, a track detector located at a tight bend could detect a train and initiate a '*wheel screeching*' sound. Similarly, a DC or DCC train-on-track detector could initiate a station announcement when a train enters a station. If you don't want certain trains (e.g. goods trains) to initiate the announcement, then you could use a reed relay under the track as the detector and only fit magnets to the train that should trigger announcements.

Synchronised sounds

This approach allows a trigger to synchronise sound to other events on the layout.

So, for example, a signal to a Servo4 board to operate a crossing barrier could also start the sound of crossing bells. Similarly, the flash from a welder (LED) could be accompanied by a '*frazzling*' sound, or a flash of lightning could be followed by a roll of thunder.

Manual operation

The simplest approach is to have a manually-operated switch that powers or initiates a sound. Some users may wish to have control over every individual sound, while others may find it an unnecessary intrusion into the running of their layouts. Even automatic sound systems may benefit from the occasional manually-operated sound. For example, kids may like to press a button that starts all the fairground attractions light/moving while the sound of fairground organ is played.

Of course, you may also consider having a single master switch that disables all audio on the layout (very useful for those late night sessions).

On board train sounds

The previous pages covered trackside sound but it is increasingly common to also add sound modules inside the loco.

There are two types of on board sound module:

- Self-contained units that work on DC or DCC.
- DCC decoders that specially designed for handling sound

Self contained modules

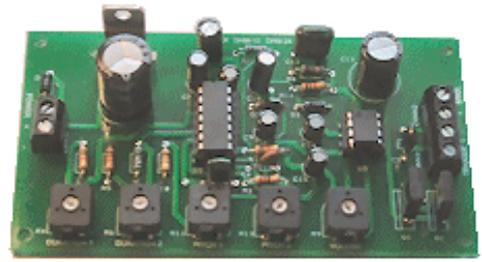
This image shows the Trax Controls DHM-12 and DHM-24 Two-Tone Diesel Horn Modules, working on 12v and 24v respectively. Its size is 110mm x 60mm and it is intended for fitting inside O gauge and larger locos.

The row of small preset controls allow the pitch and duration of each tone to adjusted separately.

For onboard use, it can be triggered by having reed

relay switch mounted under the loco; the switch is operated by a magnet set in the track.

For lineside use, it can be triggered by a pushbutton or by a reed relay embedded in the track that is operated by a magnet mounted under a passing loco.



The SWM-1 module is the Steam Whistle equivalent.

Trax Controls also produce steam and diesel sound modules that reproduce the sound of the locos in motion. The sound content and volume alters as the voltage to the track increases.

In the steam module, the sound varies from a hiss while stationary to a chuff repetition that increases with increasing track voltage.

For more details, see:

www.traxcontrols.com/

Mr Rails produce the MistyMK2 sound module, as shown in this image.

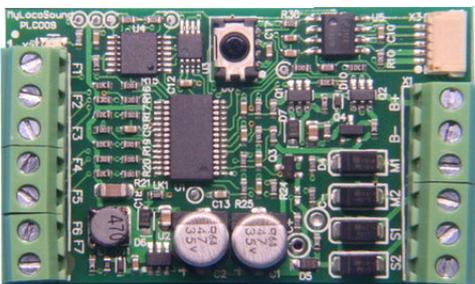
There are two versions, for OO and O scale. They use the same 50mm x 29mm board, with the O gauge version using a larger speaker.

It uses a 9v battery which has to be fitted somewhere on the train.

Like the DHM-12 module, it senses the voltage level coming from the track and produces a chuffing rate that matches it. It has a control for adjusting the voltage/chuffing rate for the particular loco it is fitted to.

For more details, see:

www.mrrails.com/html/misty.html



This is the image of a MyLocoSound Large Scale Sound Module. It measures 55mm x 32mm x 12mm and there are versions for steam, diesel and electric locos.

They are '*load sensitive*' and vary the steam chuff rate or diesel growl/whine with the speed of the loco.

The modules can be operated automatically, by a handheld remote control, or by trackside magnets and reed switches on the locos.

Full details at:

www.micronradiocontrol.co.uk/sound.html

This is the sound capsule produced by Train Tech
 It is self contained with its own built in speaker and battery.
 If required, you can fit an external speaker and battery.
 It measures 25mm x 20mm x 12mm and uses real recorded
 sounds.

The SFX10 model is for steam locomotives, while the SFX20 is
 for diesel locomotives.

The sounds are activated by the module's built-in motion sensor.
 Full details at:

www.dcpexpress.com/sfx10-steam-loco-sound-capsule-p-517.html



DCC sound

If your layout is DC only, then the above sound modules are the types you have to use to
 achieve on board sound.

These boards can also be used with DCC systems but the modules designed solely for DCC
 working, although more expensive, are considerably more versatile and configurable.

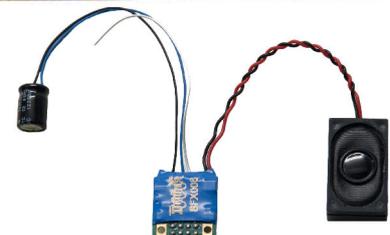
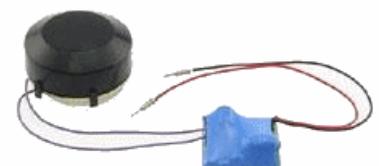
At its simplest, a DCC sound decoder is a small board with sophisticated electronics that has
 two wires that are connected to the loco's power pickups, and two wires that connect to a
 small loudspeaker.

At the other end of the range, some decoders can handle 8 simultaneous sound channels,
 multiple speakers, etc.

Because it is DCC, the decoder's sounds can be triggered by lineside events (e.g. track
 occupancy detectors), manually from the operator's handheld controller, or through a
 computer program. See the chapter on DCC for details on how the system works.

Sound decoders are available in a variety of formats.

- Some are sold pre-fitted in locos, so the user has nothing to do but learn how to
 use it.
- Some handle sound only and a separate loco decoder is needed for the motor's
 speed and direction control. The image shows an MRC
 sound decoder, available in steam and diesel versions.
 The user has to fit and wire the decoder, or have it
 professionally done. Since it is a stand-alone unit, it
 need not be fitted in the loco along with the loco
 decoder; it could be fitted in the loco's tender or first coach.
- Most combine handling sound and controlling the loco's motor (Digitrax, Zimo,
 etc.).
- Some are designed as a drop-in to
 particular loco chassis, as in this
 MRC module for a drop-in to a Kato
 chassis.
- Some are designed as plug-ins to loco decoders (e.g. the
 Digitrax Soundbug plugs into the Digitrax DH165 series
 decoders).



A useful reference is:

www.sbs4dcc.com/tutorialstipstricks/hoscalesounddecoderssummary.html

As with all DCC decoders, values are stored in configuration variables (CVs).

This table shows part of the CV list for the LokSound 4 decoder steam version. There is another table for diesel and one for electric locos.

Function	Soundslot	CV	Range	Value
Sound on/off	1	259	0 – 128	99
Sound on/off	2	267	0 – 128	99
Whistle (playable)	3	275	0 – 128	128
Bell	4	283	0 – 128	128
Coal shovelling / Oil burner (AUX2 Firebox)	5	291	0 – 128	128
Air pump	6	299	0 – 128	128
Station announcement #1	7	307	0 – 128	128
Coupler sound	8	315	0 – 128	128
Cylinder blow	9	323	0 – 128	128
Conductor whistle	10	331	0 – 128	128
Sanding valve	11	339	0 – 128	128
Safety valve	12	347	0 – 128	128
Brake Set / Brake Release (automatic)	13	355	0 – 128	128
Drain valve	14	363	0 – 128	128
Curve squeal	15	371	0 – 128	128
Short whistle	16	379	0 – 128	128
Rail joints	17	387	0 – 128	128

Basics of digitised sound

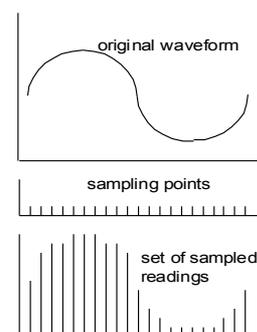
Sound in the real world is analogue; it contains infinite variations in amplitude. However, the computer only stores data in digital format. Therefore, the circuitry on your sound card has to convert analogue sounds into their digital equivalents.

Analogue-to-Digital Conversion

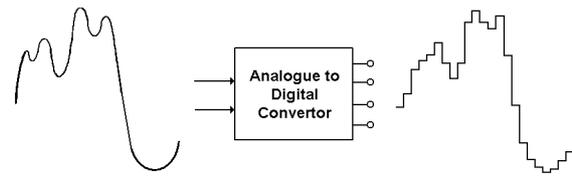
Sound is fed into your sound card in analogue format, usually from a microphone or other audio source.

The conversion is carried out by a chip in the sound card called the 'ADC'- the '*Analogue-to-Digital-Converter*'. The ADC processes samples of the sound amplitude at given points in time and generates a series of numbers that are stored to disc for later replay. The numbers store the amplitude of the sound waveform at progressive points in time during the duration of the sound sample.

The diagram shows a waveform whose amplitude is varying with time. The amplitude of the waveform is measured at regular time intervals and the value read at each time interval is stored. The set of sampled readings represents the changes in the waveform and can be used to recreate the original sound using a DAC (a digital to analogue converter).



This illustration shows an ADC being fed with a sound source. The output shows the effects of digitisation.



The ADC circuitry looks at the amplitude of the audio input at regular time intervals and converts

the value that it finds at each moment into its nearest binary value. Since the input has an infinite number of possible amplitude levels, the ADC chooses the nearest binary level that the chip offers. This is known as '*quantisation*'. The chip carrying out the conversion has a binary output system. In this example, there are four outputs. This allows values from 0000 (i.e. decimal zero) to 1111 (decimal 15). If the chip had eight outputs, it would provide 256 different output levels. With 16 outputs, the resolution increases to 65,536 different values. A 24-bit system provides 16,777,216 possible amplitude levels for each sample. With greater resolution comes greater file size. An 8-bit reading can be stored by a single byte, while a 24-bit reading requires three times the amount of storage.

Two factors determine the quality of digitised sound:

- The dynamic range (i.e. the accuracy in terms of absolute amplitudes).

The larger the range of values stored, the greater the accuracy of the sound's reproduction.

- The sampling rate (i.e. the accuracy of the amplitude at any one instant).

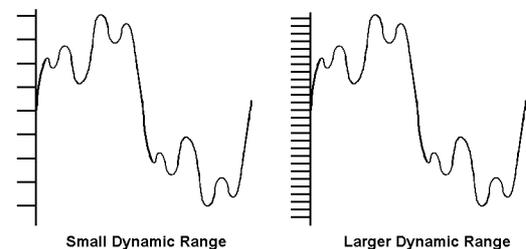
The more samples taken in a time period, the greater the sound's fidelity.

Dynamic Range

The more complex the waveform to be stored, the greater total of different numbers required to store the sound. The span from the lowest amplitude to the greatest amplitude is known as the '*dynamic range*'. This is sometimes also described as '*resolution*' or '*bit-range*'.

In the left diagram, only a small number of bits are allocated to store the waveform, so it is incapable of handling the small amplitude variations in the waveform and these details are averaged away.

In this case, the ADC produces a series of numbers that approximate to the overall waveform but the harmonics that make a piano sound different from a



guitar are lost as are the harmonics that differentiate between different human voices.

The right-hand diagram shows a greater dynamic range, allowing the same analogue signal to be converted into a greater number of digital levels. This allows for a greater clarity of reproduction, as the replayed sound is closer to the original sound.

8-bit resolution is sufficient for some purposes (e.g. voice samples and digital telephony) but does not provide a high quality sound sample. 16-bit provides the quality expected from a domestic audio CD system. This is not a fully professional quality, since each of the 65,536 levels is a linear step while the human ear responds to amplitude variations in a logarithmic way. This means that many of the discrete stored levels do not store changes that can be detected by the ear and so are wasted.

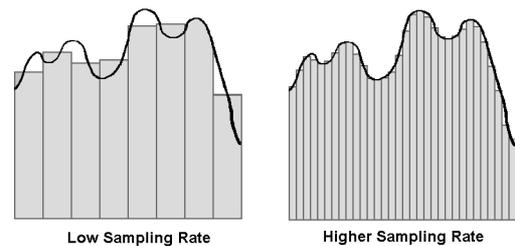
The maximum dynamic range of a digital signal is the ratio between the smallest and the largest value that can be stored, with 0dB as the threshold of human hearing. A 16-bit resolution card has a maximum range of 96dB, while an 8-bit resolution card has only a 48dB maximum range. Since a 96dB range is the approximate range perceived by the human ear, it was the natural choice for CD audio and music.

The logarithmic nature of hearing means that lower amplitudes are more accurately encoded than higher amplitudes. This is particularly suitable for speech processing, where most of the voice information is low levels.

Sampling Rate

The dynamic range determines the accuracy of the amplitude reading at any one point in time. Of equal importance is the frequency of taking these readings. If the readings are too infrequent, an amplitude change will pass undetected. If the readings are too frequent, the conversion will produce a giant series of amplitude readings. The timing of the conversions is known as the '*sampling rate*' and is measured in kilohertz (i.e. how many thousand amplitude conversions are carried out each second).

The left diagram shows the effects of a low sampling rate. The sound sample is converted into six samples with varying amplitude levels. As the diagram shows, the final output is an approximation of the original sound with a considerable loss of detail. The inertia in the loudspeaker cones acts to smooth the transitions



between different output voltage levels from the DAC. The right-hand diagram shows the same sound sample with twice the sampling rate. The audio is now stored in twelve different samples and this is much more representative of the original sound source.

Quality signal processing is based on the '*Nyquist Theorem*' which states that the sampling rate must be at least double the frequency of the highest frequency to be sampled. If, for example, the sampling rate was exactly the same as the input frequency of a sine wave, the input source would be sampled at the exact same spot on the wave for every sample - the source would appear to be a constant value.

Since the average human ear can only hear frequencies up to about 20kHz, a sampling rate of 44kHz is adequate for most uses. Indeed, the human voice itself does not produce useful sound information above much more than 3kHz.

Modern computer sound cards now usually operate up to 48kHz and sampling rates are adjustable down to as low as 4kHz.

For model railway purposes, you can decide on your own needs. Some hobbyists insist on CD-quality reproduction, particularly for home layouts.

In noisy environments, such as model railway exhibitions, some of the benefits may be lost. In addition, good quality reproduction relates not only to the quality of the audio sample but also the quality of the loudspeaker(s) on your layout.

Noise

Noise is unwanted electrical signals that exist in the signal along with the original signal. As long as the noise stays at a low level there is no serious problem - but when the noise increases to the point where it becomes noticeable, it detracts from the final sound.

The measure of sound quality is the '*signal-to-noise ratio*'. This compares the level of noise present in a signal to the amplitude of the wanted signal.

This is measured as a ratio and expressed in decibels.

$$\text{S/N ratio} = \frac{\text{average signal power}}{\text{average noise power}}$$

This ratio is a better measure of quality than simply measuring the noise level. The noise cannot be drowned out by turning up the volume level, as the amplifier will increase both the noise and the wanted signal by the same amount. The sources of noise have to be minimised, so that the signal has little noise content.

The main sources of noise are:

Quantisation Noise - caused by the nature of the sampling process, although much more pronounced with low bit-resolution. The smooth analogue input waveform is converted into a squared off, stepped version as shown. Signals with steep sides are rich in harmonics. This way, extra unwanted signals have been introduced in the digitised waveform. These are high in frequency and are heard as background noise. Increasing the bit range results in more digital levels and reduces the larger steps. For example, using 4-bit resolution results in a signal-to-noise ratio (SNR) of around 24dB, while 8-bit and 16-bit resolutions provide SNRs of around 48dB and 96dB. In practice, the introduction of other noise reduces the system's overall SNR figure.



Component Noise - caused by random or unwanted electron fluctuations within both inactive devices (such as resistors) and active devices (such as integrated circuits). Professional and high-quality equipment uses gold-plated plugs and sockets for all audio and video connections to minimise bad and dirty connections.

External Interference - caused by everything from natural sources (such as cosmic radiation and electric thunderstorms) to man-made sources (such as radiation from electrical appliances). This can be minimised by ensuring that all audio cables are adequately screened.

Crosstalk- caused by signals from one cable being picked up on another cable. This occurs due to the effects of capacitive and inductive coupling between lines that run adjacently. This can be minimised by ensuring that audio cables use screened cable and are not run alongside power cables, loudspeaker cables, etc.

Under-driving- caused by feeding a very low level of signal into an input that expects to be driven by a signal of greater amplitude. The input signal level may be hardly any greater than the general noise level present in the amplifier. In such a case, increasing the gain of the amplifier increases the noise by the same amount as the wanted signal. This can be minimised by matching the output of the driving device to the range required to sufficiently drive the input device.

Distortion

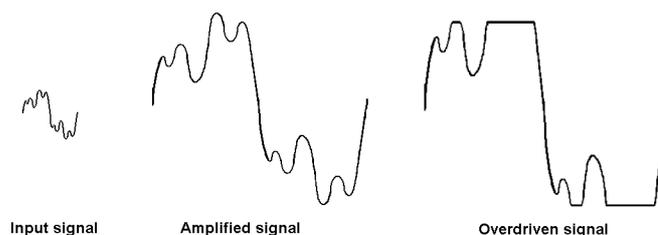
The digitised version of the analogue signal would only never be accurate if the ADC was able to convert the signal to an infinite number of digital values. This would require an infinite number of samples taken every second and an infinite number of binary values to store the reading. This cannot be achieved and the digitising process tolerates the present compromise between quality and practicality. Nevertheless, the current sampling rates and bit resolutions inevitably lead to distortion of the sound quality.

Another source of distortion is the mismatch between signal levels when devices are connected.

A computer sound card may have two audio inputs:

- The '*Microphone*' input, designed for the connection of low voltage drives up to around 100mV (i.e. one tenth of a volt).
- The '*Line In*' input, designed to be fed voltages up to around 1 volt.

The diagrams show the effects of connecting a 200mV input signal to both types of input connector. In the first case, the input signal is applied to the 'Line In' socket. The input swing is within its operating range and it successfully handles the signal. It is passed on to the



digitising circuits. In the second case, the signal is connected to the 'Microphone' socket. This signal greatly overdrives the sound card amplifier circuitry. The signal is amplified to such an extent that it exceeds the maximum output signal swings of the amplifier. This effect is known as 'Clipping' and introduces a great deal of distortion into the signal prior to digitising.

Sound cards often employ 'AGC' (Automatic Gain Control) circuitry to minimise clipping problems. The AGC components detect large input signals and reduce the gain of the amplifier to prevent it being over-driven.

Storage Overheads

Stereo channels are standard on music systems and some televisions but are not often used in model railways. Stereo is used to improve the character of the reproduced sound but requires double the storage space.

	bytes per sec	bytes per min
8-bit mono	44,100	2,646,000
8-bit stereo	88,200	5,292,000
16-bit mono	88,200	5,292,000
16-bit stereo	176,400	10,584,000

The table shows the amount of disk space required for even short digitised samples, when sampled at 44.1kHz. The storage figures in the table can be halved if the sampling rate is dropped to 22.05kHz and can be halved again if an 11.025kHz sampling rate is used.

From the table, it can be seen that a four minute audio sample in CD-quality stereo occupies over 40MB of disk space.

The storage requirements of an uncompressed audio sample can be calculated thus:

$$\text{Storage} = \text{sampling rate (in samples per sec)} \times \text{sample duration (in secs)} \times \text{bit resolution} \times \text{number of channels (i.e. mono/stereo)} \div 8 \text{ (to convert to bytes)}$$

Dividing the answer by 1024 will produce the value expressed in kilobytes, and dividing by a further 1024 expresses the answer in MB.

So, a one-minute audio clip in stereo at 16-bit CD-quality is

$$44,100 \times 60 \times 16 \times 2 \div 8 = 10,584,000 \text{ bytes} \quad \text{- around 10MB}$$

while a one-minute clip in mono at 8-bit with a reduced sampling rate of 22,050 is

$$22,050 \times 60 \times 1 \times 8 = 1,323,000 \text{ bytes} \quad \text{- around 1.2MB}$$

For model railway purposes, many hobbyists will prefer sacrificing some quality in exchange for substantial reductions in file sizes.

Audio File Formats

Most audio players that we can use in model railways are types known as MP3 and WAV (e.g. whistle.mp3 or bang.wav). The three letters after the dot are known as 'file extensions' and they tell you how the audio file inside was processed.

Many audio files that you can download from the Internet are of these types but many have other file extensions. Also, Windows 10 Voice records saves its files as m4A types.

You may find that you need to convert files from one type to one that suits your audio player.

Despite the range of file types out there, they fall into three major categories.

- Uncompressed
- Compressed with losses
- Lossless compression

What is compression

The large files required to store audio files, and the long times taken to download them over the Internet, encouraged the development of a range of storage methods that involved compressing the file on saving and decompressing again on playback. This means that the processed file is smaller and therefore requires less space to store, less time to load from hard disc and less time to transmit over the Internet.

This can be achieved without any loss in the quality of the audio sample.

If desired, the file can be made even smaller although this amount of compression results in some loss of quality.

Some systems offer a range of possible compression ratios, ranging from low-compression lossless samples to high-compression samples with loss of some detail.

Lossless compression may only reduce an audio clip to half its original size. *'Lossy compression'*, on the other hand, can reduce a clip to one tenth or one twentieth of its original size.

Uncompressed

Uncompressed audio consists of sounds that have been captured and converted to digital format without any further processing. As a result, they tend to be the most accurate but occupy lots of disc space.

PCM

PCM stands for Pulse Code Modulation and the file only contains a long sequence of numbers that represent the audio amplitude at regular time intervals.

There is no compression, so the contents are a close-to-exact representation of the original analogue sound. PCM is the most common audio format used in CDs and DVDs (remember them?). Variations on PCM audio are files with extensions such as RAW and SOU. Since the file contains no information about the playback speed, you have to find out by trial and error. Some SND files are raw files with a short header with details of the playback speed.

WAV

WAV stands for *'Waveform Audio File Format'* and was developed by Microsoft and IBM. Most WAV files are also PCM stored but contain a short file header.

This is the most common audio file type, along with MP3 files.

AIFF

AIFF stands for *'Audio Interchange File Format'* and was developed by Apple but they are also playable on Windows computers.

Most AIFF files are uncompressed PCM, although there is a version called AIFF-C which compresses files up to a 6:1 ratio, with some loss of sound quality.

Lossy compression

MP3

This is by far the most popular way of storing audio files in a compressed format.

It is pretty much universal and can be used in PCs, Macs, Androids, iPhones, Smart TVs, and - of course - MP3 players.

It has been used by radio stations for high-quality links to other sites, without incurring the much higher costs of renting noise-free dedicated broadcast links. The encoding and decoding is carried out in this instance by studio equipment.

It reduces the file size by:

- Ignoring all sounds that are likely to be beyond normal hearing.
- Reducing the quality of sounds that aren't easy to hear.
- Compressing all other audio data.

MPEG audio compression uses '*perceptual audio coding*' to reduce file sizes. The amplitude and harmonic relationship between frequencies in the sample, along with the set of frequencies present in the sample at any one moment results in parts of the signal not being heard by the human ear. If a particular frequency is very loud at one point, the ear does not distinguish any nearby frequencies of lower amplitude. These lower frequencies can be filtered out without any loss of perceived quality. . Since the signal contains less detail, it occupies less space. A typical audio track may occupy around 50MB when stored as a WAV file and just over 4MB when stored as an MP3 file.

The choice of compression ratio will depend upon the type of audio source being recorded.

Typical compression ratios are:

Sound Quality	Typical Compression ratio	Bit Rate	Bandwidth
CD Quality	Up to 12:1	128kbps or greater	20kHz or greater
Almost CD Quality	12:1 to 16:1	96kbps to 128kbps	15kHz or greater
FM Radio Quality	24:1	64kbps	11kHz
AM Radio Quality	48:1	16kbps	4.5kHz
Telephone Quality	96:1	8kbps	2.5kHz

Even at compression ratios of 24:1 and above, the finished product is still of much better quality than trying to reduce file size by reducing the sampling rate or the dynamic range.

AAC

AAC stands for Advanced Audio Coding. Its compression method is superior to MP3, so produces a higher quality sound for the same bit rate as MP3.

AAC is used by YouTube, Android, iTunes, iPhone, iPad, some Nintendos and some PlayStations.

WMA

Windows Media Audio is the compressed format developed by Microsoft to compete with MP3. It uses the WMA extension (e.g. "*songbird.wma*") and produces files that are smaller than MP3 files at the same quality.

The format supports sampling rates of 8kHz to 48kHz, mono or stereo, 8-bit or 16-bit, with bit rates between 5kbps and 192kbps.

Although less popular, its compression method is better than with MP3.

Other less well used lossy formats include OGG (Open Source), MPC (Musepack), VOC (Creative Labs), ATRAC (Sony), RM (Real Networks) and WMA lossy.

Lossless compression

Lossless compression is able to reduce the file size without losing any audio information in the file. The original uncompressed data can be re-created from the compressed version. The price for this is a larger file size than lossy compressed versions - up to 2x to 5x larger for the same source file.

FLAC

The Free Lossless Audio Codec compresses an audio source file by up to 60% or more without any loss of the original audio content. FLAC is open-source and royalty-free. It is viewed as the main alternative to MP3 for music.

m4a

This is the file extension for the Apple Lossless Audio Codec (ALAC). ALAC was developed by Apple and it eventually became open source and royalty-free. Although ALAC is slightly poorer at compression than FLAC, it is supported by iTunes and iOS.

Less well used lossless formats include WMA Lossless (Microsoft), APE (Monkey's Audio), WV (WavPack - open source), TTA (True Audio), ATRAC Advanced Lossless (Sony) and MPEG-4 SLS.

Inside an audio file

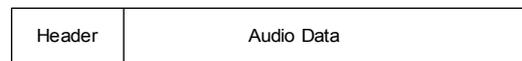
All audio file types, apart from RAW files, are called '*self-describing*' and comprise two information sections.

Header

Also known as the '*wrapper*' or '*Format Chunk*'.

The header information of different formats vary but generally contain:

- Name of the clip
- Clip size
- Duration of audio
- Number of channels (e.g. mono/stereo),
- Sample size (in bits)
- Sampling rate (in kHz)
- Type of compression used



The header of a WAV, for instance, provides the option for the inclusion of a whole range of additional information, such as:

- Artist the name of the artist appearing on the clip
- Source the source of the recording
- Genre the category of the sound (e.g. rock or classical)
- Medium the medium on which the original recording was made
- Comments general comments on the sound clip

Data

Also known as the '*Data Chunk*', this stores the audio information in binary format. This is the actual audio data stream that is used to reconstruct the original sound.

As mentioned earlier, RAW files contain no header information.

File conversion

There are many applications that convert files into the more commonly used formats such as MP3 and WAV.

You can download '*audio-converter-any-format*' which is free from the Microsoft Store. It can convert between .wav, .m4a, .acc, .mp3, .flac, .aiff, .wma, .ac3, .amr, .ogg, .opus, .alaw, .g722, .ulaw, and .aac.

There is a tutorial on its use at

www.cidadeapps.com/audio-converter-any-format-tutorial/

although most converters are pretty intuitive.

There are also many online converters, if you prefer not to install extra software.

A good example is at:

<https://audio.online-convert.com/>

The screenshot shows the 'Online-Convert' website interface. At the top, there is a navigation bar with 'Home', 'File formats', 'Blog', and 'Developers'. The main header features the 'ONLINE-CONVERT.COM' logo and the tagline 'Convert media free, fast and online. No software installation needed.' Below the navigation bar, a sidebar on the left lists various conversion categories: Converter, Archive converter, Audio converter (with sub-options like AAC, AIFF, FLAC, M4A, M4R, MMF, MP3, OGG, OPUS, WAV, WMA), Device Converter, Document converter, Ebook converter, Hash encryption, Image converter, Software Converter, Video converter, and Webservice converter. The main content area is titled 'Convert audio to MP3' and includes an 'Online music converter' section with instructions: 'Convert your audio file to MP3 in high quality with this free online MP3 converter. Just upload your music and download the MP3 within an instant. You can also upload video files and extract the audio track to MP3.' A large green dashed box contains a cloud upload icon and the text 'Drop Files here', with a 'Choose Files' button and options for 'Enter URL', 'Dropbox', and 'Google Drive'. Below this, a file 'Track 17.WMA' is shown with a size of '8.74 MB' and a 'Delete' button. A green 'Start conversion' button is prominently displayed. Underneath, there is an 'Optional settings' section with dropdown menus for 'Change audio bitrate' (set to 'Constant Bitrate'), 'Change sampling rate' (set to 'no change'), and 'Change audio channels' (set to 'no change'). There is also a 'Trim audio' section with input fields for start and end times, currently showing '00:00:00'. Checkboxes for 'Normalize audio' and 'Remove cover art' are present. At the bottom, a 'Save settings' section includes a 'Save settings as:' label, an input field for a name, and a '(Log in to activate)' link.