GUIDE TO SOUND

How to get the best results from your hi-fi system.

The Amplifier

VOLUME CONTROL

The volume control is like your cars' accelerator. The accelerator controls the fuel that determines the speed of the car. On your amplifier, the volume control increases or decreases the current which in turn alters the sound output.

POWER

High power really does contribute to sound quality. Take the car comparison. Imagine driving a VW and then a Porsche at 30 mph. Putting your foot hard on the accelerator in each will give an entirely different reaction because of the power reserve. It is the same for an amplifier. Sound going through an amplifier consists of many peaks and pulses. High power means greater freedom of motion for these and thus improved tonal quality.

BALANCE CONTROL

This is like the steering wheel on a car. If the wheel is not balanced, you become disorientated. It is the same for your speakers. To make sure that your speakers are even on both sides:

- If fitted, push the mono switch on the amplifier.
- Adjust the balance until the sound image is dead center of the speakers (try closing your eyes and have a friend adjust the distance between the speakers until this sound image is really strong).
- Change to stereo again and you have nicely balanced sound.

If this strong central image occurs when the balance control is not in the "0" position, do not worry. Room acoustics may affect the position of the central image.

LOUDNESS SWITCH

This is similar to the choke on your car. The loudness switch is used when conditions are below average. At very, very low listening levels, high and low frequencies are not heard as well as when the volume level is high. So the loudness control compensates for your ears when the volume is low. It boosts the treble and bass at low levels. But, as for the choke, do not use it under normal conditions.

The Amplifier

TONE CONTROLS

Those are very similar to the heating and ventilating system on your car. When all is normal, you do not need them. But when it is too hot or cold, your comfort depends upon them. With today's high standards of recording techniques, you will find that most of your records are perfectly balanced for tone. But the environment you are in may affect the bass and treble response so you may add either or both to compensate. Use of tone controls also depends on your own taste, listening and acoustic conditions.

CHECK YOUR SYSTEM

Check these items before playing your equipment for the first time.

- All tone controls should be in the flat or "0" position. This will provide the best response when you calibrate the acoustic characteristics of your speaker system.
- All filters should be in the OUT or OFF position.
- Set the selector switch to your preferred source and make sure that the component output connection is in the correct digital or analog left and right inputs.
- After following this procedure, you can then adjust the other controls according to your taste.

SPEAKERS

Certain speaker positions may produce coloration of sound owing to sound reflections and resonance of materials in your room. To eliminate these mount the speaker off the floor, preferably on specially designed loudspeaker stands.

Remember to leave a space of 30 to 50 cm's between the speakers and the wall.

"Live" or "Dead" rooms.

Stereo sound depends on room acoustics. Some rooms are "live" in that they echo sound, others are "dead" and sounds can be muffled. Check the acoustics of the room by clapping your hands and listening to the result.
A loud report with an echo indicates that your room is live.
A muffled sound that disappears shows the room is acoustically dead.

If the room is too live, add curtains, carpets, sofas and other sound deadening elements to bring the echo down. If your room is dead, do the reverse and try to add reflecting surfaces such as glass fronted pictures, lamps and glossy wallpapers etc...

The Network Player

Network players are designed to ‘stream’ music from the internet or from other computers or storage devices in your home, via a connection to your network, usually via your wireless broadband router. These could be files that you have either downloaded from an internet service such as iTunes or files that you have ‘ripped’ to your computer from your own CD’s, and if connected to the internet, radio stations can be streamed and so can music services such as **Qobuz** and **Spotify**.

As their name suggests, network players are connected to your home network via your router, either by Ethernet cable or via Wi-Fi. The higher the quality/size of the music file you wish to listen to, the more ‘bandwidth’ is required to play the file back via your network player. Therefore connection via a network Ethernet cable is preferable as opposed to Wi-Fi.

DIGITAL MUSIC

Before the CD was invented the music we enjoyed at home was all analog based. The original music was recorded in a studio using analog multi-track tape, mixed to a stereo analog tape and then mass produced as an analog vinyl record or audio compact cassette tape. Professional digital recording began in the late 1970’s but it wasn’t until the advent of the compact disc (CD) in the early 1980’s that digital music became a mass consumer medium. Digital recording is achieved by converting an analog audio signal into binary code. Binary code is a numerical value made up of a series of on’s and off’s (or signal and no-signal) which are usually represented as ones and zeros. Each of these ones and zeros is referred to as a bit, and these bits form digital sounds. Bit depth refers to the length of each digital sound. Just to be clear, mass produced compact discs use a digital sample rate of 16bit 44.1KHz. (Bit Depth and Sample Rates are explained in detail below.)

Digital music files vary enormously in both size and quality. It is usual to assume that the smaller the file size, the lower the quality. Many popular music file formats such as MP3 are very heavily compressed and are not true copies of the original recording. In effect, much of the original detail from the original CD audio is lost forever once converted to a compressed file type such as MP3 or WMA.

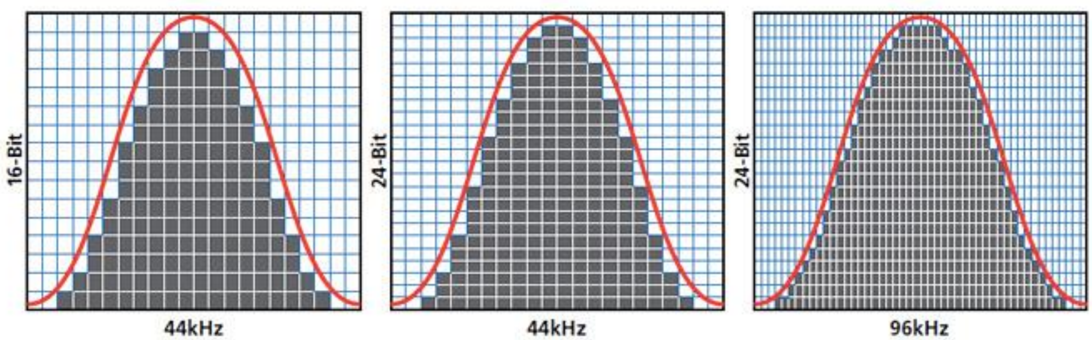
DIGITAL AUDIO SIGNALS

SAMPLE RATES
(Samples Per Second or Frequency)

Sound is made up from pressure waves. A single constant wave has its frequency measured in Hz (oscillations per second). Humans can hear from a lowest frequency of 10's of Hz, up-to higher frequencies just below 20,000 Hz, or 20 KHz.
Sample rate is the number of samples of audio carried per second, measured in Hz or KHz (one KHz being 1 000 Hz). For example, 44,100 samples per second can be expressed as either 44,100 Hz, or 44.1 kHz. Bandwidth is the difference between the highest and lowest frequencies carried in an audio stream. The sample rate of playback or recording determines the maximum audio frequency that can be reproduced.

BIT DEPTH
(and Amplitude)

Bit depth is the number of bits of information in each sample, and it directly corresponds to the resolution of each sample. Examples of bit depth include Compact Disc Digital Audio, which uses 16 bits per sample, and High Resolution Audio which can support up to 24 bits per sample.



As Bit depth and sample rate increase, more information is captured, this resulting in higher quality audio.

When creating a digital music file by 'ripping' a CD, the maximum quality possible, (if a lossless copy is made), is 16Bit 44.1KHz. So you may well ask, if CD quality has been perfectly good and fully accepted as the best possible quality for the last 40 plus years, is there a need for 24Bit 96 KHz or 24Bit 192 KHz files? Is it a marketing ploy? Well, yes and no. Yes it is a ploy in that 'more' is not necessarily 'better'. No, as it is possible to acquire 24Bit 'studio master' quality music files from an increasing number of high-end music download services. However, it is arguable that in order to hear a real difference your hi-fi system would have to be very high-end to match, with a pretty expensive amplifier and equivalent speakers. However, there is no doubt that the re-mastering process that older recordings must go through to become a 24bit 'studio quality' does make a huge difference to the overall sound quality. It is possible to then make lower bitrate copies and hear no difference in quality.

DIGITAL AUDIO SIGNALS

Consider these 3 images as representations of bit depth:

8 bit

16 bit

24 bit

8 bit has the worst detail, it looks coarse, for audio it sounds coarse, but there is not too much difference between 16 bit and 24 bit, they are both reaching the limits of perception. Audio CDs are 16 bit, whereas SACDs and HD Audio are 24 bit. Again you may ask, is it a marketing ploy?

Well yes and no.

Yes, as most people cannot hear the difference between the two.
No, as many modern 16 bit audio CDs have been volume compressed meaning the quiet parts are made louder, so that when played on the radio or TV the overall track sounds louder. (a 1980's CD would sound quieter and more dynamic in comparison to one from 2015). This means the overall quality of the music on the modern CD release is actually sub-standard.

Also, the fact that the record companies are now fully behind making all new recordings available in 24bit ‘studio master’ quality and also that the back catalogues of the major labels are slowly becoming available as 24bit ‘re-masters’ will benefit all of us music consumers as the re-mastering process will breathe new life into many older recordings and you will hear a noticeable improvement.

DIGITAL AUDIO SIGNALS

COMPRESSION

When talking audio, compression can have two meanings: volume compression where the volume levels are 'compressed' to make the overall piece appear louder and digital audio compression, used to reduce the file size. We are discussing digital audio compression, of which there are two types: **LOSSY** and **LOSSLESS**

LOSSY

The majority of compressed audio files are lossy, IE: when encoding, audio quality is sacrificed to achieve higher rates of compression. How much quality is lost depends on the encoder and settings used for compression. Bit rate plays the biggest role in determining final quality. Higher bit rate files have better quality than lower bit rate files. Bit rate is normally presented in Kbps (Kilo-bits-per-second). When compressing to MP3 files in is possible to adjust the bit rate.

Typically these are:

- 32 kbit/s - generally acceptable only for speech
- 96 kbit/s - generally used for speech or low-quality streaming
- 128 or 160 kbit/s – low-to-standard bitrate quality; difference can sometimes be obvious.
- 192 or 256 kbit/s - a commonly used high-quality bitrate
- 320 kbit/s - highest level supported by MP3 standard

Other lossy formats include:

- **AAC** (Advanced Audio Coding) is a standardized, lossy compression and encoding scheme for digital audio. Designed to be the successor of the MP3 format, AAC generally achieves better sound quality than MP3 at similar bit rates.
- **Ogg Vorbis**. Vorbis is a free software / open source project headed by the Xiph.Org Foundation (formerly Xiphophorus company). The project produces an audio format specification and codec for lossy audio compression. Vorbis is most commonly used in conjunction with the Ogg container format and it is therefore often referred to as Ogg Vorbis.
- **WMA** Windows Media Audio is an audio data compression technology developed by Microsoft. It is a proprietary technology that forms part of the Windows Media framework. The original WMA codec, known simply as WMA, was conceived as a competitor to the popular MP3 and RealAudio codecs.

DIGITAL AUDIO SIGNALS

LOSSLESS

Lossless data compression is a class of data compression algorithms that allows the exact original data to be reconstructed from the compressed data during playback. Audio which is compressed using lossless can be uncompressed exactly the same (bit for bit) as the source file. It is without loss. Lossless is slowly gaining ground on Lossy, the main advantage being once your CD collection is ripped into a lossless format that's it, no more re-ripping, unlike lossy where the need to re-rip might present itself if a newer encoder is released. Lossless can be converted to any other Lossless format without degradation. Lossless can be converted to any Lossy format and has the same quality as though ripping from audio CD.
Lossless audio formats include:

- **FLAC** (Free Lossless Audio Codec) Digital audio compressed by FLAC's algorithm can typically be reduced to 50–60% of its original size, and decompressed into an identical copy of the original audio data. FLAC is an open format with royalty-free licensing and a reference implementation which is free software. FLAC has support for metadata tagging, album cover art, and fast seeking.
- **WMA Lossless** – Windows Media Lossless is a lossless audio codec developed by Microsoft which compresses audio signals without loss of quality from the original using VBR (variable bit rate). When decompressed, the audio signal is an exact replica of the original. The first version of the codec, WMA 9 Lossless, and its revisions support up to 24-bit 96 kHz audio.
- **ALAC** (Apple Lossless Audio Codec) Apple Lossless is an audio codec developed by Apple Inc. for lossless data compression of digital music. Lossless data is stored within an MP4 container with the filename extension .m4a. ALAC is the only lossless format supported in Apple's iTunes software or on iOS devices, so users of iTunes software who want to use a lossless have to use ALAC.

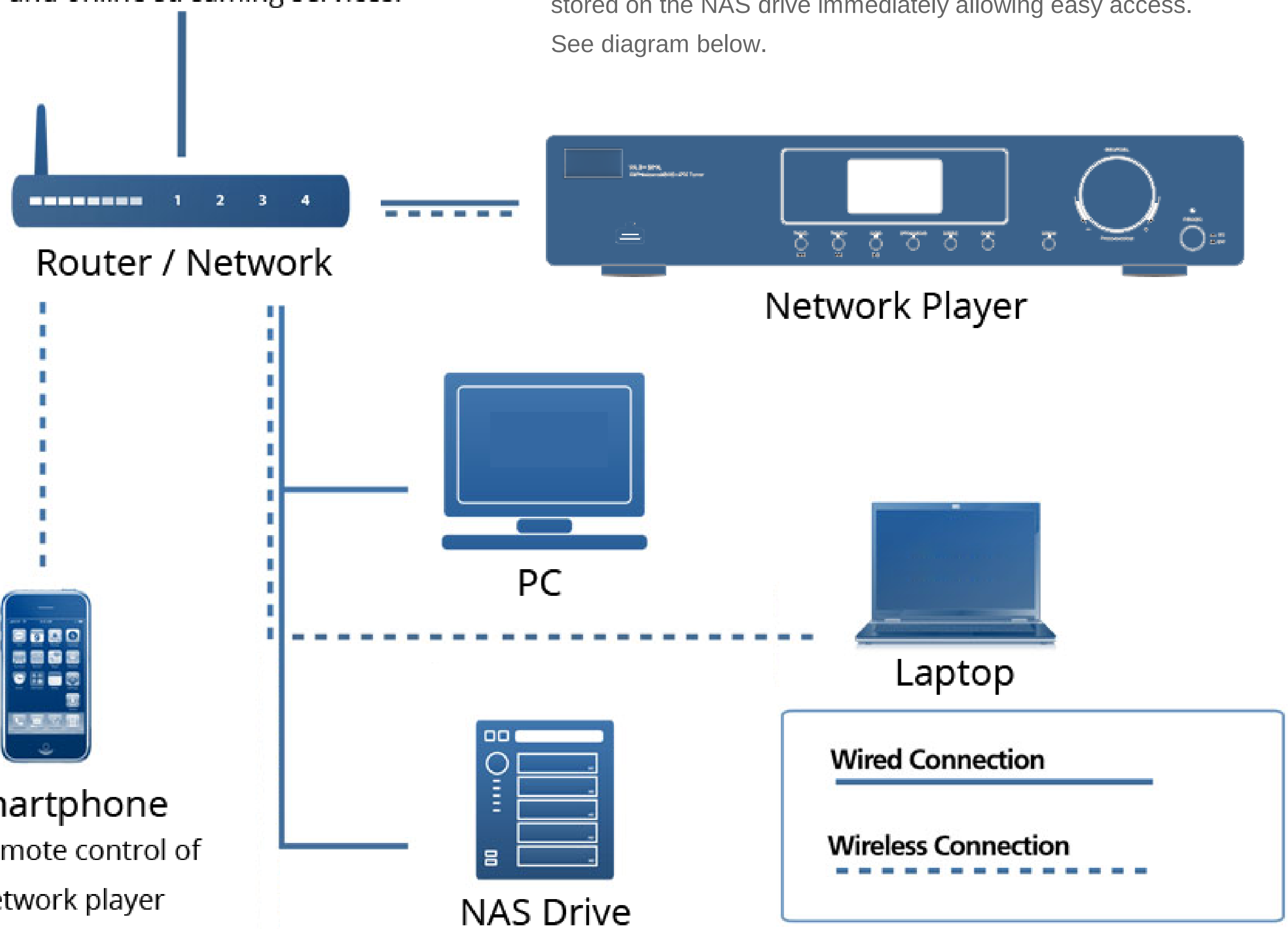
UNCOMPRESSED

Waveform Audio File Format (**WAVE**, or more commonly known as **WAV** due to its filename extension) is a Microsoft and IBM audio file format standard for storing audio on computers. It is the main format used for raw uncompressed audio. The usual bit stream encoding is the linear pulse-code modulation (LPCM) format, the standard audio file format for CDs. WAV audio can also be edited and manipulated with relative ease using software. Uncompressed WAV files are large, so file sharing of WAV files over the Internet is uncommon. Apple have their own uncompressed format call **AIFF**. Unlike WAV files, AIFF can support metadata so artist, song and album information can be stored in the file and read by programs such as iTunes.

NETWORK FILE STORAGE

NETWORK ATTACHED STORAGE

Network Attached Storage (NAS) systems are becoming more and more popular and provide the best basis for storing and accessing music files in a network audio environment. A NAS drive usually contains one or more hard drives, often arranged into logical storage containers or RAID arrays. (A RAID array uses two drives that can be set to behave as one, one drive being a mirror or back up of the other). A NAS drive can be connected directly to your router via USB or via an Ethernet cable and can then be seen by your network player without the need for your PC or laptop to be switched on. You can also make the NAS drive the default location for you music files, meaning when you download music or rip a CD the files can be stored on the NAS drive immediately allowing easy access. See diagram below.



The CD Player

COMPONENTS

A CD player has three major mechanical components: a drive motor, a lens system, and a tracking mechanism. The drive motor (also called spindle) rotates the disc between 200 and 500 revolutions per minute. The tracking mechanism moves the lens system along the spiral tracks in which information is encoded, and the lens reads the information using a laser beam, typically produced by a laser diode. The laser reads information by focusing a beam on the CD, which is reflected back to a sensor. The sensor detects changes in the beam, and interprets these changes to read the data. This data is processed, and eventually converted to sound using a digital-to-analog converter (DAC).

FEATURES

CD players can employ a number of ways to improve performance, or reduce component count or price. Features such as oversampling, one bit dacs, dual dacs, interpolation, anti-skip memory, digital and optical outputs are likely to be found. Other features improve functionality, such as random play and repeat, or direct track access. Yet others are related to the CD player's intended target, such as anti-skip for car and portable CD players, pitch control and queuing for a DJ's CD player, remote and system integration for household players. Description of some features follows:

- Oversampling is a way to improve the performance of the low pass filter present at the output of most CD players. By using a higher sampling frequency, a multiple of the 44.1khz used by CD encoding, it can employ a filter with much lower requirements.
- Dual DACs were sometimes advertised as a feature because some of the early CD players used a single DAC, and switched it between channels. This required additional supporting circuits, possibly degrading sound quality.
- Interpolation, while not usually advertised, is present in most recent CD players and is a method used to correct errors that may be present on a CD’s surface, perhaps due to dust, scratches or dirt.

POSITIONING AND CONNECTING

In the case of integrated CD players, vibration control is important and your player will benefit greatly from being mounted on a stable, resonance-free surface. You may be surprised at the improvement in sound resulting from the player being securely held down. Also, in the case of 75-ohm coaxial connections, impedance mismatching caused by the use of improper cables can cause reflections within the signal which can degrade the data stream into an inaccurate semblance of the original. So always make sure to invest in a good quality digital connector cable whether optical or coaxial.

The Speakers

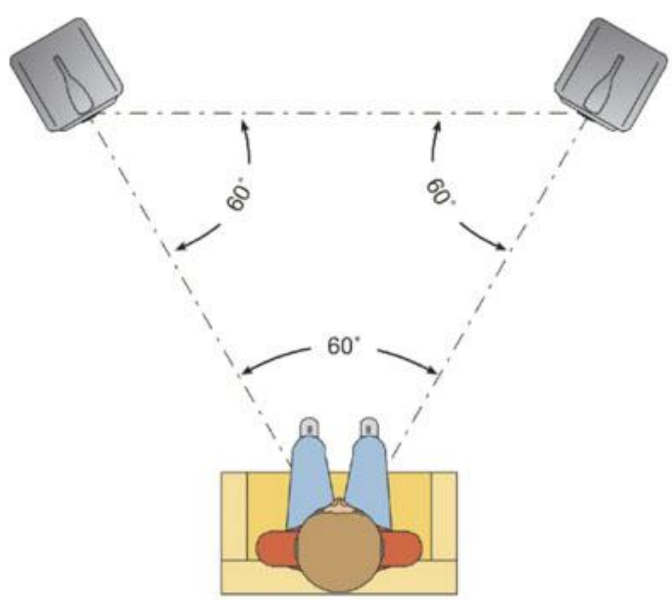
POSITIONING AND PLACEMENT

In its own way, speaker placement and positioning is just as critical as the setup of all your other components. In order for speakers to properly produce the correct tonal balance, image and sound stage, and "disappear," leaving the illusion of music filling the room from a three-dimensional soundscape, they must be placed accurately. The ear must be the final arbiter in getting them to resemble live music as closely as possible, but the techniques for getting there are well-known and time-tested. There are five basic parameters with which to be concerned:

- Distance of the speakers from each other.
- Distance from rear walls.
- Distance from side walls.
- Degree of toe-in (the angle which the speakers are aimed at the listening position)
- Height of the speakers

Begin by deciding where in your room your primary listening position will be and then place the speakers in the left and rights corners facing you. Start with the speakers at least two feet away from the rear wall if possible and approximately the same distance from the side walls.

Next, determine how close or how far apart the speakers should be from one and other. If 'D' represents the distance between the speakers, your listening position should be at least the same as 'D' or better still approximately 'D' x 1.5 or as close to this as possible



mitchell & johnson

The Speakers

Select a well-recorded piece of music such as the Mitchell & Johnson demo tracks available from www.mitchellandjohnson.com and begin playback.

If the speakers are too far apart, you'll hear a "hole in the middle" effect, where the sound seems to come from both speakers, with a noticeable lack of sound between them. If they are too close together, the sound will be bunched up in the middle, depth and imaging will be lacking, and the sound will not appear to extend outside the speakers' edges.

With the speakers facing straight out move them closer together or further apart until you hear a continuous, even sound-field. At this point, you'll start to hear depth and imaging. Next, determine the distance from the rear walls. Too close, and sound will suffer as mentioned above. Experiment until you get the best combination of image solidity, three dimensional soundstage, and tonal balance.

Now we're ready to adjust the degree of toe-in. With dynamic speakers, some degree of toe-in is usually called for. A good rule of thumb is to aim the speakers toward the listening position so that when viewed, the inside edges of the speakers are just at the point where they can't be seen. Again, experiment until you get the best combination of image focus, centre fill, and soundstage. When you think you've located the optimum position, mark the locations of the speakers.