Naim ND 555

Network Audio Player
Design, Engineering and Technology

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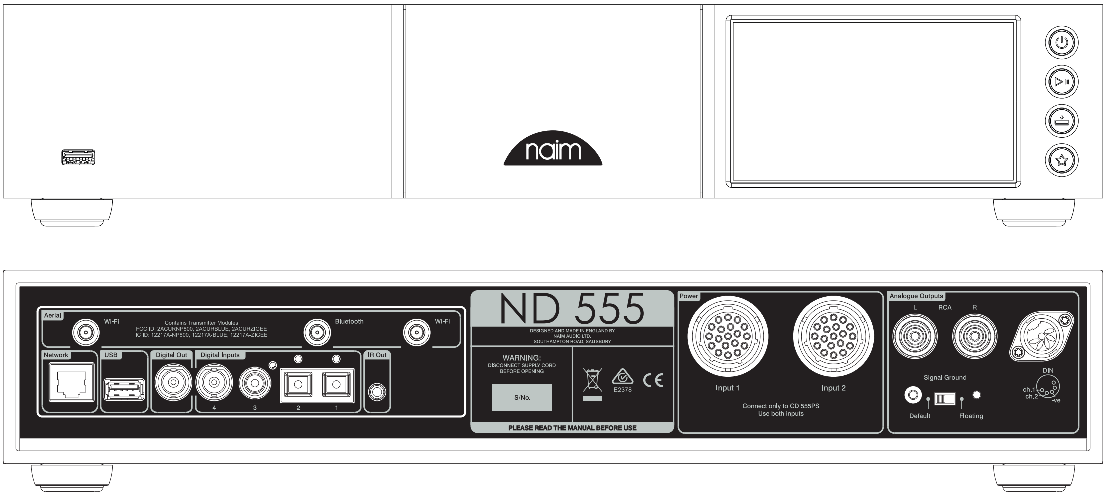
Naim Audio

Overview



ND 555 is the first network player from Naim Audio to earn 500 series status. It is, in short, the very best network audio player that Naim knows how to make. As such, it's a typical piece of holistic, evolutionary Naim engineering: it builds on past developments, it pays as much attention to mechanical design as electrical design (particularly in combating the effects of vibration), and it takes trouble over important minutiae that other brands routinely ignore. It is not festooned with snappily-named 'revolutionary' technologies; rather, although radical in traditional Naim fashion, it is a considered, thoroughgoing and painstaking piece of engineering – one for which the final arbiter in all design decisions has been sound quality.

Design highlights

- Ground-up new design
- High-performance Naim-exclusive NP800 streaming card provides unique clock master capability
- Latest IEEE 802.11ac Wi-Fi
- Enhanced 'stream catcher' buffer enables whole track playback from local memory
- Support for 384kHz PCM and double-rate DSD (DSD128)
- Bluetooth connectivity using the latest aptX HD codec
- Chromecast Built-in support
- Zigbee radio frequency remote control
- Software update via network
- LVDS
- 13 internal Naim DR discrete voltage regulators
- Requires 555 PS DR power supply (as this optimises internal power and ground wiring)
- MIPI-controlled display reduces power requirement and minimises electromagnetic radiation

Digital board

Physically and functionally, the circuit of the ND 555 – shown as a block diagram in Figure 1 – is divided between three principal functional sections housed on a total of 10 printed circuit boards (PCBs): the digital card, the digital-to-analogue converter (DAC), and the output filter. Each is considered in turn in the following description.

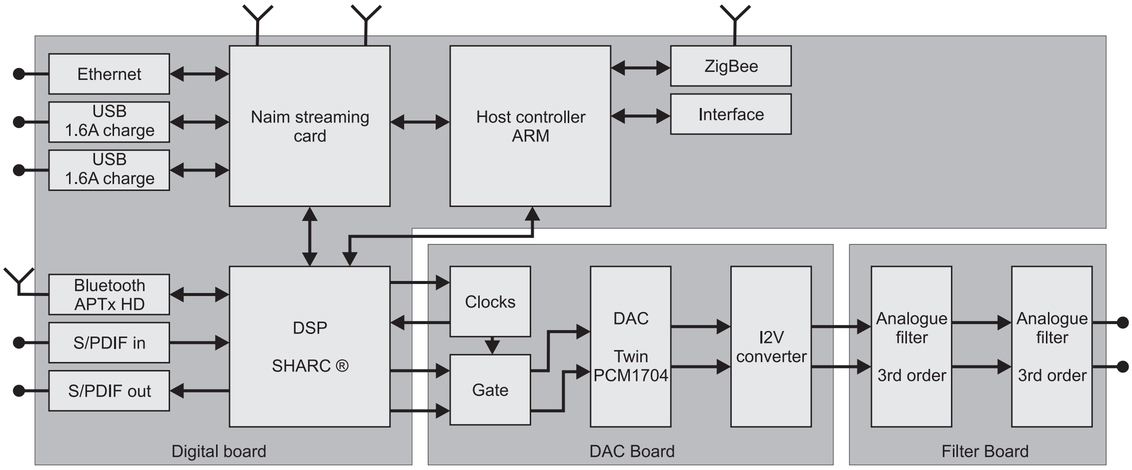


Figure 1. Block diagram of the ND 555's circuit. Signal processing is divided between three functionally and physically separate sections

Network connection, digital input/output, remote control and DSP (digital signal processing) are all provided by a six-layer digital board which carries a new, much higher performance, six-layer streaming card – the NP800 – which was co-designed by and is exclusive to Naim.

It offers three key features not found elsewhere:

1) It allows ND 555 to be the clock master, determining the rate at which audio data is streamed in. The clock controlling this is located close to the DAC chips, where it must be to ensure low jitter.
2) It uses LVDS (low-voltage differential signalling) to convey audio data, which reduces radiation of RF interference (LVDS is described in more detail in the box-out on page 4).

The only holes in the box are small cut-outs where wire connections pass through, making it more effective at containing high radio frequencies than a perforated screen.

Aluminium is used, as it is for the main casework, to prevent 'magnetic distortion' arising through the interaction of signal-related magnetic fields with ferromagnetic enclosure materials.

The left block diagram in Figure 2 illustrates the method of clocking Ethernet data used in the previous-generation NDX/NDS. A fixed clock on the streaming board controls the flow of data from the network. To accommodate small differences in frequency between this clock and the master clock

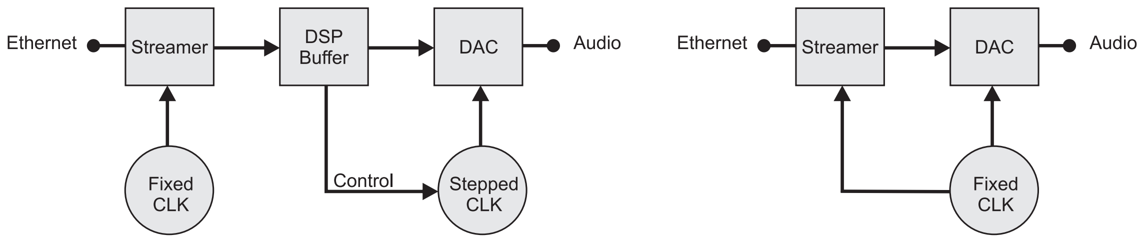


Figure 2. Old (left) and new (right) methods of controlling data flow from the network

3) It incorporates sufficient RAM to buffer almost five minutes of 44.1kHz/16-bit audio.

Suppression of RF interference is further enhanced by the digital board being enclosed within a six-sided aluminium screening box within the main enclosure, which acts as a Faraday cage.

controlling the DAC chips, Naim's RAM buffer de-jitter method keeps the average frequency of the two clocks the same. (This method is described in more detail below for the handling of S/PDIF input.) In the new arrangement (right block diagram in Figure 2) the DAC master clock is sent using LVDS to the NP800 streaming card to control data flow. Now the fixed DAC clock is master for the whole process, directly ensuring very low jitter.

Digital board

Low Voltage Differential Signalling

Digital audio and clock data is sent, separately, from the digital board to the DAC board using the I²S (aka I2S or IIS) serial bus interface developed by Philips for communicating PCM audio data between integrated circuits (I²S is short for Inter-IC Sound). Usually the I²S bus is run single-ended, with the digital signal carried on one conductor and referenced to ground (Figure 3).

ND555 uses instead a superior method called LVDS (low-voltage differential signalling) in which lower signal voltages are carried differentially (ie with opposite polarity) on two conductors (Figure 4). This results in lower electromagnetic radiation (a) because of the reduced signal voltage and (b) because of electromagnetic coupling between the two conductors which substantially cancels their radiated fields. LVDS is also used to convey digital audio to and from the streamer card.

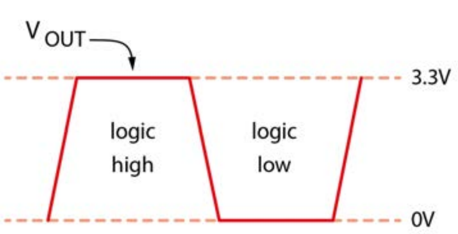


Figure 3. Conventional single-ended digital signal

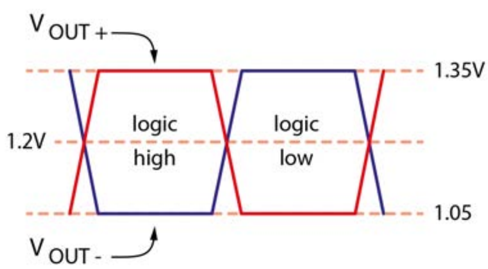


Figure 4. LVDS balanced digital signal

The NP800 card, operated by a 1GHz ARM Cortex A8 processor, supports faster data streaming rates to exploit gigabit networks.

Wi-Fi connectivity is dual-band (2.4GHz or 5GHz) and uses the latest IEEE 802.11ac protocol, a ‘supercharged’ version of 802.11n. Theoretically capable of maximum data throughput of almost 7Gbps (as opposed to 600Mbps for 802.11n), it allows the ND 555 to acquire audio data very much faster than real-time via either wired Ethernet or Wi-Fi, buffer it and play it out to the DAC stage from memory.

Memory on the card has been increased to 512MB of DDR3 RAM (16 times the 32MB on Naim’s previous streaming card), 50MB of which is allocated to the audio buffer allowing the NP800 to store almost 5 minutes of a 44.1kHz/16-bit stereo audio signal.

Streaming from a network, ND 555 is compatible with PCM sampling rates of up to 384kHz and with double-rate DSD (DSD128, sampling rate 5.6448MHz). Because the ND 555’s PCM1704 DAC chips are not DSD compatible, DSD signals are transcoded to PCM within the DSP. First the DSD signal is downsampled to 352.8kHz, 40-bit floating point PCM and low-pass filtered to remove DSD’s noise-shaped ultrasonic quantisation noise. Then it is upsampled to 705.6kHz/24-bit for passing to the DAC stage.

Bluetooth connectivity is provided and supports aptX HD, the latest Qualcomm audio codec, which is compatible with LPCM signal data up to 48kHz/24-bit and is backwards-compatible with previous aptX codecs.

For highest signal quality a wired Ethernet connection is recommended; when not in use, both the Wi-Fi and Bluetooth interfaces are disabled to reduce radio frequency (RF) interference.

Digital board

While the ND 555 is principally intended for streaming audio from a local area network (LAN) or wireless local area network (WLAN), the NP800 also provides for connection of local digital audio sources via coaxial or optical S/PDIF. It also has a USB Type-A socket for the connection of a flash drive (USB memory stick) or USB-connected external hard disk drive containing PCM or DSD audio files.

Electrical S/PDIF input is available via either BNC or phono socket, while two Toslink sockets provide for optical connection. The coaxial S/PDIF input, like the Ethernet input, is galvanically isolated to prevent connected equipment from injecting ground noise. Maximum supported sample rates are 192kHz for the electrical inputs and 96kHz for the optical inputs. A single S/PDIF output, on BNC, is provided for connection to an AV system and can be disabled via the control app, which benefits sound quality.

No DAB/FM option is offered on the ND 555 as internet radio provides a more diverse alternative, independent of the different terrestrial transmitter networks available worldwide.

Whereas the ND 555 operates as clock master for audio signals ‘pulled’ from a network, with an S/PDIF input the signal is clocked out by the source component and that clock has to be recovered and de-jittered by the receiver.
The ND 555 does this using a refinement of the method first developed for the Naim DAC. As the recovered clock frequency will never exactly match that of the ND 555’s own clock oscillators, a means is required to adapt the ND 555’s clock frequency to that of the incoming data. This is achieved by sending the data to a FIFO (first in, first out) buffer, from which it is clocked out under control of the ND 555’s own clocks. Disparity in the incoming and outgoing clock frequencies will either cause the buffer to become fuller with time (if the incoming data rate is higher than the outgoing data rate) or less full (if the incoming data rate is lower than the outgoing). The ND 555 monitors the state of the buffer and, over long time intervals, step-adjusts its internal clock frequency in order to keep the buffer roughly half-full.

In the Naim DAC stepping of the clock frequency was achieved by providing 10 selectable fixed clocks, which gave quite large intervals between available clock frequencies. In the ND 555, by contrast, fractional divider clocks are used to provide much finer gradations of clock frequency to allow much closer matching of the incoming and outgoing data rates. In this way the ND 555 is able to adapt its clock frequency to that of an S/PDIF source while completely decoupling its DAC circuits from clock jitter generated either by the source or within the S/PDIF interface.

Naim has traditionally used infra-red (IR) remote control, based on the RC5 instruction set. With the ND 555, Naim is switching to Zigbee radio frequency remote control instead. This provides up to 10m range, removes the need for line-of-sight between the remote handset and the equipment, and has the further advantage of being low-power. The ND 555 handset, pictured, is encased in aluminium.

Because of the need to integrate ND 555 with Naim components using RC5, the digital board features a 3.5mm mini-jack output socket that provides for wired RC5 connection to downstream components. This ensures that the new remote control can continue to provide input selection and volume control on a partnering Naim preamp.

For the first time in a Naim 500 series product, ND 555 provides for software updates to be installed via internet connection.

The ND 555’s Zigbee RF remote control handset

DAC

Digital audio data from the digital board is passed via flexible cable link, carrying the LVDS I²S interface, to the DAC board. Flexibility of all cable connections to the DAC board is essential because it is compliantly isolated from the ND 555 chassis to minimise the effects of external vibration. The same isolation arrangement is used as in the CD555 and NDS: the PCB is bolted rigidly to a thick, 2.6kg brass plate beneath it, which both stiffens the board and provides the necessary mass to achieve a low natural frequency on the six steel coil springs which support the brass plate above the ND 555's chassis. This arrangement forms a mechanical low-pass filter which progressively isolates the board from external vibrations above the suspension natural frequency of 10Hz, ie from below the audible frequency range. No damping is applied to the suspension to ensure maximum isolation.

The DAC chips in the ND 555 are Burr-Brown PCM1704U-K devices. This is the highest specification of PCM1704 but Naim grades them further and uses only the highest-performing in the ND 555. Although it doesn't measure as well as more modern delta-sigma DACs, the legendary PCM1704 is regarded by many – including Naim – as the best-sounding DAC chip ever made. It was listed by Texas Instruments as NRND (not recommended for new design) as long ago as 2004 and about a decade later was discontinued. Naim Audio acquired a stockpile of PCM1704s while still available, which has allowed this DAC chip to be used in the ND 555, perhaps the last commercial product to use this iconic device.

R2R or 'ladder' DACs like the PCM1704 use a resistor network to convert a digitally-encoded signal into an analogue waveform. In principle the operation is very simple; in practice R2R DACs are expensive to manufacture because extremely accurate trimming of the resistors is required to achieve high performance. Modern low-bit delta-sigma DACs, which operate on an entirely different principle, are less costly to make and provide superior measured performance – but R2R conversion is preferred by many audiophiles for its superior sound quality. The difference between R2R and low-bit DACs is described in more detail in Appendix 1.

Although the PCM1704 is described on its datasheet as a 96kHz DAC, it utilises 8× oversampling so the DAC stage can actually operate at up to 768kHz (8 × 96kHz). In the ND 555, oversampling and bespoke digital filtering is performed in an Analogue Devices SHARC ADSP21489 digital signal processing (DSP) chip located on the digital card. This DSP chip has the advantage of low power requirement .

Digital signals at multiples of 44.1kHz sampling rate (44.1kHz, 88.2kHz, etc) are oversampled to (16 × 44.1 =) 705.6kHz, and those at multiples of 48kHz sampling rate (48kHz, 96kHz, etc) to (16 × 48 =) 768kHz.

The use of integer-factor upsampling ('interpolation' in DSP jargon) is mathematically much simpler than non-integer sample rate conversion, which reduces computational load – always desirable as it reduces current draw, improving power supply performance, and reduces RF radiation. Further reduction in RF radiation is achieved in the ND 555 by enclosing each PCM1704 chip within a metal screening can.

Proprietary digital filtering, designed by Naim, was first used in the Naim DAC. Intensive ongoing development at Naim has sought to develop an even better-sounding filter but, despite promising new developments, that end has yet to be achieved. So ND 555 retains the modified Butterworth digital filter, to which additional poles have been added to prevent too much phase shift occurring within the audio band. The filter is implemented using just five lines of assembly code which ensures both low arithmetic noise (fewer additions and multiplications that cause rounding) and low power supply noise (since the DSP draws less current when it isn't calculating). As background, the essentials of oversampling and the role of digital filtering are described in Appendix 2.

Output current from the PCM1704 DAC chips (one per channel) is converted to voltage using Naim-designed discrete-component I2V (current to voltage) stages whose circuit resembles that of a Naim power amplifier in miniature, comprising a differential (long-tailed pair) input stage, voltage amplifier and emitter-follower output stage. Connecting the DAC output to the input stage's inverting input and using shunt feedback from the output operates the stage as a transimpedance amplifier (TIA), where the output voltage tracks input current.

The I2V stages use through-hole components exclusively, raised above the board surface to further enhance isolation from vibration, and to isolate the outer foils of the capacitors from radiated fields from PCB tracks. Kinks in the lead-out wires, applied by a special tool, enhance the vibration isolation thus achieved. Polystyrene capacitors are chosen as they offer the best audio characteristics including low dielectric absorption, low leakage, excellent temperature stability and low distortion, ensuring highest sound quality. As they would be destroyed by flow soldering they are hand-inserted and hand-soldered. The discrete transistors of the I2V stage have their vibration isolation enhanced by being contained within thermal housings, which also stabilize the thermal conditions under which they operate and hence the transistor characteristics.

Power Supplies

Power supply to the DAC board is via nine Naim DR discrete-component voltage regulators located beneath the board, close to the circuit sections they supply. The features and enhanced performance of the DR regulator over three-pin monolithic voltage regulators is summarised in the adjacent box-out.

Naim DR discrete-component voltage regulator

- Adapted Naim power amplifier circuit achieves much faster voltage recovery than the previously used monolithic regulator (Figure 6)
- A subsurface Zener voltage reference, fed from the regulator output, also ensures much lower noise (Figure 5)
- Low current demand from the regulator circuit reduces load on the supply's reservoir capacitors, further enhancing sound quality
- Discrete components, including tantalum capacitors, are used throughout (no ICs)
- As with all Naim equipment, development was conducted as much in the listening room as on the test bench to ensure maximum benefit to musical enjoyment

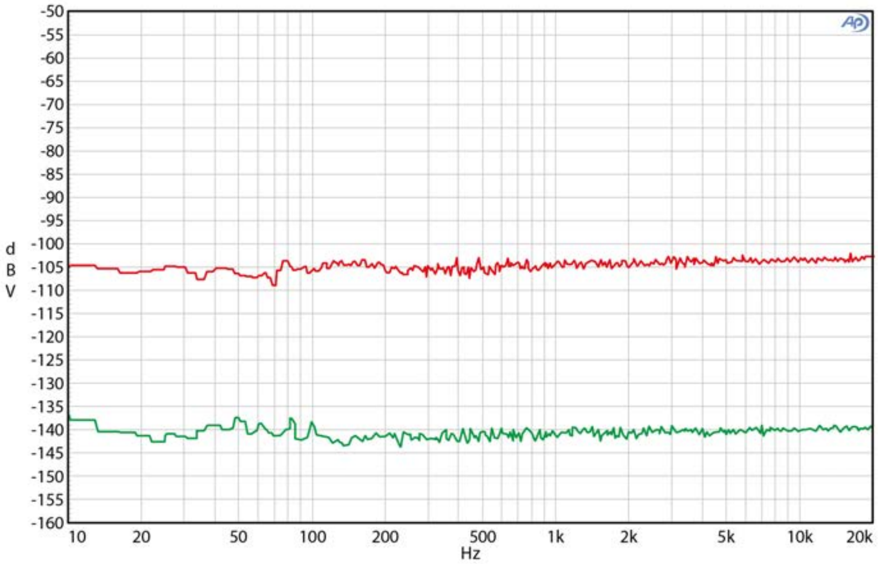


Figure 5. Wideband noise spectra, 20Hz-20kHz, of the outgoing HiCap (green trace) and the latest HiCap fitted with the DR (red trace). The midband improvement is about 15dB (5.6×) with still larger improvements at low and high frequency

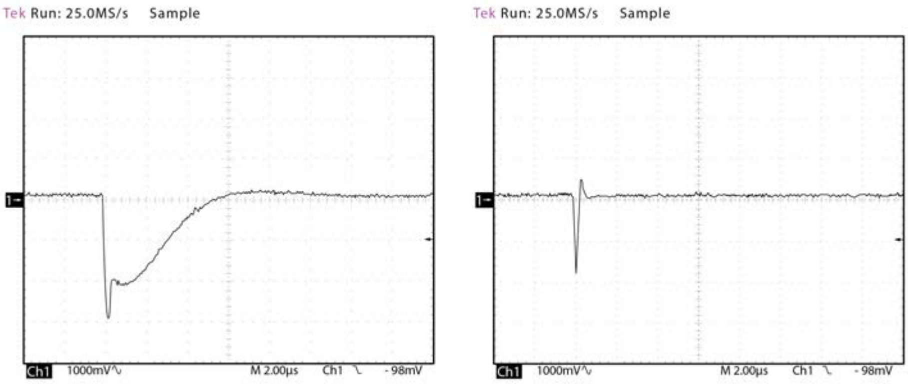


Figure 6. Transient performance comparison of the old regulator circuit (left) and DR (right), showing the change in output voltage when a load current of 0.65A is suddenly applied (rise time 250ns). The much faster settling time of the DR is obvious

Power Supplies

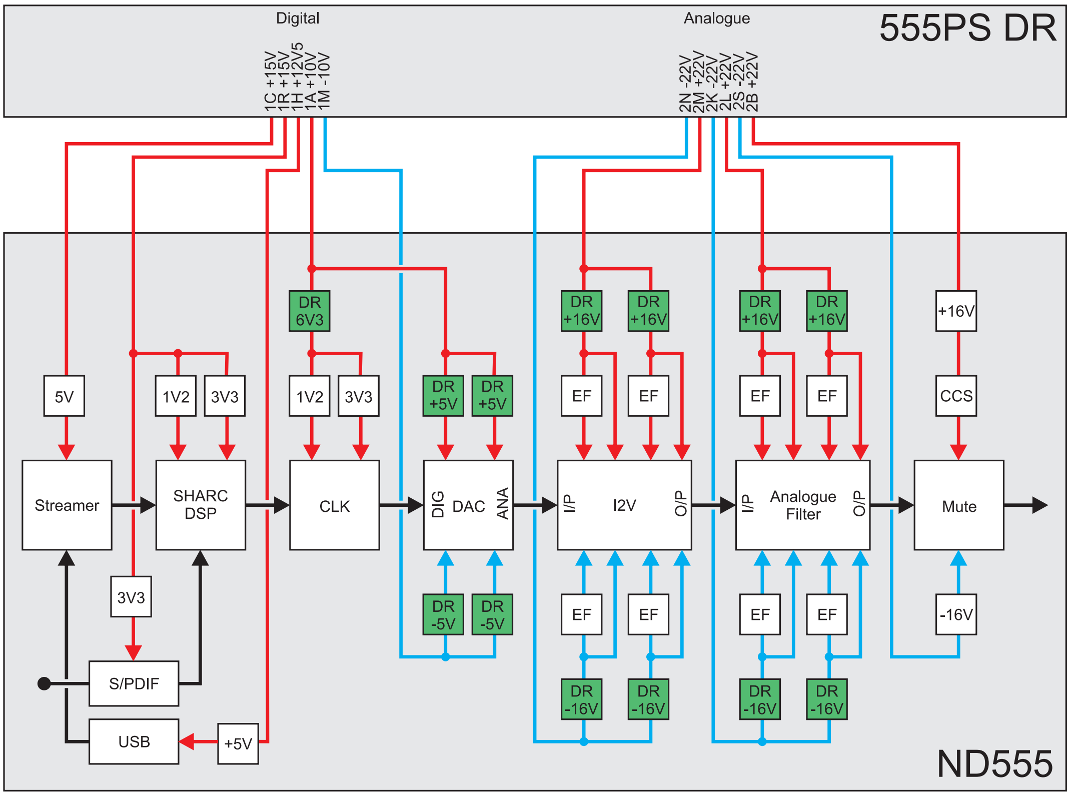


Figure 7. Block diagram of the power supply arrangement for the entire ND 555 showing the complete separation of supplies to the digital and analogue sections. Naim Discrete Regulators are shown in green; downstream of them 1.2V and 3.3V monolithic regulators provide further smoothing for the clock circuits and emitter follower (EF) 'capacitance multipliers' for the I2V and analogue filter stages. CCS = constant current supply for the output relays

Output filter

With oversampling to 705.6 or 768kHz and compatibility with maximum sampling rates of 352.8/384kHz, the lowest frequency at which digital image frequencies can appear in the output of the DAC stage is 529.2kHz. The task of the analogue output filter, immediately downstream of the I2V stage on a separate PCB, is to provide sufficient attenuation by this frequency to reduce the content of image frequencies in the ND 555's output to insignificant levels.

This is achieved using a sixth-order Butterworth filter, realised as two third-order active filters in series, as first used in CD555 and subsequently Naim DAC . Naim discrete-component unity-gain buffer amplifiers form the active elements, within Sallen and Key low-pass feedback filters which have the generalised structure shown in Figure 8. The corner frequency (–3dB) is 32kHz for each filter section (–6dB overall) and the ultimate roll-off rate 36dB per octave (120dB per decade), providing approximately 150dB attenuation at 529.2kHz and still greater attenuation at higher image frequencies. Achieving the desired performance requires the filter's resistors to be closely matched, so they are meticulously graded at Naim prior to use. The output filter is fixed: it remains unchanged whatever the sampling rate of the source signal.

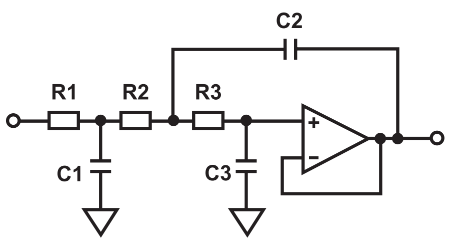


Figure 8. Circuit configuration of a Sallen and Key third-order low-pass filter

As with the DAC board, the output filter PCB is bolted to a brass plate beneath which is compliantly isolated on six coil springs. Again, DR voltage regulators – four in total – placed beneath the board provide low-noise, low-impedance supply voltage to different sections of the circuit, with downstream emitter-follower 'capacitance multiplier' circuits (Figure 9) for further smoothing. Through-hole components are again placed clear of the board surface to further enhance vibration isolation. Polystyrene capacitors within the filter circuits, which would be destroyed by flow soldering, are hand-inserted and hand-soldered.

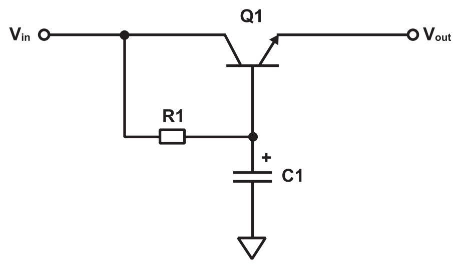


Figure 9. Emitter-follower 'capacitance multipliers' provide further smoothing downstream from the DR voltage regulators for the ND 555's I2V and analogue filter circuits

Attention to detail

Throughout the ND 555 care is taken – in Naim tradition – to dress connecting wires in such a way as to minimise the transmission of vibration, with the wires carefully placed along paths that have been meticulously determined as optimal. In Naim equipment, even the positioning of cable ties is carefully optimised and religiously adhered to in production.

Ground Switch

As with other Naim source components, the ND 555 is fitted with a ground lift switch which offers the option of either connecting the ground terminal of the output connectors to chassis ground (and thence mans earth) or isolating it via a low-value resistor. This allows optimum earthing to be achieved, without creating a ground loop, whatever the internal earthing arrangement of the preamplifier (or other equipment) to which the ND 555 is connected.

Correct setting of the ground switch has a significant effect on sound quality. The recommended default position is with the audio ground connected to mains earth as this provides an electrically 'quieter' environment for the ND 555's electronics, resulting in better sound quality. The floating position should only be used if an item equipment connected to the ND 555 connects audio ground to mains earth itself.

Appendix 1

DAC topologies: R2R and Delta-Sigma

Conceptually, the simplest way to convert a PCM-format digital signal into its equivalent analogue waveform is to arrange for each bit in the digital signal to control a switch which is turned on when its associated bit has the value 1 and off when the bit is 0. If the switches are all connected on one side to the same stable voltage reference and on the other to a resistor ladder comprising resistances of R and 2R (Figure A1.1) then each bit contributes appropriately to the overall output voltage.

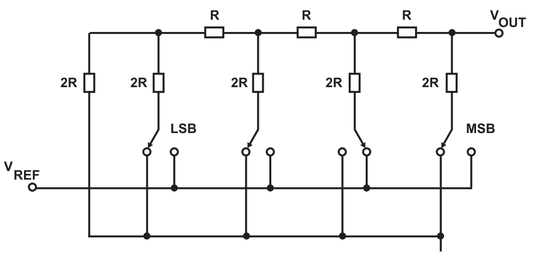


Figure A1.1. Voltage-output R2R DAC (4-bit). The switch labelled LSB is controlled by the least significant bit in the digital signal; at the other end of the R2R resistor ladder, the switch labelled MSB is controlled by the most significant bit

This scheme has the advantage that only 2N resistors is required (N being the number of bits) and they have one of only two values: R or 2R (hence the common designation 'R2R' to describe this type of DAC). The principal downside is that the resistor values have to be trimmed extremely accurately, particularly in a 24-bit DAC, if the output is to be monotonic, ie if a progressive decrease in coded signal amplitude is to result in a progressive decrease in analogue output voltage. (0.0015% accuracy is needed for a 16-bit DAC and 0.000006% accuracy for a 24-bit DAC.)

Realising the switches of a voltage-output R2R DAC as semiconductor devices in an IC poses design difficulties so a slightly different topology is often used instead (Figure A1.2) Again a voltage reference, R2R resistor ladder and one switch per bit are required but now output current, not output voltage, is proportional to the encoded signal amplitude. To convert this to an equivalent voltage, the output current is passed to a subsequent current-to-voltage (I2V) stage.

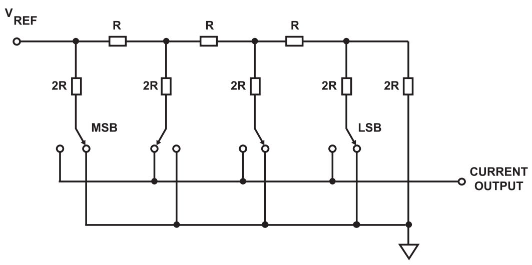


Figure A1.2 Current-output R2R DAC (4-bit)

Because of the manufacturing difficulties and costs associated with accurately trimming an R2R resistor ladder, DACs of this type have largely been replaced by alternatives employing a reduced number of bits combined with noise-shaping. The reduced number of bits (typically four or five, although 1-bit DACs were developed for audio use in the early 1990s) allows output monotonicity to be more easily achieved but, without further action, would substantially reduce the theoretically achievable signal-to-noise ratio (to 25.8dB for a 4-bit DAC or just 7.8dB for a 1-bit DAC, as opposed to 98.1dB for a 16-bit DAC and 146.2dB for a 24-bit DAC). This is avoided by implementing a digital feedback technique (Figure A1.3) which 'shapes' the DAC's quantisation noise so that the bulk of it is removed to ultrasonic frequencies, above the audible frequency range.

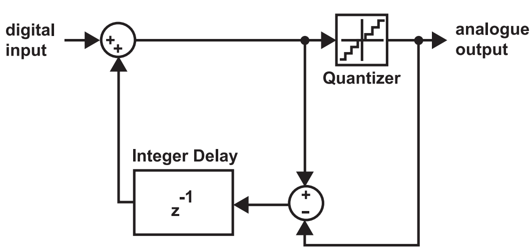


Figure A1.3. Block diagram of a simple one-stage noise-shaping delta-sigma DAC

The key benefit of noise-shaping delta-sigma DACs is that their low-bit quantizers are more easily designed and manufactured for high linearity, ie low distortion. Downsides are the high level of ultrasonic noise they generate, which has to be filtered, and the fact that high-order noise shapers can have nonlinearities and instabilities which introduce distortion and spurious 'idle' tones.

The benefits of noise-shaping delta-sigma converters are such that they have come to dominate the design of high performance audio DACs in the past two decades. But the sound quality achieved by the best R2R ladder DACs is still considered by many critical listeners to be superior.

Appendix 2

Oversampling and digital filtering

PCM digital audio classically requires the use of two low-pass filters which are critical to achieving low-distortion performance. The first – the anti-alias filter – precedes the analogue-to-digital converter at the recording stage where it prevents signal frequencies of above half the sampling frequency (the Nyquist frequency) entering the conversion process. Without this, signal frequencies of greater than the Nyquist frequency are aliased – they are mistaken for signal frequencies of less than the Nyquist frequency, introducing a form of nonlinear distortion which, if it falls within the frequency range of human hearing, is highly objectionable.

The second low-pass filter – variously known as the output, reconstruction or decimating filter – appears after the digital-to-analogue conversion stage in the replay equipment. This filter removes the image frequencies which are generated above the Nyquist as part of digital-to-analogue conversion. Figure A2.1 illustrates diagrammatically the output spectrum of a non-oversampled DAC prior to analogue output filtering. The signal spectrum is shaded; all the frequency components above this are image frequencies which it is the output filter's task to remove. In early digital audio equipment this required a costly, complex analogue network that was prone to the effects of component tolerances and temperature coefficients, and introduced phase distortion.

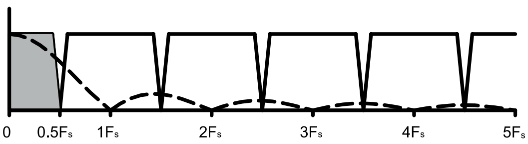


Figure A2.1. Diagrammatic representation of the output spectrum of a non-oversampled DAC before low-pass reconstruction filtering. The shaded area represents the desired signal; above it, images appear either side of multiples of the sampling frequency (Fs). The dotted line shows the filtering effect of the DAC's sample and hold function

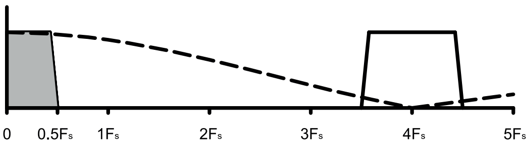


Figure A2.2. Diagrammatic representation of the output spectrum of a 4x-oversampled DAC before analogue output filtering. As before, the shaded area represents the desired signal; above it, digital filtering has removed some of the images, allowing a simpler analogue output filter to be used. The dotted line shows the filtering effect of the DAC's sample and hold function

Oversampling was introduced in the first Philips/Marantz CD players as a means of overcoming these problems through a combination of digital and analogue filtering. Oversampling comprises two signal processing stages: first the input signal is 'zero stuffed' to increase its sampling rate by an integer factor (eg 2×, 4×, 8×, 16x), then a low-pass digital filter is applied which attenuates much, but not all, of the unwanted image spectrum (Figure A2.2). What remains can then be removed using an a much simpler, cheaper, more stable analogue filter than required in the non-oversampled case.

Oversampling has benefits other than simplification of the analogue output filter, principally:

1) digital filtering is much more accurate as it is unaffected by component tolerances and temperature,
2) the digital filter can be configured to provide a range of filter characteristics from linear-phase to minimum-phase, allowing compromises to be struck between filter impulse performance and phase performance,
3) DAC quantisation noise is reduced: in the early Philips CD players, 4× oversampling allowed their 14-bit DACs to achieve 16-bit noise performance.

Appendix 3

ND 555 Product Specifications

High-res streaming	Up to 32bit/384kHz as standard
Network Connectivity	Ethernet (10/100 Mbps), Wi-Fi (802.11 b/g/n/ac) Dual rear aerial
Internet Radio	vTuner
iPod/MP3/USB	USB - yes iPod direct - no
Online Updates	Yes
Display	5” colour TFT
Storage Ability/Capacity	USB: 1 x front, 1 x rear
Bluetooth (aptX)	Yes (aptX HD)
Upgrades	Super Lumina, 2 × 555 PS
Control	RF remote control, front panel buttons, app
Ethernet	RJ45
FM/DAB/DAB + Tuner	No
USB	1 x front type A (1.6A) 1 x rear type A (1.6A)
Wired Remote	1 × 3.5mm output
Digital Audio Processing	SHARC ADSP21849 DSP
DAC	PCM1704U-K
Volume Control	Digital volume control for streaming services
Digital Inputs	1 x BNC , 1 x RCA, 2 x optical
Digital Outputs	1 x BNC
Analogue Outputs	1 × 5-pin DIN line, 1 x RCA line
AirPlay	Yes
TIDAL	Yes
Chromecast Built-in	Yes
Spotify Connect	Yes
Partymode	(ASRC) master & client 320K & 16/44 wav

Play Queue	Yes
Playlist	Asx, m3u, pls, wpl
UPnP Server	Yes local USB content
Seek to Time	All formats excepting VBR
Formats	WAV, FLAC, AIFF, ALAC, MP3, AAC, OGG, WMA, DSD64, DSD128, M4A, AAC
Gapless	Yes
UPnP Client	Yes
Languages	English, German, French, Spanish, Italian, Japanese, Russian, Chinese (simplified & traditional), Korean, Hindi, Portuguese, Arabic
Time Zone Aware	Yes
RF Remote	Yes Zigbee®
System Automation	RF & IR out
Power Supply Combinations	External only
Weight	14 kg
Product Dimensions	314mm deep x 87mm high x 432mm wide
Standby Switch	Front panel button, RC
Certifications	Apple AirPlay, Chromecast Built-in, Bluetooth, aptX HD, Spotify Connect, Zigbee RF4CE, Rovi, HDMI, NTFS, HFS, Wi-Fi