Wadia series 9 Decoding Computer system Overview

A Definition

What is a series 9 Decoding Computer system?

The Wadia series 9 Decoding Computer system is comprised of a software based controller, 931 Digital Controller, and individual mono digital-to-analog converters, 921 Decoding Computer, for each channel of audio output.

Consider a typical digital-to-analog converter comprised of a fixed input section, digital processing, a digital-to-analog conversion section, current-to-voltage conversion, and output stages. All of the subsections of the digital-to-analog converter (DAC) are limited in their overall scope and capabilities as a result of being contained in a single chassis. Not so with the Wadia series 9 Decoding Computer system.

The series 9 Decoding Computer system comprises exactly these same stages of circuitry as found in a typical DAC, however, it is able to adapt to emerging technologies, decoding schemes, and configurations and is flexible unlike any other digital product. The unique mono architecture allows the series 9 Decoding Computer system to reproduce two-channel audio at a level never before possible as well offering expandability into multi-level amplification, multi-channel audio systems, distributed audio, and even unforeseen configurations with equal aplomb. In addition, the series 9 Decoding Computer system features digital audio technology and performance well beyond the currently perceived “state-of-the-art.”

The series 9 Decoding Computer system

Wadia’s series 9 Decoding Computer system includes at least three separate chassis. The 931 Digital Controller takes the place of an analog preamplifier, receiving digital data directly from the source, acting as a user interface, and routing signals to the mono 921 Decoding Computers via Wadia’s proprietary Dual Fiber Interface. Each of the 921 receives the digital signal and converts it to analog output that is then passed on to amplification.

Comparing the series 9 Decoding Computer system to a Conventional Audio System

In conventional audio systems, digital source signals are converted to analog as early as possible, in either the CD player or the D /A converter. The analog signal is then sent via analog interconnects to a pre-amplifier, and then a power amplifier. Converting to analog early in the chain followed by many stages of analog electronics and interconnections inevitably leads to signal degradation in the form of added noise, loss, and distortion.

By comparison, the series 9 Decoding Computer system keeps signals in the digital domain much farther along the audio chain. The digital data from the original digital source is transmitted without any changes to the 921 Decoding Computers, the last components in the signal path prior to amplification. The series 9 Decoding Computer system represents a simplification of the overall signal path with far fewer circuits and interconnections than found in conventional analog-based systems.

Wadia 931 Digital Controller

The 931 Digital Controller acts as the central controller or "brain" for the series 9 Decoding Computer system. Although the 931 Digital Controller may look somewhat similar to an analog preamplifier, there are some very important differences. The 931 processes only digital signals and does not alter the signal in any way, unlike an analog preamplifier. The 931 routes data from its inputs to its outputs as determined by the user and will not impart a sonic signature in the manner of an analog preamplifier.

The 931 also decodes current digital audio input signals and converts them to Wadia's proprietary Dual Fiber Interface format. If the Audio/Video industry standardizes a new digital format, the Wadia 931 can be upgraded, while keeping the Dual Fiber Interface as its own system interconnection standard. This prevents the 921 Decoding Computers from needing regular updates or upgrades to support potential new formats that become accepted standards in the future.

In addition, the 931 Digital Controller acts as the user-interface for the series 9 Decoding Computer system, receiving commands from the remote control and front panel buttons, learning programmed configurations, and displaying status on the front panel display.

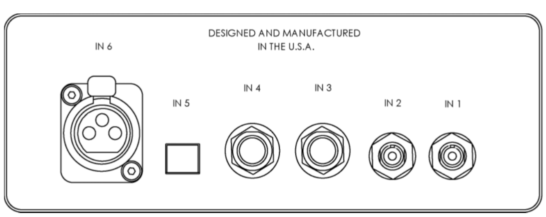
Input Circuitry

The 931 Digital Controller has six digital inputs capable of receiving and decoding 24-bit signals up to 96 kHz sample rate in AES/EBU or S/PDIF format. Special provisions have been made to allow the 931 to accept 24-bit 192kHz sample rate data when a standardized input is agreed upon. Additionally the 931 has been designed to decode DSD data for the SACD format. A proprietary DSD input board featuring a ST Glass Fiber Optic connection will be available with the forthcoming introduction of a series 9 Disc Transport.

The 931 inputs support two established Wadia jitter-reduction technologies: RockLok and ClockLink.

RockLok is Wadia's proprietary circuit that uses cascaded Phase-Locked-Loops (PLL) to recover the clock signal from the incoming data. RockLok produces a low-jitter clock signal from any standard digital source and can be used with all sources that comply with industry standards.

With ClockLink, the clock signal embedded in the incoming data stream is ignored in favor of a local crystal oscillator. This requires that the source component be synchronized to the 931 via a ClockLink output. *(See Distributed ClockLink)*



Any of the inputs can be user-configured for either ClockLink or RockLok mode.

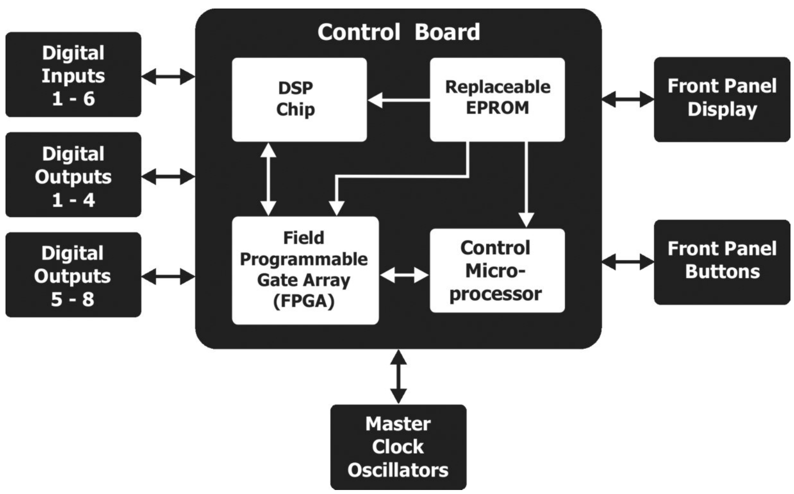
Field Programmable Gate Array (FPGA)

All inputs and outputs in the 931 Digital Controller are routed to a very large field-programmable gate array (FPGA) on the main control board. From the FPGA, the data can be directed to the Digital Signal Processor (DSP) chip, to any of the eight outputs, or to connectors for future circuitry. With 100 input and output lines and over 20,000 gates, this gate array provides tremendous flexibility for digital signal routing. The gate array also has the ability to perform operations on the data, including decoding or re-configuring the data for transmission in other formats.

The capability of the FPGA is far greater than what is required to execute the routing and processing functions performed by the current software release of the Wadia 931. In addition, the FPGA can easily be re-configured by changing the software contained on a socket-mounted EPROM to facilitate potential future upgrades, such as decoding new digital audio formats.

DSP Capability

The digital audio data is routed via the FPGA to a powerful Motorola Digital Signal Processing (DSP) located on the main control board of the 931. This DSP chip provides the flexibility to execute features that may be implemented in the future. For example, the DSP could provide custom digital filtering or



crossovers for multi-amplified speaker systems and room correction. It can potentially decode compressed digital audio data such as MLP and mp3. This embedded DSP is another example of the forethought that makes the 931 Digital Controller well positioned for upgrading easily as new features are required.

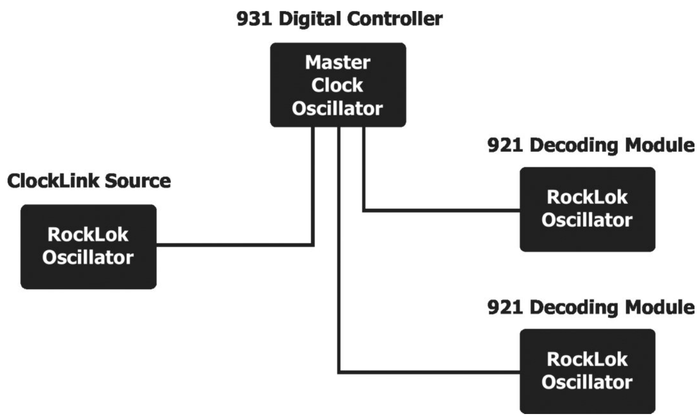
Distributed ClockLink

The Wadia 931 features a new implementation of Wadia's ClockLink jitter reduction technology in the form of multiple-mono ClockLink *(further information regarding Wadia's ClockLink system and the causes of jitter can be found at http://www.wadia.com/technology/clocklink/sld001.htm)*. As in previous versions, the function of ClockLink is to position a master oscillator as near as possible to the D>A converter chips to reduce transmission-induced jitter. In a stereo DAC unit, this can be accomplished by locating the clock oscillator adjacent and equidistant the to the DAC chips for each channel. All upstream transports are then synchronized to the DAC's oscillator clock signal.

In a synchronous system with multiple mono DAC units, such as the 931 with single-channel 921 Decoding Computers, the DAC clock must be located as near to all channels as possible. If the master oscillator were located in one of the 921, then all other towers would need to be synchronized to it, making for uneven clocking between channels. The series 9 Decoding Computer system uses the Distributed ClockLink scheme pictured above. The 931 Digital Controller contains an ultra-stable Temperature Compensated Crystal Oscillator (TCXO) that provides the reference clock for the entire system. The clock is transmitted to the source components and 921 Decoding Computers via dedicated ST optical glass fiber interconnects as part of the Wadia Dual Fiber Interface.

Each 921 Decoding Computer and all ClockLink enabled digital sources synchronize to the master clock in the 931 using the cascaded PLL RockLok circuit that locks on to the incoming ClockLink signal. Typical

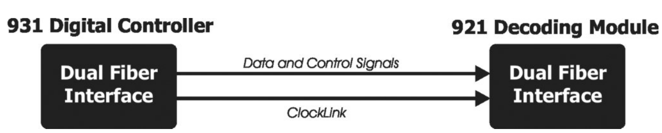
Voltage– Controlled Crystal Oscillators (VCXO) used in digital audio products must be able to lock on to all signals that satisfy the Level 2 specification for the S/PDIF interface. This implies that these devices must have a "wide mouth" or "wide-open window" input range in order to lock to the full range of possible clock signals required by the Level 2 specification. The Level 2 specification allows for greater variance



in clock signals so that lock can still be maintained with the less stable and accurate clock signals traditionally found in inexpensive equipment. In the distributed ClockLink system, the second PLL in the RockLok circuit, a precision VCXO need only lock onto the very narrow and precise range produced by the ultra-stable oscillators in the 931 Digital Controller. Utilizing a VCXO for the narrow frequency variation results in a more precise clock signal than typical VCXO circuits.

Wadia Dual Fiber Interface

As previously mentioned, the 931 Digital Controller converts all signals to Wadia's proprietary Dual Fiber Interface for transmission to the 921 Decoding Computers. The Dual Fiber Interface utilizes two ST glass fiber cables per channel that are capable of an extremely high data rate while also maintaining complete electrical isolation between the 931 and 921s.



One fiber optic cable carries digital audio data in its present implementation. The Dual Fiber Interface is capable of 192 kHz transmission and can be upgraded via software. This cable also carries control signals for communications between the 931 and the 921's, such as Volume, Balance, Emphasis, Mute, and Standby. The second ST cable carries only a dedicated clock signal. Having the clock on an isolated cable has significant advantages over interfaces where the clock signal is embedded in the data, as it is with the industry standard AES/EBU interface.

The AES/EBU digital transmission format uses bi-phase encoding to transmit data and clock on a single digital audio data stream. In 1992, Malcolm Hawksford and Chris Dunn submitted and published a paper titled, "Is the AES/EBU / SPDIF Digital Audio Interface Flawed?" to the Audio Engineering Society (AES). Their paper suggested that when the DAC word clock is recovered from the AES/EBU bi-phase encoded digital audio data stream, the jitter spectrum will be correlated to the spectrum of the encoded data. For example, when decoding a digitally encoded sinusoidal signal and its embedded clock from an AES/ESU data stream, the recovered clock will have a jitter spectrum with spectral lines at the frequency of the sinusoidal signal. This signal-correlated jitter has been identified as being particularly harmful to sonic performance. With Wadia's Dual Fiber Interface, the clock signal is independently transmitted, eliminating the possibility of signal-correlated jitter.

Glass Fiber Transmitters and Receivers
The ST glass fiber optic transmitters and receivers used in the Wadia 931 and 921 are not off-the-shelf integrated devices used by most other audio manufacturers. Such devices are optimized for the long distance fiber transmission commonly used in telecommunications systems. With exceptionally strong output signals and ultra-sensitive receivers (designed for transmissions over 1 km), these systems are vulnerable to degraded performance due to optical signal overload and internal cable reflection. The transmitters and receivers implemented in the series 9 Decoding Computer system are application specific discrete circuits optimized for the shorter (less than 30 meter) transmission distances found in home audio systems.

User Interface
The 931 Digital Controller performs all user interface functions. All commands entered via the remote control or front panel buttons are interpreted by the 931's control microprocessor. If the given commands are to be executed by the 921 Decoding Computers, such as "Standby" or a volume level change, the command is routed via the Dual Fiber Interface to the 921.

If it is a command to be executed locally, such as a programming change or the naming of a specific input or output, the microprocessor performs the function within the 931.

Any necessary status and command information is displayed within the easily read alphanumeric display window of the 931. The display can also be turned off if preferred to eliminate a potential source of noise within the 931.

Modular Design
The 931 Digital Controller's physical architecture and layout was designed and constructed with flexibility for future upgrades in mind. For example, the rear panel is a metal insert that can easily be replaced to accommodate new input or output connectors and input sections. All software is easily updated and can be installed in the field by an authorized Wadia dealer.

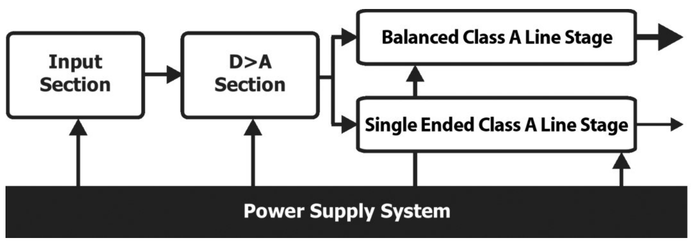
In addition, connectors and data paths are available so that new circuit boards can be "piggybacked"onto existing boards with access to data, control, and power signals. This access allows the 931 to accept any secondary processors, software, or filters necessary for decoding additional or future digital audio formats.

931 Power Supply
Although the 931 Digital Controller processes only digital signals, a great deal of attention was placed on the design of a robust and well-isolated power supply. For example, dedicated voltage regulators for the master oscillators provide clean, pure DC power for maximum clock stability. The toroidal transformer is triple-faraday shielded, damped, and housed in a custom enclosure to keep stray radiated fields from affecting the surrounding circuitry via noise and mechanical vibration. Multiple stages of regulation ensure quiet, stable DC power is available for all sensitive circuits. As noise is the most common source of jitter, the quietest possible source of power ensures the most accurate master clock signal and clock signal transmission to the 921.

In addition, ferrite filtering prevents digital noise from escaping via the AC input connector to the AC mains. This noise, in the form of harmonics, could otherwise affect the performance of other audio components in your system by modulating their response in the analog domain or increasing jitter present in their clocking schemes.

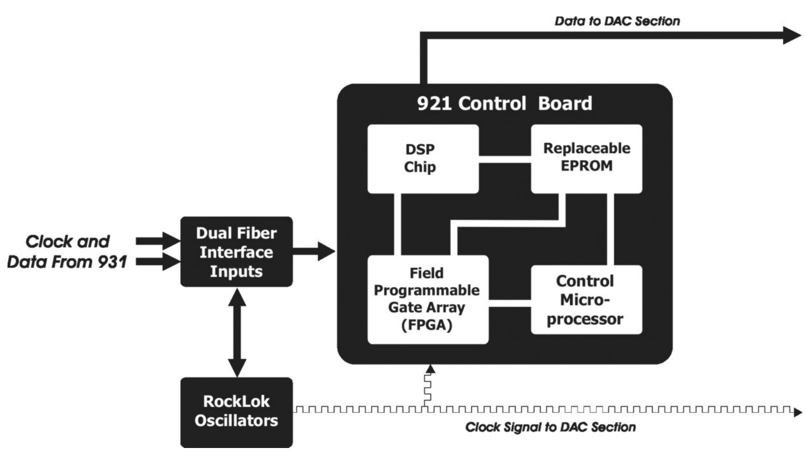
Wadia 921 mono Decoding Computers

Each 921 Decoding Computer is comprised of four major stages: the Digital Input Section, the D>A Section, the Variable Level Output Stage, and the Power Supply system.



Digital Input Section

The Digital Input Section functions as a receiver of information from the 931 Digital Controller via the Dual Fiber Interface, performs the DigiMaster interpolation filtering and volume control, and sends the digital audio signal to the D>A section.



Dual Fiber Interface

The 921 input section accepts signals from the 931 Digital Controller that are encoded using the Dual Fiber Interface format. As with the 931, the 921 hardware is capable of receiving 192 kHz data to support future formats.

RockLok Clock Recovery

As described in the section on Distributed ClockLink, the RockLok oscillator used in the 921 Digital Input section is a higher-performance version of the RockLok circuitry found in other Wadia products. Because it locks to the precisely controlled clock from the 931 Digital Controller, the 921’s VCXO is optimized for a narrower locking range, which results in much improved jitter performance compared to earlier Wadia products.

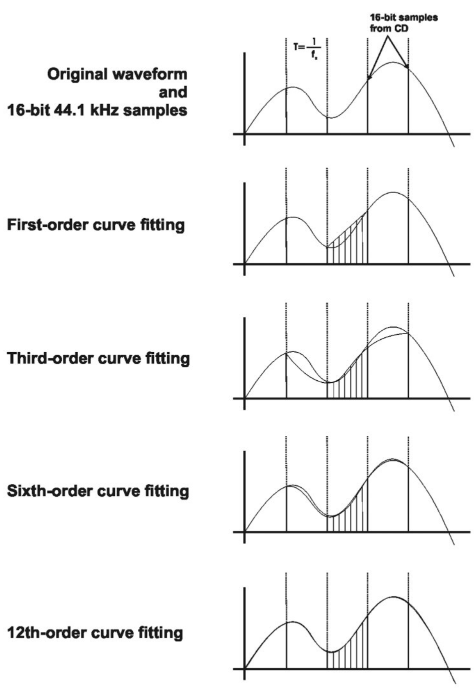
Field Programmable Gate Array (FPGA)

The 921 uses a Field Programmable Gate Array (FPGA) to route data and control signals on the control board. Additionally the FPGA performs the processing required for time staggering data.

Similar to the architecture of the 931 controller, the function of the FPGA can be changed by replacing the EPROM holding the configuration software, allowing the 921 to be easily upgraded.

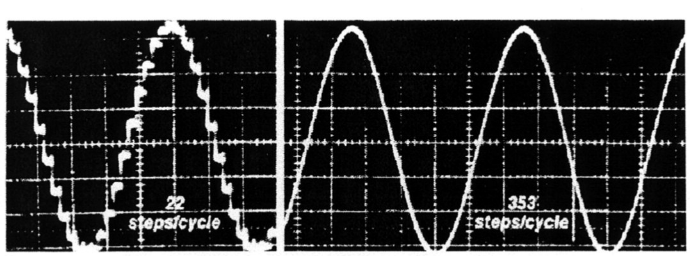
DigiMaster 1.4 Interpolation Filter

The Wadia 921 uses the latest generation of Wadia's proprietary interpolating digital filtering system, first introduced in 1991. The DigiMaster 1.4 Waveform Algorithm has been optimized for a wide range of performance parameters. Incoming data samples are reconstructed with special emphasis placed on time and phase accuracy. In contrast, typical off-the-shelf digital filtering is optimized for flat frequency response and fails to recreate the subtle musical presence and details that are preserved only with accurate time-based algorithms.



DigiMaster curve-fitting and interpolation. Only 8 times re-sampling shown for clarity.

The DigiMaster 1.4 interpolator uses a 12th-order, curve fitting, spline interpolation algorithm to precisely reconstruct the original analog waveform. Using digital signal processing (DSP), a curve is fitted that conforms to the current sample plus future and prior samples calculated at 48 bit resolution and output at 24-bit precision.



The DigiMaster 1.4 operates at the rate of 64-times re-sampling of 44.1 kHz, that is, 63 interpolated values are calculated for each original sample from the CD. Although the interpolated samples are calculated by DigiMaster 1.4 to 24-bit precision, the original data samples from the compact disc are not altered in any way. As shown in the illustrations below, a conventional 8x oversampling filter provides interpolated samples resulting in an approximated waveform with discrete stair steps. The DigiMaster 1.4 algorithm, operating at 64x oversampling, provides dramatically improved resolution in the waveform. Hence, the DigiMaster 1.4 algorithm more precisely reconstructs the original waveform.

D>A Section
The 921 D>A section converts the digital audio data into an analog signal.

As described above, the DigiMaster 1.4 algorithm produces digital audio data at a very high sampling rate. For example, using the standard CD format, the DigiMaster 1.4 algorithm calculates 63 interpolated values for each original sample. The output to the D>A section is 2.8224 million samples per second, each sample with 24-bit resolution. This number gains even greater significance when compared to the DSD (SACD) system, which decodes 1-bit data at this rate.

The 921 D>A section utilizes eight Burr Brown PCM 1704 DAC chips to convert the positive and inverted halves of the signal with a theoretical resolution beyond 24-bits; four for the positive half of the signal, and four for the inverted half. The individual circuit blocks are described in greater detail below.

Isolated Digital Coupler Array
The first element in the D>A section is an array of Burr Brown ISO150 Isolated Digital Couplers.

These devices are placed between each of the incoming digital signal lines and the DAC chips to provide complete electrical isolation of all signal and ground lines. This prevents any digital noise and ground currents from the digital circuitry from adversely affecting the performance of sensitive analog signals in the 921 D>A section.
Some digital products use optically-coupled isolators to isolate the digital inputs. The ISO150 avoids problems commonly associated with optocouplers, as optically isolated couplers require high current pulses and allowance must be made for LED aging. Optical isolators can exhibit a 50% reduction in light output from their internal LED within one year. In a worst-case scenario, this degradation could cause a performance or reliability issue. Since the ISO150 uses no LEDs, aging is not a factor and reliability and lifespan are greatly increased.

The ISO 150 devices used in the 921 Decoding Module utilize a tiny internal air gap that acts as a capacitor by transferring high frequency signals. These devices allow data up to 80 Mbits/sec to be transmitted, while still achieving a very high degree of DC and low frequency isolation. Although the air gap is small, each device can withstand peak voltages up to 2400 volts.

Time Shifted DAC Array
To handle the 2.8224Mhz, 24-bit data stream from the DigiMaster 1.4 interpolation filter, the 921 D>A section uses a total of eight each of the Burr-Brown PCM 1704 24-bit DAC chips per channel configured in a balanced, time-shifted array. The Time-Shifted DAC array increases the speed at which digital information is converted, as well as increasing system resolution.

In most designs, digital-to-analog conversion rates are limited by the settling time of the DAC chips and their current-to-voltage (I/V) amplifiers. When the conversion rates exceed the typical 8-times at 44.1 kHz (or 352 kHz) rate, insufficient settling time between samples degrades sonic performance. Wadia's patented time-shifted DAC array overcomes this problem by feeding data to the DAC pairs sequentially. As a result, each DAC only has to decode a fraction of the total samples, allowing ample settling time between successive data values.

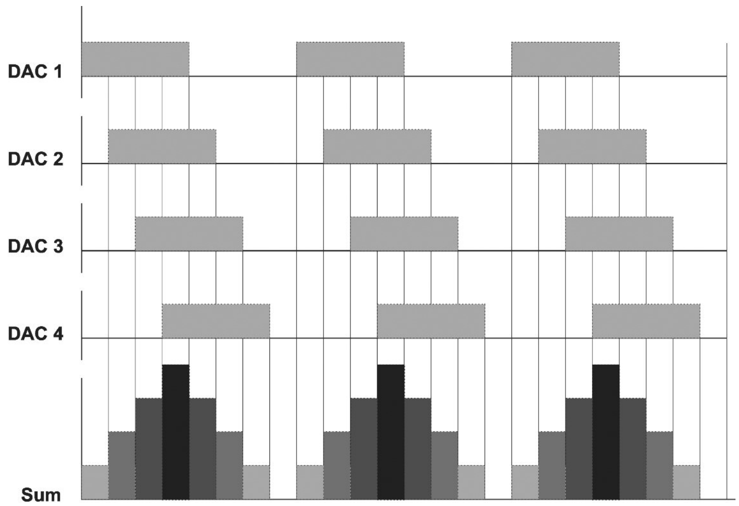


Figure 6: Time Shifting DACs for Increased Resolution

As mentioned above, the Time Shifted DAC array provides higher resolution, e.g., when the current output of each DAC in the array is summed, the resulting signal has higher resolution than would be possible using individual DAC chips due to a lower level of noise in relation to the audio signal.

The illustration at right provides an example. Here, each DAC puts out a LSB (Least Significant Bit, the bit with the smallest value or representing the smallest portion of audio data) signal, representing the smallest amplitude signal possible for that DAC.

When the currents from each DAC are summed, the resulting summed output signal has an improvement in signal-to-noise ratio (SNR): the noise floor remains the same, yet the overall output has increased, yielding an improvement in the theoretical resolution.

This improvement is measured by the square root of the number of devices that are employed. For example, using two DAC chips would increase SNR by the square root of two, which is 1.41. By employing four DAC chips, the SNR is improved by a factor of two (equivalent to 1 bit). By using a total of eight DAC chips, the Wadia 921 Decoding Module achieves a resolution that is 2.82 times greater than a single DAC could provide.

Wadia SC-2D Discrete Current Conveyor

Conventional digital-to-analog (D to A or D>A) converters utilize an integrated circuit operational amplifier (op amp) for current-to-voltage (I/V) conversion. In the early days of compact disc playback, many manufacturers and enthusiasts alike learned that by experimenting with different op amps, they could improve the overall sound of the player. We have since learned that the large amounts of negative feedback and the demands on settling time required by the integrated I/V circuits are detrimental to sonic performance.

The reason for this becomes clear when considering the nature of the DAC section's output. The rapidly switching current output of a DAC contains the fastest transients of any point in the audio signal path. It is therefore especially critical to transform these current waveforms into an analog voltage with the least possible dynamic distortion. Using a feedback amplifier (typically an op-amp) results in transient inter-modulation distortion (TIM), slew induced distortion (SID), settling time errors, etc., due to the inability of the op amp and feedback loop to keep pace with the rapid changes from the DACs. In general, any op amp based I/V converter will suffer from these phenomena and can severely limit overall sonic performance.

The 921 uses a proprietary discrete circuit known as the SC-2D Current Conveyor to perform the critical current-to-voltage conversion at the output of the digital to analog converter. The Current Conveyor function does not utilize any global feedback, providing a significant improvement over conventional I/V amplifier circuits. Hence, the Current Conveyor allows the 921 to realize the goal of transforming the DAC output current into a pure analog voltage while avoiding the dynamic distortions common in many other products.

The latest generation of Current Conveyor circuit design is based on an innovative, patented connection of Wilson current mirrors with additional circuitry to optimize transient response and a novel output current mirror, which stabilizes output impedance. The net result is a constant impedance response regardless of frequency output from the DAC section. Maintaining a constant impedance from the DAC section results in linear output response that will not vary or degrade with dynamic changes in the frequency of the output signal. The SC-2D system is then able to generate voltage output with drastically reduced distortion by driving a single, high-quality 0.1% metal film resistor directly, again resulting in linear response regardless of the frequency applied to the resistor while also drastically simplifying the signal path by connecting directly to a high current Class A output stage. Discrete circuit implementation also avoids the use of polysilicon resistors. Polysilicon resistors are inexpensive, commonly used microcomponents in integrated circuits, and are of poor quality in comparison to the larger and more expensive metal film resistors used in the SC-2D I/V section.

Class A Variable Level Output Stage

The line-level outputs on the 921 Decoding Computer are typically connected directly to an analogue amplifier and perhaps to the sub-woofer amplifiers found in some loudspeaker systems. The line-level output stage is a new, discretely implemented Class A design and is capable of driving the input section of any amplifier, even through extremely long interconnect cables. This design takes the place of a typically implemented Class A/B integrated circuit output buffer. The use of a high-current Class A design significantly reduces noise by eliminating the crossover distortion created when the output transistors are switched on or off to reproduce of each half of the output waveform – in a Class A design, the output devices are never turned off.

The maximum output level of this output stage can be adjusted to match the overall sensitivity of the system installation and maintain the highest possible resolution from the digital volume control system via a set of switches easily accessed through the rear panel of the 921. Each switch change actuates a relay that routes the audio signal through a single high-quality, 0.1% metal film resistor to attenuate the full scale output from the output section. This is in comparison to resistor networks, where the audio signal is routed through different combinations of multiple resistor arrays in order to attenuate the maximum audio output of the circuit. The benefits of utilizing a single resistor in place of a resistor network are two-fold.

First, the 0.1% tolerance of a single resistor is much tighter and accurate than the combined tolerances of multiple resistors.

Second, the use of a single resistor avoids the complex signal path required in the implementation of a resistor network, which would direct the delicate audio signal through an additional number of separate components, solder joints, and circuit board traces.

Further information regarding the Wadia Digital Volume Control is available at:
http://www.wadia.com/technology/dvcontrl/sld001.htm

Power Supply System

The 921 Decoding Computer power supply system comprises six independent and separately bussed power supplies per channel, with each supply powering only a portion of the 921 Decoding Module. Each power supply utilizes transformers, filtering, and regulation most appropriate for the circuitry it supplies. This high degree of independence minimizes inter-modulation between power supplies, which was found to be particularly important during listening tests.

Current Pre-regulators

Five of the power supplies in the 921 use current pre-regulators between the filter capacitor output and the voltage regulator input. Current regulators have high (ideally infinite) source impedance. That is, they supply the same current regardless of the variations in the load impedance. The high impedance blocks noise between source and load, thereby providing significantly improved isolation compared to conventional voltage regulators.

Inductor Input Filtering

Although inductor input filtering is not a new concept, it is surprising that it is not more widely utilized. Compared with capacitor-input designs, our experience indicated significant improvements in sonic performance.

Placing a carefully sized inductor between the output of the bridge rectifier and the filter capacitor input provides three distinct advantages.

First, by smoothing the flow of current from the rectifier, the inductor eliminates the current charging spikes that occur when a bridge rectifier charges a large capacitor directly.

Second, the inductor provides a second filtering element, doubling the effectiveness of the filter in attenuating high-frequency AC line noise.

Third, the inductor significantly reduces line-frequency AC ripple by providing a steady current discharge. Following is an explanation of each of these advantages.

Inductor-input Filters: Elimination of Current Pulses

In all products with capacitor input filters, current from the rectifier to the filter capacitors flows in a series of steep, high magnitude pulses. Charging current pulses are caused by the fact that current only flows from the rectifier to the filter capacitor for the very short part of the AC cycle when the rectifier voltage is higher than the voltage stored on the capacitor. This short charging interval means that all of the current must be delivered in short bursts. With very large power supply, these pulses have been measured at over 50 amps peak. Due to their large amplitude, these current pulses are a source of noise radiated within the component, as well as back onto the AC line, often affecting the performance of other components in a system. With inductor input power supply filters, the inductor acts as a current smoothing element between the rectifier and the filter capacitor. This produces a smooth, continuous flow of current, eliminating charging current pulses.

Many vacuum tube amplifiers use “pi-network” filters. These filters place the inductor between two sets of filter capacitors. As a result, the interaction between the rectifier and first set of capacitors is the same as with conventional capacitive filters without inductors, and they still suffer from the effects of charging current pulses.

Inductor Input Power Supplies: AC Line Noise Filtering

Noise from light dimmers, high-speed switching circuits (think: television, VCR, and DVD player), radio transmissions, and the like contaminate the AC line voltage. Typical power supplies act as a single-pole filter to attenuate AC line noise from entering the audio circuitry. With the inductor acting as a second filter element, inductor input filtered power supplies act as a 2-pole filter, further reducing AC line noise.

Inductor Input Power Supplies: Reduced Ripple

As previously mentioned, in capacitor filtered power supplies, the capacitor is charged by the rectifier during intervals when the rectifier voltage is higher than the voltage stored on the capacitor.

During the rest of the cycle, when no current flows form the rectifier, the capacitor is discharged by the load. This charge/discharge cycle results in power supply ripple.

With inductor-input filters, the inductor stores a charge as it develops a magnetic field during the part of the cycle when the rectifier voltage is high. Then, when the rectifier voltage is low, the magnetic field provides continuous current to the capacitor. The inductor input filter results in smooth, continuous capacitor charging and a dramatic reduction in power supply ripple.

High-Performance Filter Capacitors

The main filter capacitors for the line-level output stage were selected based on listening tests involving numerous filter capacitor configurations. The selected parts are custom manufactured for Wadia by a company well known for high-grade capacitors.

Every power supply in the 921 has additional capacitive filtering, and large values are bypassed by higher-speed capacitors in parallel.

The Final Word

We hope your understanding of some of the design principles and unique component selections utilized in the Wadia series 9 Decoding Computer system as well as your own sonic evaluation results in a level of appreciation similar to our own.

Although many manufacturers claim to invest painstaking hours in the development of their products, the series 9 Decoding Computer system can truthfully be represented as the result of nearly 12 years of continuous research. In fact, we are continuing to develop this technology as we build for the future.