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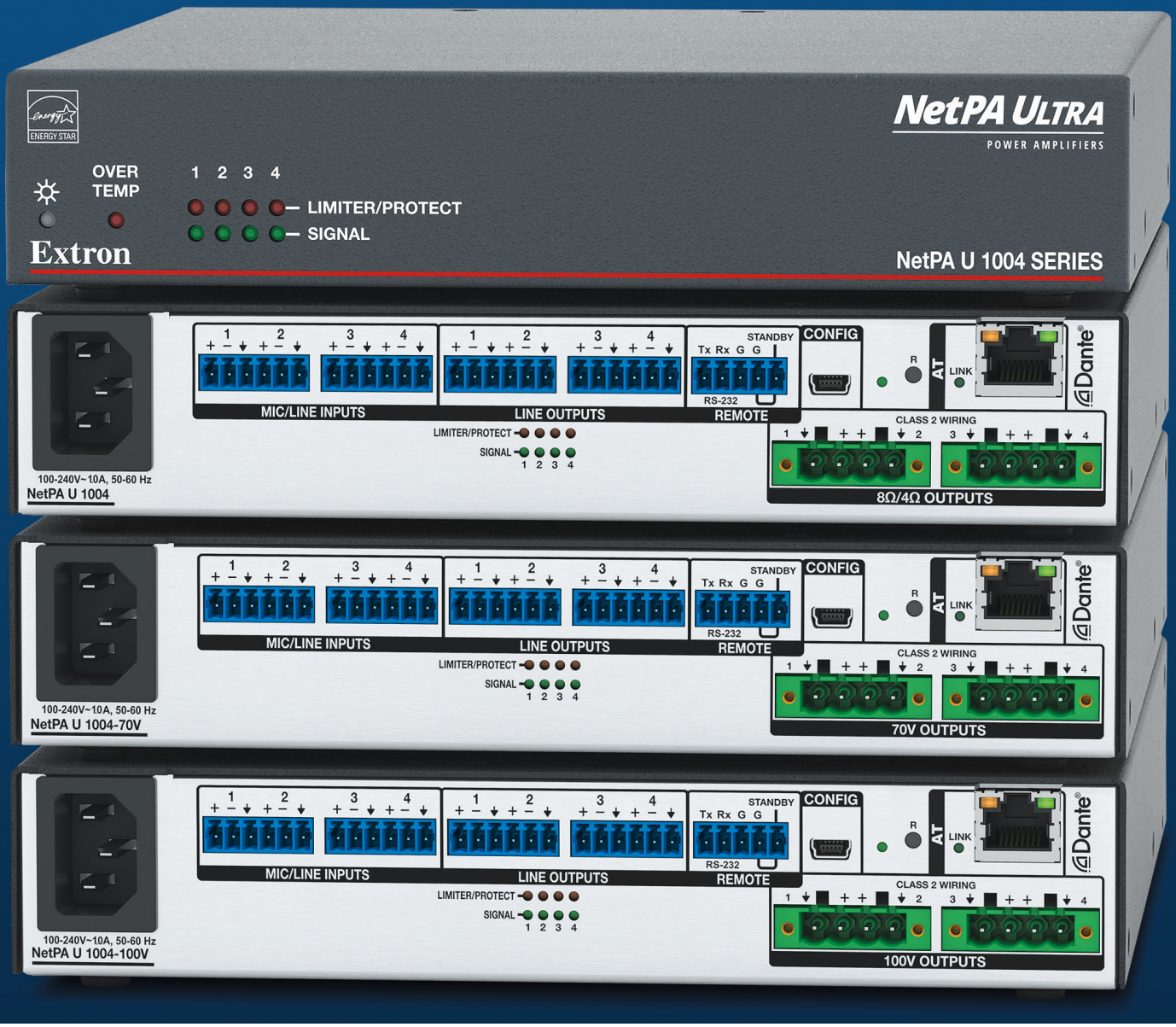
Extron NetPA Ultra Series Amplifier



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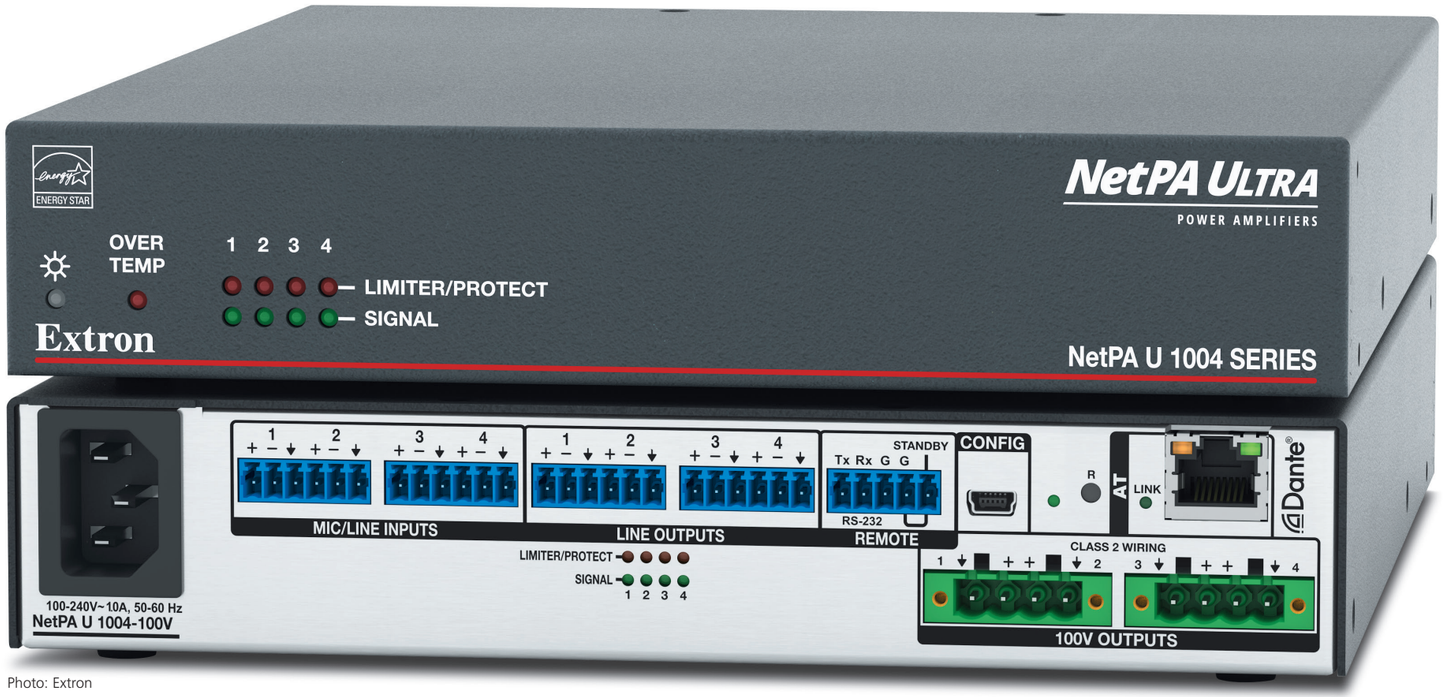


Photo: Extron

Extron NetPA Ultra Series Amplifier with DSP System and Dante Interface

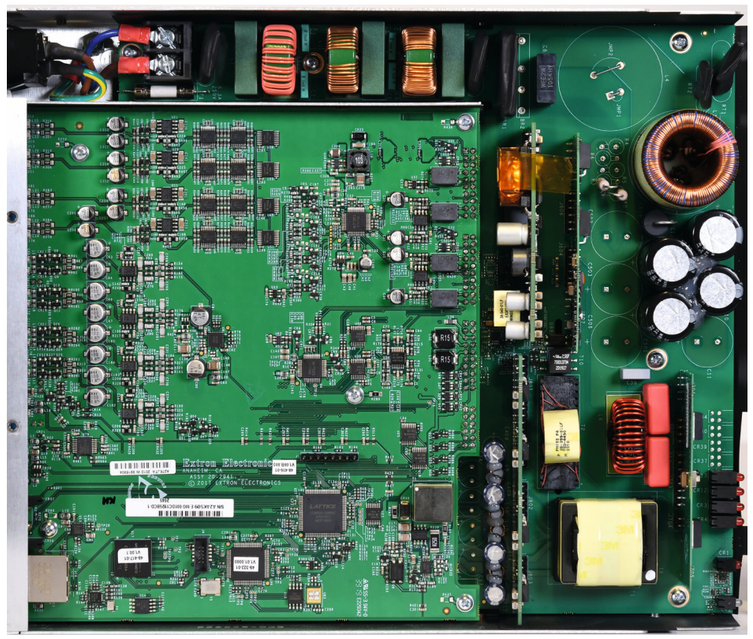
With the new NetPA-Ultra amplifier models, Extron now also offers a compact 9.5" Ultra Series amplifier with integrated DSP and Dante interface. We tested the four-channel NetPA PA U 1004 for Low-Z operation with 4x 100 W at 8 Ω or 4 Ω loads.

Text, measurements & pictures: Anselm Goertz

In AV Systems, amplifiers are required in many places. This can be for simple speech reinforcement, for music playback in the foreground or background, or for multimedia systems with picture and sound. In addition to simple amplification, audio signal processing filters and functions, such as compressors and limiters, are often required for optimal speaker output. Another aspect that is becoming more and more important for users is the option of networking via Ethernet, so that all devices can be set and monitored from a central point without direct access to the hardware. Since the audio source signals are usually not only local, but include a selection from many sources, the next step towards audio transmission via the network is a logical step. The Extron NetPA amplifiers with DSP and Dante interface offer all of this. The combination of NetPA technology with the Ultra amplifier series is new. The new Ultra Series Amplifiers were presented at the beginning of last year and initially appeared as XPA Ultra models without DSP and network interface. With the NetPA Ultra series, Extron now offers a combination of both.

NetPA Ultra series

A comprehensive test report of the Ultra Series Amplifiers was already published in the Professional System, issue 1/2020. The report ended with the conclusion that a NetPA version of the Ultra amplifier would be desirable for many applications. As far as the actual amplifier technology is concerned, it remains on what is already known from the Ultra series. All amplifiers in the NetPA Ultra Series are 9.5"Devices with 1 U height and →



Extron NetPA U 1004 interior view

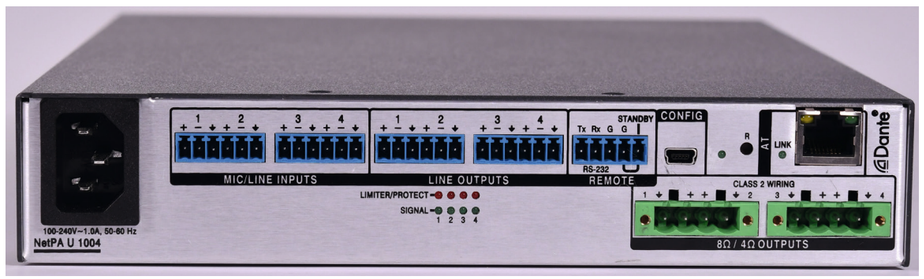
Figure 03: Interior view of the NetPA U-1004 with the DSP and network board above the actual output stages. The power supply unit is in the right third, the line filter at the top at the edge.

Extron NetPA U 1004 front view

Figure 01: Front view of the NetPA U 1004, which is similar to the XPA Ultra models.

Extron NetPA U 1004 rear view

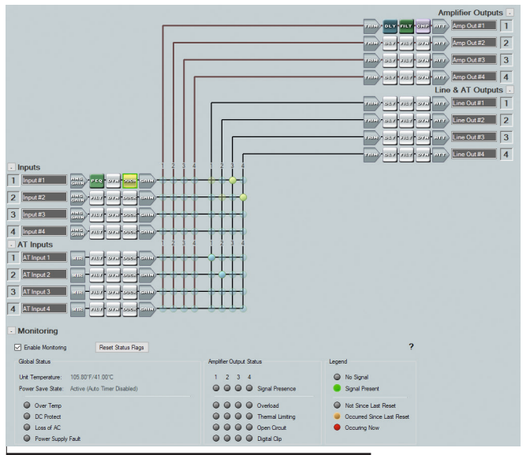
Figure 02: Rear of the NetPA U-1004 with network interface and four analog balanced line-level outputs for connecting additional devices.



fanless passive cooling. Currently there are the two basic models, the NetPA U 1002 and NetPA 1004 with 2x 100 W and 4x 100 W, respectively. Both models are also available in a Low-Z version or for 70V or 100V systems. The NetPA U 2002 SB and NetPA U 8001 SUB models are new. The NetPA U 2002 SB delivers a maximum of 2x 200 W at 4 Ω or 8 Ω or 400 W bridged at 8 Ω or 16 Ω.

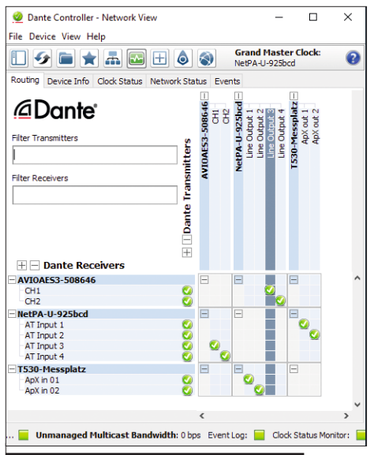
The 400 W is also available in bridged mode for 70V or 100V systems. The single-channel NetPA U 8001 SUB with 800 W power into 8 Ω is specially designed for operation with subwoofers.

The NetPA U 1004 was provided for testing, with 4x 100 W for low impedance output in 4 Ω or 8 Ω. It's probably the most popular model because it can be used



Block diagram in the DSP configurator setup

Figure 01: Block diagram of the DSP system as it appears in the associated PC software. There are eight inputs and eight outputs that are linked by a complete mix matrix.



NetPA U 1004 in the Dante controller

Figure 02: The Dante controller shows the NetPA Ultra amplifier with four inputs and four outputs on the network.

Gain	0 dBFS	Noise	
dB	entspricht	dBFS	dBFS(A)
0	+22 dBu	–104	–106,5
20	+2 dBu	–104	–106,5
40	–18 dBu	–99,5	–101,6
60	–38 dBu	–80,5	–82,6

Table 01: Clip-limit and interference level depending on the gain value. The equivalent input noise (EIA) value is –118.5 dBu.

in a wide range of applications from conference rooms to classrooms, and even hotel bars.

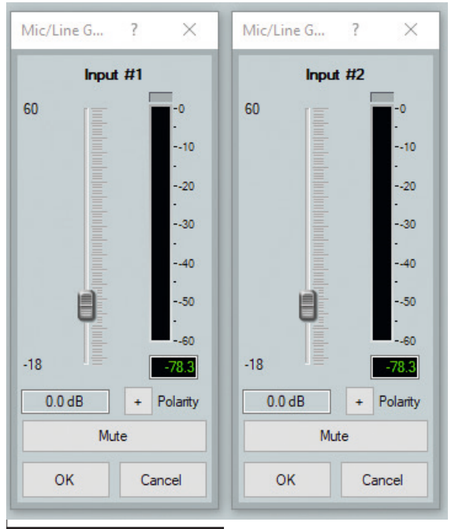
DSP system

The focus of this article is the new integrated DSP with its extensive functions, as the power amplifiers and their power supplies were already covered in detail in the test report of the XPA Ultra Series (issue 1/2020). The Extron DSP Configurator Software is the user interface for full control and management of the DSP functions. The DSP is accessed via the Ethernet interface, which also provides the interface for the Dante network. The NetPA U 1004 appears in the software with its block diagram (FIG. 01). Here you find four analog and four Dante-based inputs. The analog inputs feature a preamp with a wide gain range from -18 to +60 dB. In each input path a filter bank with three filters, a compressor/limiter and a ducker/adaptive gain function, as well as a digital gain, is available. The output side looks similar with eight Channels. Four are directly connected to the internal power amplifiers. The other four are routed to the analog line level outputs and in parallel to the four Dante-output channels onto the network. The processing on the outputs consists of gain, delay, a compressor/limiter and a filter bank with nine filters. The inputs and outputs linked together via an 8x8 matrix mixer. The DSP offers much more than just simple controller functions for the connected loudspeakers. Additional conventional power amplifiers can be controlled via the analog outputs. The inputs can be used for microphone signals routed to the internal amplifier or to the Dante network. Let's take a simple example with a small conference center with several rooms and a NetPA Ultra amplifier in every room. The Amplifiers can feed the speakers and additional delay lines in each room. The analog inputs take all signals meant for playback. If an additional active subwoofer is used, the signal can be filtered and routed through the analog line level output. At the same time, signals from a central source or the other rooms can be sent over the Dante network. With just a few, easy to configure devices it's possible to build a complete functional conferencing system.

The Dante interface in the NetPA Ultra amplifier utilizes the UltimoX4 chip from Audinate, with four inputs and outputs to the Dante network. Figure 02 shows a simple example of how the NetPA Ultra amplifier is connected to the APx555 measurement system with the Dante controller software. The measuring system communicates with the PC through USB. The PC is equipped with the Dante Virtual Soundcard (DVS) to access the Dante network.

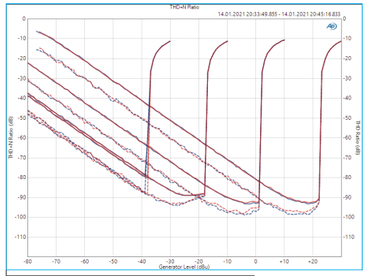
ADC, preamp and DAC

The four analog inputs of the NetPA U 1004 are symmetrical and equipped with preamplifiers that also allow the connection of microphones. In the setting with 0 dB gain, the input sensitivity for 0 dBFS full scale on the digital side is +22 dBu at the analog input. The gain can be increased in 1 dB steps up to a maximum of +60 dB, where full scale is already achieved at -38 dBu. This is more than sufficient for most microphones. The gain setting allows even values below 0 dB down to –18 dB, where the sensitivity is then at least mathematically +40 dBu. It should be noted, however, that the –18 dB ➔



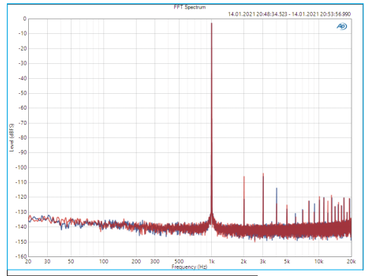
Gain setting of the inputs

Figure 03: Gain setting for the inputs with values from 18 dB to +60 dB. At 0 dB, 0 dBFS corresponds to a level of +22 dBu on the digital side.



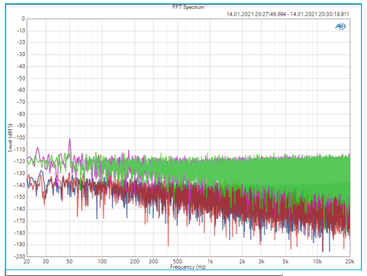
THD + N and THD at the inputs

Figure 04: THD + N (solid lines) and THD (dashed) depending on the input level (x-axis) at 0, 20, 40 and 60 dB gain. Measured as an example for input 1 (bl) and 2 (rt).



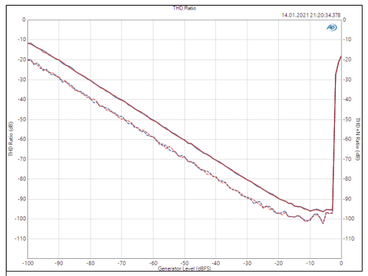
FFT spectrum at the entrance

Figure 05: FFT spectrum at 0 dB gain and +19 dBu input level



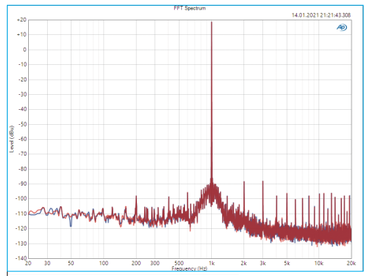
FFT spectrum of the interference component at the input

Figure 06: FFT spectrum of the interference component at 0 dB gain (bl, rt) and at +60 dB gain (rs, gr). The S / N is 104 dB or 80.5 dB.



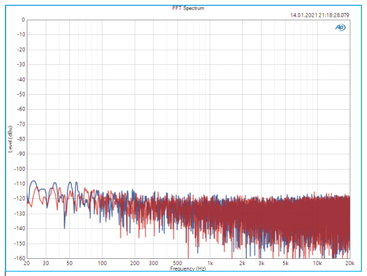
THD + N and THD for the DAC with output stage

Figure 07: THD + N (solid lines) and THD (dashed) depending on the input level (x-axis) for the DAC with output stage. The maximum output level is +22 dBu. A digital gain of 3 dB was set for the measurement .



FFT spectrum at the output

Figure 08: FFT spectrum at +19 dBu output level



FFT spectrum of the interference component at the output

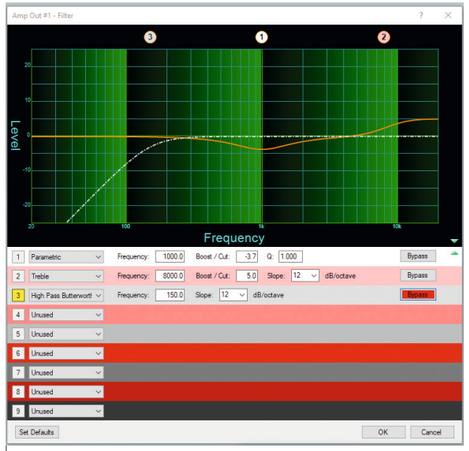
Figure 09: FFT spectrum of the interference component at the output with 85.6 dBu

can only be set via a digital gain and not via a PAD, as is known from mixer inputs. The level limit remains independent of the setting at +22 dBu. Higher levels lead to clipping, which in practice is unlikely to occur at all, since the +22 dBu corresponds to the typical maximum line level.

Some users may really miss the lack of phantom power at the inputs, which is required for almost all microphones, except for a few dynamic models. The reason given by Extron is that with phantom power it would no longer have been possible to meet ENERGY STAR conformity. That may be, and is of course also praiseworthy, but on the other hand it leads to the fact that a separate power adapter must be used for each microphone with phantom power, which also requires its own power supply unit. This is even less favorable than an amplifier with its own phantom power. To argue the other way, in 80% of the cases no phantom power is required at the inputs, because mostly wireless microphones are used, and then the solution without permanently installed phantom power is preferred.

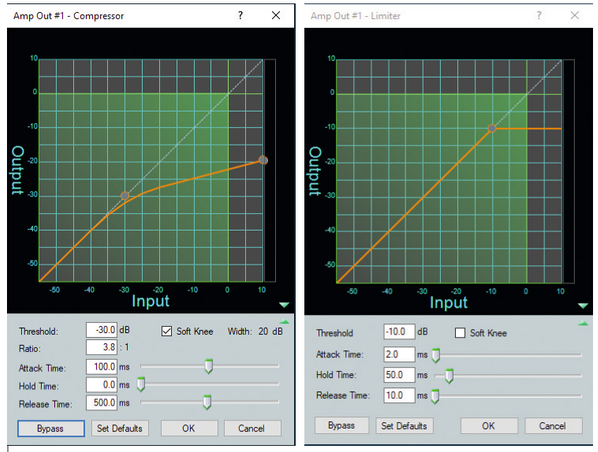
More interesting are the measurements, since the microphone preamps are located at an extremely sensitive point in the signal path, where an extremely low microphone level must be amplified to line level.

Figure 04 shows measurements of THD + N and THD depending on the input level for gain settings of 0, 20, 40 and 60 dB. The control limit for 0 dBFS is where the curves bend steeply upwards on the digital side. Even with a maximum gain of 60 dB, a THD of –90 dB (= 0.003%) is achieved. Since the composition of the distortions is important, in addition to the total distortion value, Figure 05 shows the FFT spectrum 3 dB under full scale for a 1 kHz sinewave. K2 and K3 and a little K4 are dominant. All higher harmonics are below –120 dB and are therefore completely negligible. K2 and k3 are also more than 100 dB below the fundamental wave, which requires no further discussion. In addition



Filter settings in the DSP Configurator software

Figure 10: Filter setting in the output channels with nine freely configurable filters



Limiter and Compressor

Figure 14: Setting the compressor/limiter functions in the DSP. The limiter (right) limits the output signal hard. The compressor (left), on the other hand, only reduces the gain in a ratio defined by Ratio.

to distortion, the Signal-to-Noise ratio (S/N) is an important aspect for the microphone input. For the preamp, including the needed ADCs, this is easily determined by terminating the input with a resistor? (here 200 Ω) and then measuring the Noise level in the digital domain as linear or A-weighted sum level. This value corresponds directly to the S/N since the reference level is always 0 dBFS.

It is also important to look at the FFT spectrum for the interference signal (FIG. 06), as individual single-frequency components can be much more disturbing than a broadband, evenly distributed white noise. Also, here the NetPA U performs in an exemplary manner. Except for a negligible small line at 50 Hz in the setting for 60 dB gain, the FFT spectra only show finely distributed noise.

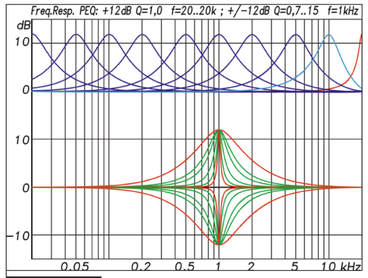
After the maximum of eight input signals (4x analog and 4x via Dante) have passed through the internal processing and the mix matrix, eight output paths are available. Four of them feed directly the four output stages in the NetPA U 1004 and are not available externally. The other four outputs are routed as analog symmetrical line-level outputs and are in parallel also available on the Dante network.

For the output side with the DAC and the associated analog output stage, comparable measurement criteria can be applied as on the input side. The distortion values are also very good at –95 dB. The maximum output level is +22 dBu and is sufficient for all following devices. The interference level at the output is linear weighted -85.6 dBu and A-weighted -88.2 dBu, which results in a particularly good S/N of 110 dB calculated.

Filter

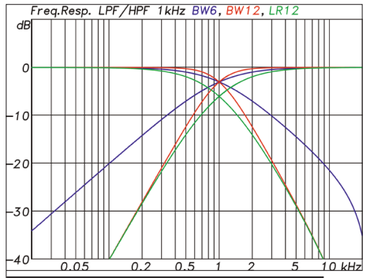
In addition to the gain, delay, and mixer functions in the DSP of the NetPA U 1004, various filters and functions that influence the signal dynamics are available to the user in each input or output path. Figure 10 shows the user interface for the filters in the outputs. In addition to fully parametric Bell filters, there are high and low pass functions and shelving filters, which are named here with Treble and Bass. All filters can be tuned from 20 Hz to 20 kHz, the gain is a maximum of +/- 12 dB.

The range of adjustments for the Bell filter is shown in Figure 11. The Filter Q can be adjusted from 0.70 to 15, which allows all uses, from a wide band adaptation to sharp notches. The upper part in Figure 11 shows a Bell filter with 12 dB gain and a Q factor of 1, for various frequencies varied from 20 Hz to 20 kHz. Up to 10 kHz, the filter curve remains unchanged. At even higher frequencies, approaching half the sampling rate at 24 kHz, leads to a compression of the curve. This effect is created by transforming the filter function from the analog world with a mathematically infinite frequency axis into the digital world, where the frequency axis ends at half the sampling rate. The compression of the filter curve does not mean a deterioration in the audio quality, but only a deviation from the known analog filter function. To avoid that, you would have to use a higher sampling rate such as 96 kHz or curve would be corrected by calculation. With mixing consoles in music production, both methods have a certain relevance. In the case of the NetPA Ultra amplifier it is not important since typically the filter is set once and never touched again. ➔



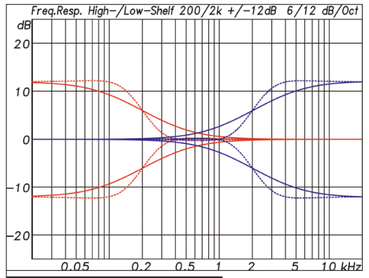
Bell filter

Figure 11: Filter functions of the Bell filters with a maximum gain of ± 12 dB for frequencies from 20 Hz to 20 kHz and a Q from 0.707 to 15. The upper curves show a compression of the filter curve at high frequencies close to half the sampling rate.



Butterworth and Linkwitz-Riley Filters

Figure 12: High and low pass filters. Butterworth filters with 6 and 12 dB/Oct and Linkwitz-Riley filters with 12 dB/Oct are available; here as an example for a corner frequency of 1 kHz each.



High and low shelf filters

Figure 13: High and low shelf with 6 and 12 dB/Oct slope and a gain of maximum ± 12 dB; here as an example for corner frequencies of 200 Hz and 2 kHz.

The high and low pass filter (FIG. 12) are limited to Butterworth and Linkwitz-Riley functions with 6 to 12 dB/Oct slope. This is sufficient. Only for the separation of subwoofers you might wish for an even steeper filter with 24 dB/Oct. Those who are familiar with filters can build this using two 12 dB/Oct filters. The corner frequencies can be adjusted freely up between 20Hz and 20 kHz.

The shelving filters (FIG. 13) are often used as simple tone controls, which also explains the naming as treble and bass. Typically, such filters would be placed on the user interface, where the user can quickly and easily adjust something if the music is too thin, or the speaker sounds too dull. The type of shelf-filter with 6 or 12 dB / Oct slope and which frequencies are suitable should be tested on site during commissioning. The gain is a maximum of +/- 12 dB. The set frequency is defined at the point where half gain in dB is reached.

Dynamics processors

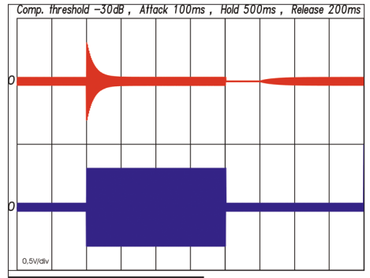
Dynamic functions in the signal path play a significant role in all systems where signals are played or spoken live, when it is not always possible to predict how loud a signal will be.

The microphone on a lectern sometimes captures sound with a low level from a distance of half a meter and outside the main axis and sometimes in a loud voice from a short distance. This can result in level differences of 20 dB or more. You do not want to expose the audience to these harsh jumps in signal level. To catch such extremes, a compressor is placed in the input path, which starts to reduce the gain from a certain threshold value. The compressor is set via its parameters: Threshold (threshold value), Ratio (ratio of compression), Attack (response time constant), Hold (hold time after an event), and Release (reset time constant). The window for setting all these parameters and a clear characteristic curve is shown in Figure 14 If you select the Limiter setting, the ratio value is fixed at ∞: 1. The limiter function is primarily in the outputs to protect the speakers and avoid Clipping of the amplifier. Figure 15 shows how the limiter works in the NetPA Ultra amplifier.

Two other functions worth mentioning related to the Input channels are the Ducker and the adaptive gain. In contrast to the compressor or limiter, both are not controlled by their own signal (target), but by the signal on another channel (source). Figure 16 shows how the Ducker works. There is a signal for the first channel (target, red curve above). As soon as another signal exceeds a certain threshold value on the second channel (source, blue curve below), the first channel is regulated down by an adjustable value. As with the compressor, the Ducker has the parameters Attack, Hold and Release. A typical application would be to automatically lower the signal level of a music signal during announcement on the source channel. A source channel can control multiple target channels and the target-channels can be influenced from multiple source-channels. The source signal itself on his channel passes the Ducker unchanged.

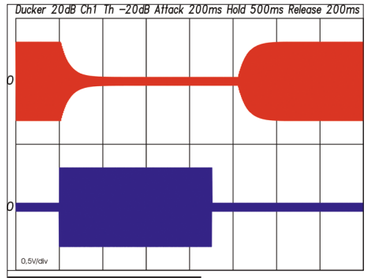
The Adaptive Gain behaves very similar, but in a reversed action for the target channel. If the signal level on the source channel exceeds the threshold value, the gain in the target channel is raised.

A possible application for Adaptive Gain is to capture the room noise level with a microphone connected to the controlling source-channel, so that the level of the signal in the target channel is automatically raised when the room noise is above a given threshold. A source



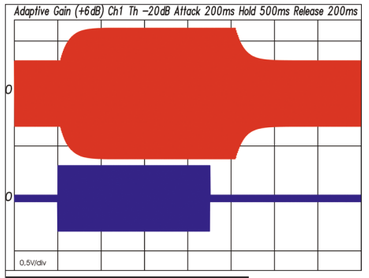
Function of the limiter

Figure 15: How the limiter works: Below the input signal (bl) with a base level of -30 dB and a level jump of +20 dB from 1 s to 3 s. The limiter reacts and then reduces the level with an attack time constant of 100 ms the threshold of 30 dB. After the end of the burst, the 500 ms hold time passes before the release takes place with a time constant of 200 ms.



How the ducker works

Figure 16: Ducker function: The ducker acts similarly to a compressor, with the difference that the control signal for the compressor comes from another channel, the source channel. Above the signal of the compressed target channel (rt) and below the control signal on the source channel (bl).



How the adaptive gain works

Figure 17: Adaptive gain function: The adaptive gain is comparable to the Ducker. Here, however, the target signal (rt) is not compressed, but amplified as soon as the source signal (bl) exceeds a threshold value.

channel can control multiple target-channels.

Class-D amplifiers and Everlast power supplies

Details about the power amplifiers and the special Everlast power supplies have already been explained in detail in the test report in the 1/2020 issue of PROFESSIONAL SYSTEM.

What follows is only a brief summary. The output stage channels are based on a special Class D driver module that controls two FETs in the power stage. The driver outputs are completely isolated from the rest of the module and can directly control power levels with a very high supply voltage. The structure is therefore simple and flexible. You need a modulator that generates the PWM signal from the audio signal, the aforementioned driver module and two more power FETs for the output stage. The voltage range you want to cover for the output can then be set via the supply voltage from the switched-mode power supply. To make the output signal of the class D output stage usable, a low-pass filter is required at the output, which acts as a reconstruction filter and filters out the RF components from the signal. The audio signal is then available again after the low-pass filter. The function is comparable to that of a DA converter only in this case for high output power. For the low-pass filter, Extron uses its own patented circuit, called CDRS, which suppresses RF components in the signal particularly well and reduces EMC exposed to the environment.

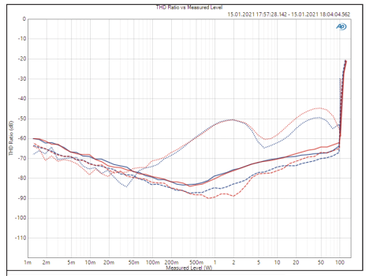
The power supplies for the Ultra models, like most other Extron devices and external power supplies, use an in-house development called Everlast. The name „Everlast" is an allusion to the topic of operational safety. Since the power supply units play a vital role in all electronic devices, a failure is usually not without profound consequences. For tough 24/7 use, Extron did not want to rely on the widespread standard power supplies, so they developed their own Everlast power supplies.

Measured values output stages

In the test series a year ago, the eight-channel models of the XPA Ultra series with 8x35 W and the two-channel models with 2x100 W were tested and measured. The version with 4x 100 W was not represented in the test field at the time. Therefore, two of the most important measurement results for the power amplifier are presented here, although the focus of this test is on the NetPA DSP module.

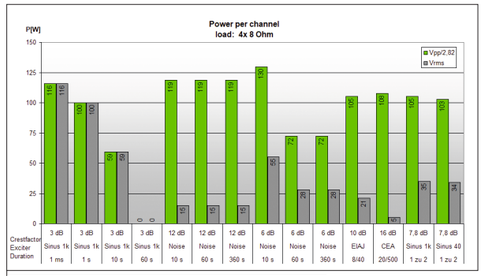
Figure 18 shows the distortion values as a function of the output power measured at frequencies of 100 Hz, 1 kHz, and 6.3 kHz for a 4x 8 Ω load on the output stage. The clip limit is independent of the Frequency exactly at 100 W. In summary, the measurement shows that the distortion increases depending on the frequency, so that values from -80 to -70 dB are achieved at 100 Hz and from -70 dB to at 1 kHz -65 dB. At 6.3 kHz, the THD values are sometimes -50 dB and a little higher. The specification from the data sheet with 0.1% (–60 dB) THD at 1 kHz 3 dB under full modulation (50 W) is exceeded by a few dB. The fluctuating and somewhat higher over the 6.3 kHz measurement is typical of class D amplifiers, likely caused by the PWM modulation.

The power measurement of the NetPA U 1004 ➔



Distortion values as a function of performance

Figure 18: Distortion values (THD + N) for channels 1 (bl) and 2 (rt) depending on the power (x-axis) for a load of 4x 8 Ω measured at 100 Hz (dashed), at 1 kHz (solid line) and 6.3 kHz (dotted).



Power values for different signal types

Figure 19: Power values of the NetPA U 1004 at 8 Ω per channel with simultaneous loading of all channels. Values for different types of signals.

was carried out for a load of 4x 4 Ω and 4x 8 Ω. Since the results of the power values were similar, we only show the once for 4x 8Ω in Figure 19. A 2 Ω operation is not intended.

Conclusion

With the NetPA models Extron is expanding its Ultra-amplifier series by a total of eight amplifiers. The two basic models NetPA U 1002 and NetPA U 1004 with 2x 100 W and 4x 100 W output power, are available in a Low-Z version or for 70V or 100V systems. Recently added are the two models NetPA U 2002 SB and NetPA U 8001 SUB with 2x 200 W and 1x 800 W for low-Z operation. The difference between the NetPA Ultra Series and the previous XPA Ultra Series is the addition of an integrated DSP and Dante connectivity. The compact devices are much more than just amplifiers. The NetPA Ultra has four analog inputs including microphone preamps and an additional four analog line-level outputs, Dante channels and Power amplifiers connected via a mix matrix and plenty of DSP functions. With this small 9.5 inch device you have all equipment needed for a small Public address system including networking to other Devices, rooms or control centers. With the already known advantages of the Ultra Series models, such as low power consumption, good protection circuits, passive cooling, and a completely closed and compact enclosure, the new NetPA Ultra Series amplifiers are ideal for all kinds of small and decentralized AV installations.

• [14260]

A Complete Audio DSP Platform with VoIP, AEC, Dante, and USB Audio

DMP 128 Plus Series

The **DMP 128 Plus Series** is the next generation of Digital Matrix Processors featuring Extron ProDSP™ 64-bit floating point technology. The DMP 128 Plus Series are equipped with 12 analog mic/line inputs, eight analog outputs, up to four channels of digital audio input and output via USB, up to eight audio file players, an ACP bus for audio control panels, and new configurable macros. DMP 128 Plus Series processors can be used anywhere from a credenza-based system to a large multi-rack system, and even in a large, complex, decentralized multi-building system.

Features:

- All models include a USB Audio interface, providing up to four channels of digital audio sends and returns
- Each of the VoIP lines can support generic Session Initiated Protocol – SIP connectivity, V models only
- Twelve channels of AEC - acoustic echo cancellation, C models only
- Dante audio networking provides a wide range of expansion capabilities, AT Models only
- Eight Aux inputs and outputs can be individually configured as an audio file player, USB Audio, or in V models, VoIP
- Extensive mix matrixing in every DMP 128 Plus allows all inputs to be discretely routed to any or all inputs
- Macros allow the sequencing of commands that can be sent to the local device or external devices via the LAN port

Audio Control Panels

The ACP Series of configurable audio control panels interface directly with the DMP 128 Plus audio processors to provide selectable volume and mute for mixing and zone control, plus preset or macro recall for room configuration.

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