SYSTEM CONTROLLER DIGITAL PA MASTER DSC 28

The GAE DIGITAL PA MASTER DSC 28 is a state-of-the-art digital System-Controller. Its role as integral system component, even at the development stage of a loudspeaker system, clearly sets it apart from other speaker management devices available on the market. By utilising the DSC 28, elementary system-parameters e.g. of a mid/high unit, can be decisively influenced during early development stages. As such the user-accessible presets represent the outcome of intensive development and determine, together with the loudspeaker components and the power amplifiers, the tonal and power handling behaviour of the sound system as a combined single unit in its different configurations.

This product philosophy has already been successfully realised by **GAE** in such products as the analog system-controller BF1 (programmable via interchangeable preset cards). Functions relevant to the system are stored as recallable presets and only the important user-definable variables are accessible and freely adjustable. Even in this digital-age of signal processing, the **totally user-programmable crossover** as central driving device for **GAE** loudspeaker systems, remains second choice.

Function. The GAE DSC 28 combines the functions of crossover, equaliser, delay and limiter in a 1U rack-space enclosure. A fully-parametric EQ, inserted before the crossover circuit enables an uncomplicated tuning to room acoustics. The controller has been conceived as a remote controllable, 2-channel drive unit for up to 4-way, high power sound system applications, with two analog inputs, a digital in-/output (AES/EBU) which can also be used as a digital insert and four analog outputs 1[SUB], 2[LOW], 3[MID], 4[HIGH] per channel.

Menu. Operation is menu-controlled and is achieved through use of a 2 x 24 character blue fluorescent display. Parameter input is by means of an increment-dial with additional tip-function (ENTER) as well as a group of eight tip-keys.

Remote. The remote-control interface (RS-232/MIDI) enables sound systems driven via one or more controller to be singularly or commonly controlled. A Computer can also be linked to the controller (or chain of controllers) allowing operation by means of a remote-control software. A software using the Microsoft Windows® and Microsoft® NT platforms is available.

Level-monitor. Each in- and output has its own LED bargraph indicating level-, mute-, clip- and limit- status, scaled in dBu for the input-level, in dBFS (Full Scale) for the output-level and in dBGR (Gain Reduction) for the output-limit-indicator.

Input-dynamic. Without the necessity for analog level matching and with the help of dual-range conversion and a combination of analog pre-emphasis / digital de-emphasis, the GAE DSC 28 achieves an input dynamic range of 130 dB with 28 dBu max. input level (@ < 1kHz).

Output-dynamic. Each output of the DSC 28 can be individually matched to the input sensitivity of the amplifier to be driven in four-stage maximum output levels. State-of-the-art, 24-Bit DA-converters of the Delta Sigma-type, are used. The resulting dynamic range is an unusual ≥114dB!

Noise, dynamics and headroom. The output noise-level of the DSC28 is typically –96dBu (unweighted, 22Hz – 22kHz) at the maximum output level of 18dBu. Maximum output level reduction due to amplifier matching reduces the noise level accordingly, at the same time enabling a higher resolution on the digital side. As a result the DSC28 commands dynamic values superior to those of an analog device which is quite unusual for a digital device. Important signal processing stages operate with 48-Bit accuracy. In cases where a reduction to 24-Bit is made, a noise-shaper with 1st Order Feedback is introduced so as to ensure the lowest possible noise level and distortion especially by low-level input signals.

Peak-limiter. The improved possibilities made available to limiting through digital technology are fully utilised by the DSC 28. An “in-advance” signal-analysis of the output signal allows pre-determined threshold excesses to be detected and the level to be decreased over a determined time factor with a matched time constant. The peak-value is then reduced to the exact level of the threshold setting. Even sudden transients can no longer cause audible distortion and thus maintain that the limiter threshold can be safely set to the exact limits of the power amplifier and/or the attached loudspeaker systems. Additionally the pre-masking effect can be used for strong transient impulses, without noticeable processing time before the occurrence of the impulse. The following hold-time prevents level modulation in the directly following signal passages. The “pre-viewing” signal-analysis also allows an increased efficiency in the filtering of the control signal without degrading the attack of the limiters. In this way the high-frequency distortion of the wanted signal by the control signal is eliminated.

RMS-limiter. The RMS-limiters of the DSC 28 protect the speaker components by imitating the thermal time constants of the voice-coil and magnet materials.

Delay. A master-delay for adjusting the fundamental delay-time of the unit is situated before the filter-networks whilst a further delay is situated in each of the output-paths for time-alignment of the individual loudspeaker components, which is an integral part of the correction filter. The intrinsic delay of the DSC 28 is 5...7ms depending on the path, and is effected by AD/DA-conversion, down-/oversampling and the "pre-viewing" limiter concept. All further delays are effected by signal processing in filters and are especially dependant on the group-delay behaviour of the loudspeaker system, as well as on filter slopes and cut-off frequencies.

Filter. The DSC28 calculates crossovers and equalisers as FIR-filters (Finite Impulse Response). This type of filter requires a much greater computing power than that required by digital devices which purely copy analog filters in the digital domain and employing IIR-filtering (Infinite Impulse Response). Computing power has been optimised by the use of matched down- / oversampling in each of the paths (frequency ranges) with the exception of the HIGH-path. However, this entails a bandwidth restriction for the SUB-, LOW- and MID-paths. FIR-filtering allows the realisation of system-specific correction filters with linear-phase behaviour resulting in equal delay-times to all frequency portions of a signal. Unfortunately linear-phase response, down to the lowest frequencies of the transmission range, will present too long a signal delay, unacceptable for live situations. This can be countered by the gradual fading over to minimal-phase ("analog") response below a specific frequency, which will defuse the problem. For the filtering of an active system with moderate crossover slopes (≤ 36dB/Octave), minimal-phase response up to 120Hz and with linear-phase response above 120kHz, the over-all controller delay will be approximately 30ms: basic controller delay of approx. 7ms + filter delay of approx. 23ms. Due to its conception with FIR-filter technology, the development of the GAE DSC28 as a programmable crossover with the subsequent copying of analog filters, adjustable at the user-interface, is not intended.

Equaliser. Situated prior to the cross-over network, the user is provided with a fully-parametric equaliser with 14 bands per channel. This equaliser is based in IIR-filtering and is programmable at the user-interface. The following filter-types can be selected: Bell, Low-Shelving, High-Shelving, Low-Pass and High-Pass.

Conclusion. The DSC28 is a system controller of non-compromising conception incorporating a standardised user-interface allowing for choice of set-up and the control of user-parameters. These set-ups are pre-determined and made available as relevant operation-parameters by the system manufacturer. As such the DSC28 is ideally suited for **OEM application**. Developed for use with GAE loudspeaker systems the DSC28 can, on request, be programmed to provide the necessary parameters for driving all types and makes of loudspeaker systems.

The fully programmable parametric equaliser completely replaces an additional 19"-device and enables the uncomplicated room tuning of an already optimally equalised loudspeaker system.

The concept behind the GAE DSC28 allows not only for better efficiency in design and realisation in future system-developments in loudspeaker technology. Even today the employment of the DSC28 with its FIR filter technology offers astounding advantages when driving existing systems.

The audio quality of the signal processing, determined mainly by the choice of AD/DA converters, but also influenced by the analogue components in the circuitry, is engineered to the maximum possible with regards to the present technical possibilities within the bounds of an acceptable price/performance ratio.

The **GAE DIGITAL PA MASTER DSC 28** offers state-of-the-art solutions not only to future sound reinforcement applications but also meets the demands for the driving of today's sound reproduction systems.

Block diagram and device description.

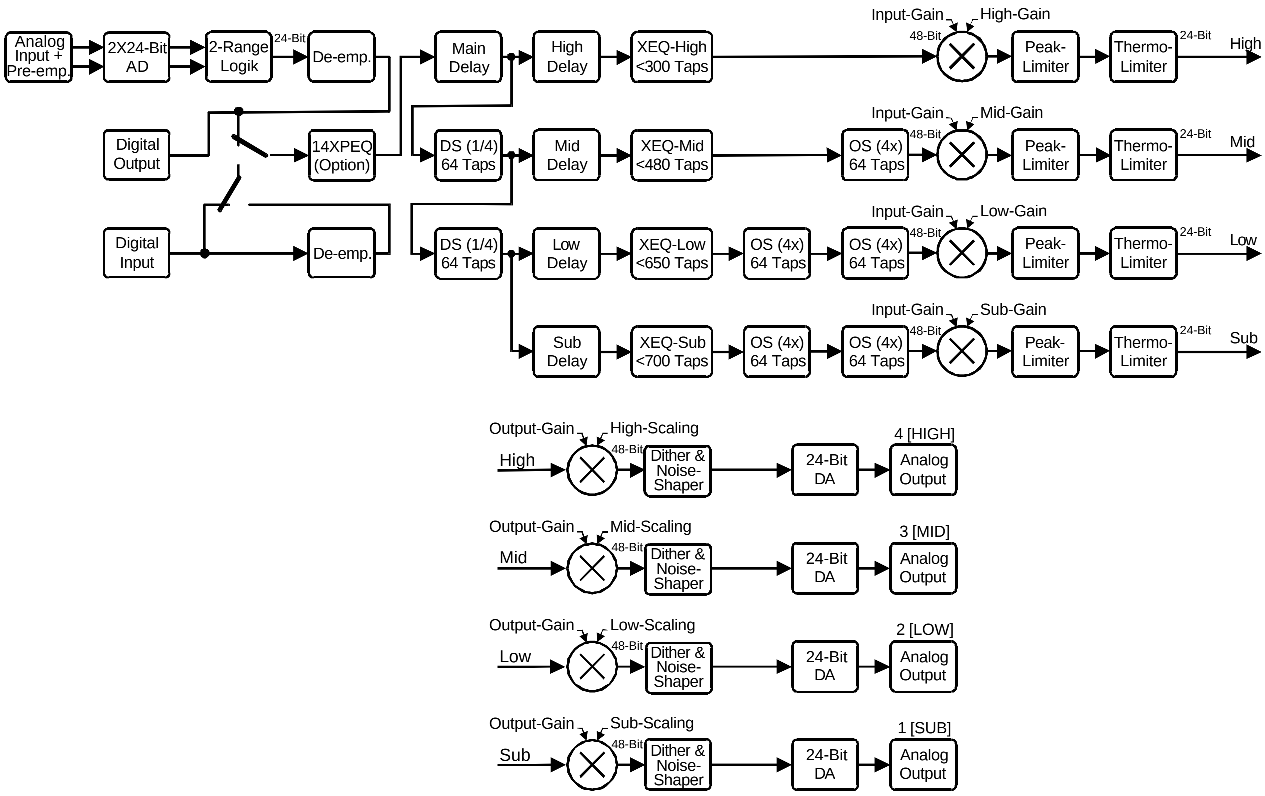

Fig. 1 The block diagram exemplary shows the complete signal flow of one channel of the DSC28.

Analog-Input. After passing through an analog pre-emphasis, the analog input leads to two separate amplifier stages with different amplification rates, which drive both inputs of a 24-Bit Stereo AD-converter.

Dual-Range-Conversion. The switching and matching of both AD-channels is performed on the digital side in the sound processing area to complete the dual-range-conversion. Here the analog input-signal is reproduced in 24-Bit resolution. Following this the De-emphasis filter is calculated and the digital signal is available for further processing. By this means a dynamic range of 127dB is obtained which greatly overshadows even the newest generation of 24-bit single-converters.

Digital In/Output. At the digital output the 24-Bit-AES/EBU formatted, AD-converted data is continually available. Alternatively to the analog inputs the audio signal can be introduced directly via the AES/EBU-digital input where a de-emphasis filter is introduced should this be required by the accompanying data of the digital signal. In combination with the digital output, this input can function as a digital insert. A device connected to the Insert should however be capable of true 24-Bit calculation otherwise a degradation of the otherwise excellent values of the dual-range-converters could be noticed.

Parametric Equaliser (PEQ). A further signal-processor is required for the following parametric equaliser with 2x 14 filters. Integrated in this module is the function EQ-Gain for the reduction of the input signal when balancing boosting filters as well as a limiter stage at the PEQ output to prevent digital overflow. Also integrated is a Noise-Shaper which reduces quantization error when re-quantizing from the full-48-bit operating EQ back to 24-Bit.

Downsampling. Of the following four signal-paths only the designated high-frequency path requires the maximum bandwidth for data-processing, allowing the sampling-rate of the remaining three paths to be reduced by down-sampling by the factors 4 (MID) and 16 (SUB and LOW). Keeping the number of filter-coefficients constant, the reduction of the sampling rate enables the increase of the length of the filter by the same factor and at the same time, dramatically reduces the amount of computing power necessary.
Down-sampling is carried out by means of 2 FIR low-pass filters with 64 taps. After low-pass filtering, the signal with the reduced sampling-rate can be subjected to further processing. For this application the filter-characteristic has been contrived for maximum rejection-band-damping (>120dB) so as to prevent disturbances due to aliasing effects.

Delays. A Master-Delay is situated in front of the filter-network and four further delays, one per way adjust the time-alignment of the loudspeaker components. A master-delay for adjusting the fundamental delay-time of the unit is situated before the filter-networks and a further delay is situated in each of the ways for time-alignment of the individual loudspeaker components. The minimum intrinsic run-time of the DSC 28 is 5ms effected by AD/DA-conversion, down-/oversampling and the “pre-viewing” limiter concept. All further delays are effected by signal processing in filters and are especially dependant on the group-delay behaviour of the loudspeaker system, on filter slopes and cut-off frequencies. The total intrinsic run-time of the controller is shown for each loaded preset in the Master-Delay menu.

Crossover- and equalising-filter (XEQ The signal is now routed to the actual crossover and equalising filters for the individual paths (SUB, LOW, MID, HIGH). Two Motorola® DSP56009/81-Processors for the XEQ-Filter of both channels of the controller provide a calculating power of approx. 80Mips which is optimally divided between the single paths. The table below shows the resulting data, taking into account the sampling-rate assigned to each of the paths:

Way-No.	Way-Name	Down -sampling Factor	Filter length Max. Taps	Filter length Taps (eff.)	Sample-Freq. KHz	Cut-off-freq. (approx.) kHz	Freq.-resolution, Hz
1	SUB	16	700	11 200	2.756	1	3.9
2	LOW	16	700	11200	2.756	1	3.9
3	MID	4	480	1 920	11.025	4	23
4	HIGH	1	300	300	44.100	20	147

Oversampler. After XEQ-filtering the sampling rate can be returned to it's original value by means of one, respectively two, 4x over-samplers. This process takes place in the reversed order to down-sampling. For the paths LOW and SUB two 4x over-samplers are connected one after the other. Analog to the down-samplers, the low-pass filters are equipped with 64 taps and are optimised for maximum rejection-band-damping.

Characteristic frequency. Ripples within the transmission-range of the down- and over-sampling filters, as well as the amplitude- and phase-response of the controller’s analog and converter modules are taken into consideration and are inversely re-compensated during the generation of the XEQ-dependant filter-coefficients.

Input-Gain. The following function Input-Gain has an extensive level-operating-range of -83...+45dB and as such can be used as a volume control. Especially when using the analog inputs this function has the task of matching the input source signal to the PA system connected to the outputs of the controller.

Output-Gain (Paths). Next, in the signal processing chain is the gain-function for the individual paths equipped with a level-operating-range of -18...+6dB. Based on the 0dB-position this function enables the fine (personal preference) matching of the individual, actively driven speaker components in a PA system, especially in the low frequency region. Sensitivity differences in the sound pressure level range of the individual components of a loudspeaker system will already have been accounted for and corrected by means of inverse scaling during the calculation and setting of the XEQ filter parameters.

Limiter-System. Two limiter functions per path are next in line. By utilising a short delay of 1.5ms for the peak-limiter it is possible for the device to limit to foreseeable peak values. This is complimented by the combination with a thermal-limiter which protects the loudspeaker components from thermal overload. The load characteristics for each driver together with the capacity characteristics of the systems power amplifiers are entered into the parameter-sets of the XEQs determining the optimal intervention of the controllers protection circuitry.

Multi-Path. The so-called Multi-Path variant of the controller is not shown in the block diagram. This variation allows the summing together of two or maximum three controller ways after the limiters which are then presented at a common output. This feature enables the creation of correctional filters even for loudspeaker systems containing passive crossovers networks. The XEQs and the limiters for the frequency ranges for each of the passively separated drivers is individually shaped, the processed ways are then summed together and subjected to a further peak-limiter and then made available at a single output. The possibilities of the Multi-Path variant and the predetermined outputs are as follows:
LOW + MID → 3 [MID]
MID + HIGH → 4 [HIGH]
LOW + MID + HIGH → 4 [HIGH]

Presets. A preset consists of a set of system parameters predetermined by the loudspeaker system manufacturer which is selected as a whole and activated for the system via the controller. A preset contains the following parameters:

- The system equalisation in magnitude and phase taking the sound pressure and sound power measurements of the loudspeaker components and the inherent frequency of the controller into consideration. The equalisation is carried out on the basis of a target function (eg. linear from freq. X to freq. Y);
- the bandpass-structure considering the crossover behaviour of the individual ways including the dispersion behaviour in the crossover region as well as the power handling characteristics of the loudspeaker components;
- the time-alignment between the loudspeaker components;

- the limiter-thresholds in relation to the characteristics of the connected loudspeaker components and power amplifiers;
- the output-scaling-factors, which guarantees the optimal balance between the amplification of the analog output stages of the DSC28, the amplification factors of the connected power amplifiers, the limiter thresholds and the highest possible digital resolution in the signal processing.

Output-Gain. Before being passed to the DA-converter, the signal can, if necessary be reduced in common-level (all four paths) by the output-gain within a range of 0...-24dB. This function operates like a volume-regulator, but can be better interpreted as a 'Safe'-regulator, as it's location after the limiters enables the setting of the maximum possible output-power of the PA-system.

DA-Converter. A State-of-the-art, 24-bit Stereo, DA-converter of the Delta-Sigma-type is incorporated into each path and switched in parallel. The resulting dynamic range is ≥114dB. A 2nd-order low-pass provides the reconstruction of the digital data-flow of over-sampled converter-output signals.

Analog-Output. The output-amplifiers of the DSC28 are electronically balanced and rated low impedance. Connection-wise they should be regarded as a transformer-output, i.e. the output-signal only flows between the two active output poles. At the maximum output-level of 18dBu the outputs can be loaded with ≥ 350Ω. The resulting dynamic value of ≥114dB is remarkable for a digital device. The matching to varying amplification-factors of power amplifiers is realised in four stages of maximum possible output-level (18, 12, 6 and 0dBu) and is finely adjusted by the XEQ filter. This enables the retention of the excellent dynamic-values and especially the high resolution of the digital signal at the input to the DA converters even when driving power amplifiers with high amplification factors.

Parameter set construction.

Parameter set construction. The Digital PA Master DSC28 together with the measurement system MF from the ITA (Institute for Technical Acoustics) at the RWTH (Rhein-Westfälische Technical University) Aachen, Germany, which also includes the parameter set construction software, provides the system designer with a universal tool for the construction of XEQ-filters and limiter functions and as such, is an integral part of the development of a loudspeaker system. The calculated parameter sets of the XEQ-filters can then be written in the Flash-RAM of the controller via the RS-232 or MIDI interface and the result can immediately be checked by means of measurement and audio assessment. After completing the system design it is possible to load several presets (parameter sets) e.g. for different stacking variations of loudspeakers, to the Flash-RAM and enabling the user to switch between set-ups depending on system configuration. When necessary, further presets can be loaded to the controller via the RS-232 interface.

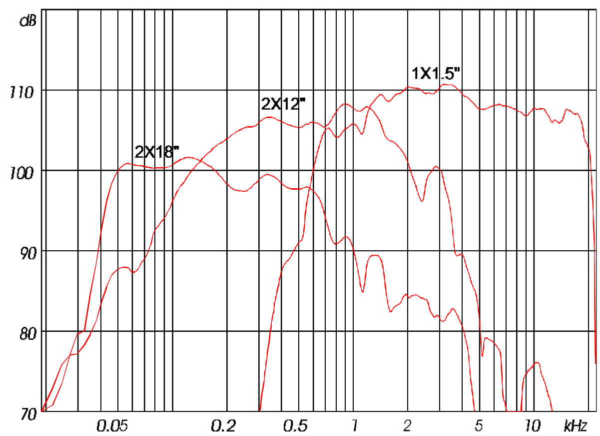

Fig. 2 Loudspeaker system, Individual component measurement, sensitivity @ 1W, 1m

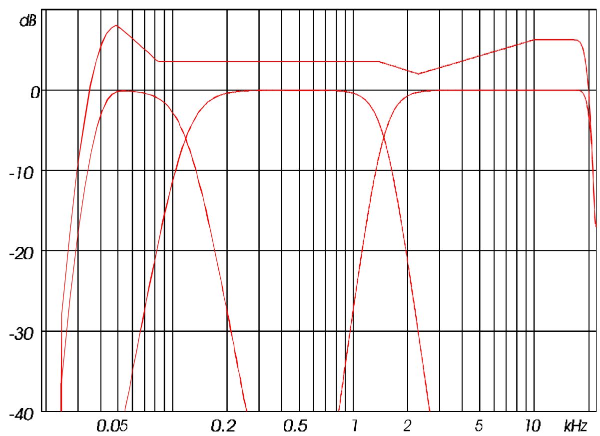

Fig. 3 Target function, system (weighted) and band-passes

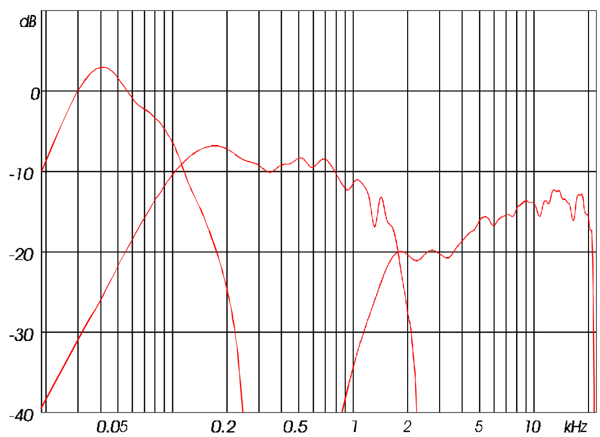

Fig. 4 Controller-outputs, level

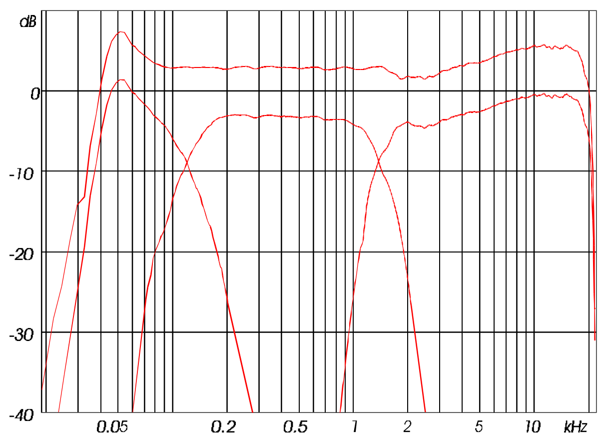

Fig. 5 Equalised system, total frequency response, individual frequency responses

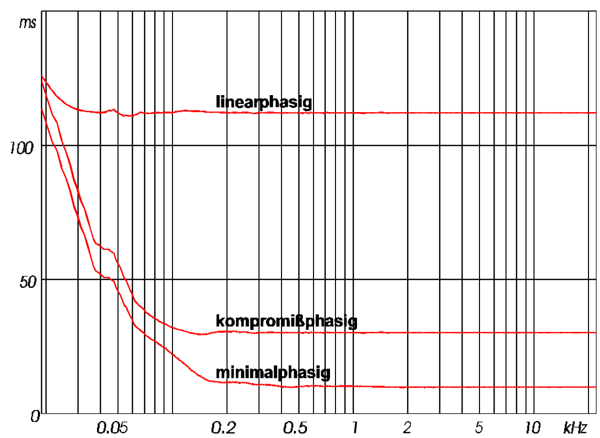

Fig. 6 Equalised system, three examples of overall time-delay

In a minimum phase system the delay time is considerably increased towards the lower frequency range – as is usual in analog controlled systems. Should an equalisation of a system be carried out to give linear phase behaviour then the delay time for the higher frequencies must be increased to that of the maximum delay-time produced at the lowest frequencies. The resulting signal delay is unacceptable for live-performance situations. The compromised equalisation allows an extended linear phase behaviour in the lower frequency area which is transferred to minimum phase behaviour upwards of approx. 100Hz.

OEM-application, procedure and costs.

At this point the interested and potential user asks himself the following question: **How can the outstanding audio-response, the “pre-viewing” limiter function and the FIR-Filter of the DSC28 be used to optimise his own loud-speaker system? What expenditure can be expected?**

The programming of presets involves classical development procedures the duration of which are entirely dependant on the complexity of the system to be equalised and the precision to which this is carried out. This time factor determines the framework for cost calculation. For this reason it is not possible to provide a standard information with regards to expense. Our experience in analysis and realisation of preset generation however, enable us to provide individual solutions to meet the required demands. A detailed written enquiry with an exact description of the system that is to be programmed should form the basis for the requested quotation.

To enable a pre-estimation of the costs a copy of an example calculation can be provided on request.

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Technical Specifications.

Analog inputs.............	electronically balanced, input impedance 20kΩ maximum input level: 28dBu @ < 1kHz, 18dBu @ 20kHz noisefloor: -100dBu; dynamic range 127dB (all measurements linear rated 22Hz - 22kHz)
Converting..................	per channel a Dual-Range-24-Bit Delta-Sigma AD-Converter with pre-emphasis, 50/15µs; sample rate: 44,1kHz
Analog outputs...........	electronically balanced, output impedance 20Ω maximum output level: +18dBu into 375Ω; noisefloor: ≤ -96dBu; dynamic range ≥ 114dB (all measures linear rated 22Hz - 22kHz) totally THD+N: < 0,005% at operating limit output-range-steps: 18, 12, and 6dBu maximum output level
Converting..................	per output a 24-Bit Stereo-Delta-Sigma converter in parallel mode
Digital in/out..............	input: 24-Bit AES/EBU with/without pre-emphasis, sample rate converter 32kHz...96kHz output: 24-Bit AES/EBU, sample rate: 44.1kHz also useable as digital insert (AD⇒DigOut ; DigIn⇒Controller)
Remote control...........	input: RS-232 and MIDI, output: MIDI; baud rate: 9600 and 31250
Intrinsic run-time.........	incl. down-/over-sampling, AD-/DA-converter and limiter: 5...7ms, path dependant
XEQ-filter....................	FIR-filter, group delay dependant on chosen presets
PEQ-filter....................	IIR-filter (quasi analog behaviour)
Limiter........................	<u>peak-limiter</u> with 1.5ms pre-view and controlled-overshoot to fully utilise the PA-impulse reserves, 48-Bit signal processing, precision-adjustment and extremely low distortion <u>rms-limiter</u> with voice-coil- and magnet temperature-modelling
Operating...................	foil keyboard with eight tip-keys, incremental-dial with additional ENTER-function
VF-display..................	2 lines, 24-character-vacuum-fluorescent-display, blue
LED-display................	2 X input (-30 ...+24dBu) + 2 X clip (≥ +28dBu) each 10 X LED green + each 1 X LED red 8 X output (-30 ... 0dBFS + 0 ...12dBGR) + 8 X mute each 7 X LED green + 3 X red + each 1 X LED rot
Power supply.............	universal-input 85...265V, < 30VA, over-voltage-protection, safety fuse M1A, M = medium, switched-power-supply is self-protected.
Dimensions................	19“ / 1U, 260mm (10.25“) depth
Weight........................	3.75kg (net without packaging)

GAE

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