

# K&F PLM 20K44

- 4 channel power amplifier platform for all the K&F speaker systems
- 'Rational Power Management' (RPM) for flexible power distribution
- Lake® platform loudspeaker management and system control
- Scaleable and combinable with additional K&F PLM Series and K&F D Series SystemAmps
- Worldwide useable due to the 'Power Factor Correction' power supply
- AES3, analogue and redundant Dante™ Inputs/Outputs

The K&F PLM Series SystemAmps provides exceptional performance and expanded flexibility in high-power audio amplification for challenging live sound systems applications and other large or particularly demanding applications. Based on the approved, road-tested and green amplifier technologies of Lab.gruppen's renowned PLM Series, the live-sound-dedicated PLM Series adds Rational Power Management (RPM™) – a new proprietary Lab.gruppen technology that rationalises power allocation and potentially reduces amplifier inventory. The Lake-variant PLM Series models benefit from the approved package of onboard Lake Processing and Dante™ digital audio networking, and also offer integration potential with K&F specific loudspeaker presets and many 3rd party matrix and proprietary DSP systems via dedicated middleware. Equipment specification, commissioning (including configuring RPM and other unique amplifier technologies) and on-going control and system monitoring are managed via the innovative CAFÉ™ software, running on Mac or PC. PLM Series features include redundant audio inputs as well as on board surveillance and load monitoring to fulfill the requirements of missioncritical voice evacuation compliance.



## K&F PLM 20K44

Processing / Network	Lake / Dante
Numbers of channels	4
Total burst power all channels (with RPM)	20000 W
<b>Max. Output Power (all channels driven) <sup>1)</sup></b>	
2 ohms	4400 W
2.67 ohms	5000 W
4 ohms	4400 W
8 ohms	2300 W
16 ohms	1150 W
Hi-Z 70 V	3300 W
Hi-Z 100 V	4700 W
<b>Max output power single channel <sup>1)</sup></b>	
2 ohms	4400 W
2.67 ohms	5900 W
4 ohms	4600 W
8 ohms	2300 W
16 ohms	1150 W
Hi-Z 70 V	3300 W
Hi-Z 100 V	4700 W
<b>Amplifier output module</b>	
Peak output voltage	194 V
Max output current	67 A
<b>Rational Power Management (RPM)</b>	Regardless of model, any channel has potential to deliver the max single channel output power
Default voltage limitation (can be lifted with RPM)	194 V
<b>Protection features</b>	Current Average Limiter (CAL), Very High Frequency Protection (VHF), Direct Current Protection (DC), Short Circuit Protection, Current-Clip Limiter, Voltage Clip Limiter, Temperature protection
<b>Mains Power</b>	
Nominal voltage	100 - 240 V AC 50 - 60 Hz
Operating voltage	70 - 265 V AC 45 - 66 Hz
<b>Power supply features</b>	
Soft start / Inrush power	Yes / Max 8 A
Power factor correction	> 0.98 for mains power > 400 W
Breaker Emulation Limiter (BEL)	Configurable current threshold and breaker profile
BEL max current threshold	32 A
Regulated switch mode power supply (R.SMPS), Power Average Limiter (PAL), Under Voltage Limiter (UVL), Mains under-/overvoltage prot., mains glitch tolerance	
<b>Dimensions and weight</b>	
Rack rail to rear panel	W: 483 mm, H: 88 mm (2 U), D: 424 mm
Overall all depth front-rear support	D: 463 mm
Weight	17 kg (37 lbs)

<sup>1)</sup> Lab.gruppen burst power (1 kHz, 25 ms burst power @ 150 BPM, 12 dB Crest factor)

<b>Amplifier platform</b>	
Inter Sample Voltage Peak Limiter (ISVPL)	Configurable Peak voltage threshold and profile
Amplifier gain	Digital configurable amplifier gain 22 - 44 dB
Pilot tone generation and analysis	LoadPilot
Load impedance analysis	Yes
Temperature control	Regulated fans and show must go on limitation (ATL, PTL)
<b>Audio Performance (Amplifier platform with digital input)</b>	
THD + N 20 Hz - 20 kHz for 1 W	< 0.05 %
THD + N at 1 kHz and 1 dB below clipping	< 0.04 %
Dynamic range	> 114 dB
Channel separation (Crosstalk) at 1 kHz	> 70 dB
Frequency response (1 W into 8 ohm, 20 Hz - 20 kHz)	+/- 0.05 dB
Internal sample rate / Data path	48 / 96 kHz / 32 bit floating point
Product propagation delay AES 96 kHz / analog input	1.61 / 1.68 ms
<b>Lake processing</b>	
Loudspeaker processing	Up to 4 modules of Classic/linear-phase/FIR crossover, EQ, delay, LimiterMax™ - peak and RMS limiters
System tuning	Group control with Raised Cosine™ MESA EQ™ asymmetric filters
Input redundancy / Matrix	Automatic 4 level input redundancy / 4 input mixers
System integration	Comprehensive 3rd party protocol over UDP Ethernet
<b>Dante Audio Network</b>	
Dante I/O	8 x 8
Network topology / redundancy	Flexible topology / Supports daisy-chained and Dual redundant networks
Sample rates / transport	48, 96 kHz / Uni + Multicast
Network latency	0.25, 0.5, 1.0, 2.0, 5.0 ms
<b>Analog inputs</b>	
Inputs	4 high quality inputs with Iso-Float ground isolation
Maximum input / digital reference	+ 26 dBu / +21 dBu
Sampling rate / resolution	96 kHz / 24 bit
Input impedance balanced / unbalanced	20 / 10 kOhm
THD + N (typical at 1 kHz unweighted)	0.00022 %
THD + N (typical at 20 Hz and 20 kHz unweighted)	0.00033 %
<b>AES Inputs</b>	
Inputs	2 AES inputs (4 audio channels)
Supported sample rates / resolution	44.1, 48, 88.2, 96, 176.4, 192 kHz / up to 24 bit
Sample rate conv. THD + N 20 Hz - 20 kHz unweighted	0.00003 %
<b>Back panel interface</b>	
Analog inputs	4 x 3-pin XLR, electronically balanced
AES inputs	2 x 3-pin XLR
Output connectors	Neutrik speakON (1 x NLT8, 2 x NLT4) or 4 Binding Posts (pairs)
Ethernet ports	2 x EtherCon RJ45 100/1000 Base-T for Lake and Dante controller and/or DLM (3rd party protocol)
Detachable mains cord	Neutrik PowerCon rated at 250 V / 32 A
<b>Front panel user interface</b>	
Display	2.5 inch, black / white, daylight readable LCD
Fault/Warning/Limit/Clip indicators	RGB LEDs and detailed fault description on display
Mute and soft function buttons	8 provided
Standby Power button	On/Standby
Mute Enable button	Enables muting of outputs and inputs via soft-button keypad
Meter button	Toggles through meter views
Menu button	Provides a menu driven interface for full function front panel control
Rotary Encoder	Yes
Exit Button	Provides a "back" function
Approvals	CE, ETL (ANSI/UL, CSA), PSE, RCM

# K&F PLM Series: Technology Overview



The PLM Series from Lab.gruppen offers an unprecedented combination of sustained high output, impeccable sonic performance, configuration flexibility, and real-world efficiencies for reduced installation and operating costs. PLM Series brings the world's most innovative, capable and proven amplifier technology to virtually any high specification live sound project, regardless of preferred DSP platform or specific matrix components.

## Proven Lab.gruppen Technologies

Reliability and durability remain the bedrock criteria for any amplifier, and in this regard the PLM Series rigorously maintains Lab.gruppen's industry-leading reputation. The amplifier output stages are the Lab.gruppen patented Class TD<sup>®</sup> which couples the efficiency of Class D topologies to the

sonic purity of Class B designs. Equipped with the Intercooler cooling system, PLM Series amplifiers dissipate heat more effectively and eliminate "end of tunnel" output device over-temperature problems. PLM Series also offers a full suite of protection features, including thermal "show-must-go-on" limiting, short circuit protection, excessive average current limiting, sustained VHF (very high frequencies) protection, DC protection and voltage and current-clip limiting. None of the limiters introduce slow, long term gain changes that can risk altering the balance of a tuned system. A Breaker Emulation Limiter (BEL) prevents power interruption while Under-Voltage Limiting (UVL) allows continued operation despite severe voltage drops.

## Rational Power Management (RPM)

At the core of the PLM Series platform is Rational Power Management (RPM), a proprietary Lab.gruppen technology that gives system designers and technicians unprecedented freedom to allocate the output power available on each channel for optimum performance with specific load conditions. RPM technology also enables the technician to minimize initial equipment costs, reduce rack space requirement and improve long-term energy efficiency – all without compromising sonic performance.

With conventional amplifiers, it is often necessary to "over-specify" amplifiers to meet the maximum power demand on one channel, leaving excess power capacity wasted on the remaining channels. RPM reduces costly excess capacity by allowing re-allocation of output power capacity among the four channels. RPM can be configured so that any channel can supply up to 4.600 W at 4 ohms regardless of power model. With RPM in the PLM Series, the maximum output channel(s) can be used for power-hungry low-frequency

systems while the remaining output power can be allocated as needed for the mid-frequency and high-frequency drivers, or for less demanding zones within a typical large project – such as concessions, concourses, VIP suites and function rooms within a sports arena or stadium.

From within the CAFÉ software, RPM allows the desired power demand to be specified for the different loads in several different ways. RPM then analyses the desired power in relation to different channel and device constraints. If all desired power levels are within constraints, RPM safeguards the balance and assures that the specified output power will be maintained regardless of demand of on other channels. If a particular zone's input is being driven beyond the specified power levels, RPM aids in limiting that zone to make sure the power is available for other zones. If the desired total power is in excess of what the power model can deliver, RPM can facilitate that the limitation is shared equally among the channels.

## CAFÉ and RPM for Green Credentials

PLM Series is configured and monitored using Lab.gruppen's CAFÉ (Configuring Amplifiers For the Environment) software suite. In addition to providing comprehensive system surveillance and configuration of RPM and other amplifier features such as ISVPL and Breaker Emulation Limiter (BEL), CAFÉ also includes valuable help to save the environment. In combination with the RPM configuration CAFÉ can accurately predict, based on the true SPL and speaker requirements of the individual loads for the given project, estimations of average mains current draw and generated heat in BTU. With PLM Series' innovative power supply technologies (true Power Factor

Correction utilizing Current Draw Modelling) the required mains draw is already best in class in relation to burst power output, but in combination with the BEL the mains draw can also be safeguarded to the predicted level. The end result is precise mains management and thermal control, which allows more accurate (rather than overspecified) provision of mains distribution, cabling and cooling. This technology suite not only saves on installation costs, it also reduces lifetime running costs and minimizes environmental impact. It also reduces demands on UPS systems in "mission critical" voice evacuation systems in arenas and stadia.

## CAFÉ and Equipment Specification Predictor (ESP)

CAFÉ also features an innovative design aid – the Equipment Specification Predictor (ESP). ESP examines the system SPL and speaker requirements for a given project and aids in transforming that data into circuit and amplifier channel

requirements. On a system level, ESP supplies a recommendation for optimized placement of channels into amplifiers for the most cost effective solution. The recommendation includes model and quantities of PLM Series required with most rational use of amplifiers, minimizing wasted headroom.

## Lake Processing

PLM Series Lake versions provide extraordinary input flexibility, the legendary power of exclusive Lake processing algorithms, comprehensive control and load monitoring via Lake Controller, and seamless integration into Dante digital audio networks. In addition, by employing third-party middleware, PLM Series Lake versions can be integrated into on other widely used networked digital matrix systems.

All three PLM Series Lake models incorporate four full-featured Lake Processing modules, with four discrete channels of audio throughput input to output. Audio signals are selectable from four channels of analog (with Iso-float ground isolation), four channels via AES3 digital inputs and eight dual redundant Dante networked digital inputs. Input signals are individually selectable for each channel, with programmable failover to a lower prioritized input. The

full-featured, on-board Lake processor includes group control with Raised Cosine MESA EQ asymmetric filters to match the responses of many loudspeaker systems. LimiterMax peak and RMS limiters set the industry standard for loudspeaker protection and sonic transparency.

The included Lake Controller software provides a unified interface for control of Lake functions and for comprehensive monitoring of both amplifier status and connected loudspeaker loads. Optimized for a wireless tablet PC, Lake Controller is easy and intuitive to operate, with the “feel” of real-time analog faders and controls. Lake Controller also features seamless integration with third party, realtime sound system measurement, optimization, and control software packages. Users can measure spectrum and transfer function and adjust system EQ at the same time, using the same user interface.

## Front Panel

The front panel provides controls for power STANDBY/ON and for amplifier channel mute as well as bidirectional select functionality between device and software. It also includes

a display which shows important status information for the PLM Series Lake platform unit as well as for each individual output channel.

## System Block

The input section (inputs, input router and input mixer) allows for mixing capabilities as well as redundant and prioritized inputs with automatic switch-over in case of signal failure.

Up to four Lake Processing modules provide user EQ and loudspeaker processing, including LimiterMax limiting. Each power output channel provides individual channel processing, including ISVPL limiter, RPM and load monitoring.