SIAP
Acoustic Systems

SIAP | MKV SPECIFICATIONS & KEY FEATURES

sounds like siap

2018

SIAP | Technical specifications

Mk V Processor setup

- Processor : 32 bit or 64 bit, floating point
- Processor power : 113,4 GFLOPS
- Memory : 24 Gigabyte
- A/D and D/A conversion : 24 bits
- Sampling rate : 48 kHz up to 192kHz
- Frequency response : 20 - 20,000 Hz ± 0,5 dB
- Phase linearity : better than 5º, 20 - 20,000 Hz
- Cross Talk : better than 90 dB, 20 - 20,000 Hz
- THD + N : smaller than 0.01%, 20 - 20,000 Hz
- SMPTE IMD : smaller than 0,01% 20 - 20,000 Hz
- Signal-to-Noise Ratio : better than 113 dB, input to output
- Time Delay AD/DA : max. 2 milliseconds, with processing
- Inputs and Outputs : digital & analogue
- Inputs : up to 192 (stacked up to 384)
- Outputs : up to 192 (stacked up to 384)
- Connectors : MADI, BNC, Ethernet, USB
- Input Sensitivity : +4dBV to -17dBV in 1dB steps, plus –20dB @ Mic
- Output sensitivity : +4dBV to -123dBV in 1dB steps
- Microphone power : 48 Volt phantom power, selectable
- Communication : Ethernet(TCP/IP, Dante™ Audio Networking) Optical (MADI), BNC (MADI)
- Dimensions : 19" mount frame, 2 HU
- Power Requirements
  - Input range : 90(85)-135 / 180-265 VAC
  - Frequency : 47-63 Hz
- Environmental Requirements
  - Temperature : 0-35ºC
  - Relative air humidity : 20-95%
- Approvals : CE and UL

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- o up to 384 independent channels (if stacked, 768 channels)
- o each channel can generate a dedicated reverberation pattern by FIR filters (128 in total)
- o early and late(diffuse) reflections can be used and modified independent
- o for each reflection pattern the following characteristics can be modified independently:
- o length of early and diffuse reflection time (up to 6,5 seconds)
- o level of the reflections
- o density of the reflections (up to 20000 per second per signal path = 2,560,000
- \- reflections per second @ 48 kHz sample rate; up to 10,200,000
- \- reflections per second at 192 kHz sample rate)
- o frequency characteristics in octave bands from 31.5 to 16,000 Hz
- o double or multiple sloped decays can be generated
- o independent dedicated acoustic zones for under balcony or stage areas with own reverberation times, envelopment and frequency depended RT
- o Instant preset switching (no loading times)
- o no compression
- o equalization of multiple filters on inputs and outputs, separately
- o the system does not use any time variant techniques in order to obtain sufficient gain before feedback, so it is fully time constant[1]
- o only a limited number of microphones necessary to pick up the direct sound on stage[2]
- \- possibility for extension microphones for congregational enhancement and/or specific situations where more coverage is needed in the hall for picking up the sources on all desired areas
- o additional line inputs/outputs for theatre sound effects, enhancement of (weak) soloists, monitoring show relay to dressing roomsetc., recordings, filmsurround etc.
- o low noise fan cooled processor, so it can be installed in the sound control room (LAeq< 30dB(A)) without disturbing the work of engineers
- o communication based on TCP/IP, therefore easy to integrate with platforms like AMX or crestron, Supports MADI I/O communication and ready to supports other audio protocols
- o multiple control terminals to be connected to one processor
- o can be used as an acoustic server, i.e. one processor for multiple rooms (simultaneously)
- o infinite amount of presets to be stored
- o easy to use control software (one push on a button)
- o all presets stored as factory presets and user presets
- o live monitoring of all outputs and inputs of the processor available
- o the loudspeaker layout is based on acoustic principles
- o Clipping protection system
- o AFRS: (SIAP Active Failover Redundancy System, switches automatically to another processor in case of failure)
- o dedicated design concepts for e.g. under balcony areas and / or stage areas
- o scalable by modular design
- o future proof (easy software & firmware upgrades)
- o optimization for small, medium and large rooms: proven results

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1 Time variance is used to obtain decorrelation to achieve enough gain before feedback. Most SIAP competitors use this. However SIAP does not need time variance because of its advance proprietary technique for decorrelation. Time variance is undesirable for a number of reasons: 1) Musical instruments which have strings, have their own resonance, as a result of which time variant system reverberation produces a disturbing pitch interference. Piano performances are not possible as the tone starts to move up and down 2) People with absolute pitch hearing cannot listen to the system (mostly musicians and conductors and of course piano tuners) 3) In natural acoustics there is no time variance (except small air movements because of e.g. audience entering the hall, heat from lights.

2 When microphones are placed in the reverberant soundfield, the sound is recirculated. In natural acoustics this is not possible, and therefore gives undesired unnatural side effects. Moreover unwanted sounds in the reverberant field (e.g. coughing, lights etc.) will be processed in the reverberation and result in unnatural effects in the reverberation

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