

**labCORE**

Enhance Your Voice & Audio Quality Measurements

labCORE is the new modular multi-channel front end of the ACQUA lab generation. A wide selection of digital and analog inputs and outputs as well as programmable interfaces make labCORE the all-in-one solution for powerful voice and audio quality measurements. Thanks to its modularity, users can expand labCORE flexibly at any time. Whether hardware or software, for analog or digital measurements: labCORE is versatile and offers a wide range of modules. This allows users to tailor the front end exactly to their individual measuring tasks.

“labCORE sets new standards in the area of measurement technology for voice and audio: Modularity combined with the enormous performance of the front end and the individual modules make it the ideal solution when it comes to testing and further developing voice and audio quality for a wide variety of applications,” says Dr. Hans W. Gierlich, Managing Director of the Telecom division. The currently common interfaces that are relevant for measurements of telecommunication devices are integrated into the front end. A USB host interface and interfaces for I²S (Inter-IC Sound), AES/EBU, ADAT and SPDIF are also available by default. labCORE is designed to be future-proof: Based on modular technology, new technologies can be added quickly and easily.

Another advantage is the connection of the front end to the software. In combination with the upcoming ACQUA version 4.0, users can configure labCORE quickly and easily and can perform automated measurements. With labCORE, users are able to simultaneously transmit up to 32 channels at 48 kHz or up to 8 channels at 192 kHz from PC to labCORE bidirectionally.
Depending on the selected configuration and the respective measurements, no further front ends are required. The user can integrate labCORE into his measuring environment in a space-saving manner. If other front ends are required for certain measurements, labCORE can be easily combined with existing HEAD acoustics front ends.

**Individual module combinations**

Various modules are available to extend the functionality of the labCORE.

**Module I/O Bus Mainboard**

coreBUS is required as bridge between the mainboard and the individual extension modules. Therefore, it is required if users want to extend labCORE with additional modules (up to ten are possible). The bus mainboard is equipped with a Field Programmable Gate Array (FPGA), which allows high data transfer rates.

**Module Power Amplifier**

The module coreOUT-Amp2 adds a two-channel power amplifier to labCORE, typically for the artificial mouth. Each amplifier channel provides 20 Watt RMS per channel (≥ 4 Ohm, Class D). Both channels are provided via a four-pin Speakon socket. The coreOUT-Amp2 module can be added to labCORE twice. Thus, up to four channels are available – four times more than in the ACQUA front end MFE VI.1 and without the need of additional external amplifiers. Test Setups with two artificial head measurement systems, such as for In-Car Communication testing or acoustic-to-acoustic (end-to-end) testing are easy to implement.

**Module Binaural equalization**

labCORE can be extended to a binaural equalizer by the module coreBEQ. Thus, the front end allows the digital equalization of artificial head signals, all in real time.

**Module Microphone Input**

The module coreIN-Mic4 provides four microphone inputs (LEMO 7-pin) for externally polarized microphones. With this module, labCORE enables concurrent connection of the ears of the artificial head measuring system and up to two measuring microphones. Alternatively, users can connect two artificial heads simultaneously. Users choose which inputs should be used via the ACQUA software. Thus, labCORE enables fully automated measurements, as a manual swap of microphones is no longer necessary.

**Modules Analog Inputs and Outputs**

The modules coreIN-A2 resp. coreOUT-A2 provide two-channel analog high-end inputs and outputs each. The connectors can be used in balanced (XLR) and unbalanced (BNC) mode, switchable via the software. The analog inputs support ICP and have a 48 V phantom feeding (switchable per channel). Their THD+N value is -112 dB over wide frequency ranges and the S/N is better than -118 dB. In addition, sampling rates of up to 192 kHz are possible.

**VoIP Modules**

With the upcoming module coreIP, labCORE is able to serve as a VoIP reference gateway. Users can connect VoIP test objects to the front panel interface for measuring the voice quality of digital communication devices and transmission systems. As an option, modules will be available which extend labCORE with important functionalities: Simulation of network impairments even with activated DTX (coreIP-IMP), AMR codec (coreIP-AMR), EVS codec (coreIP-EVS) and two-channel audio codec OPUS (coreIP-OPUS).

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Overview of labCORE modules
(further modules resp. functionalities are in preparation)

labCORE has considerably more processing power due to the improved motherboard. More complex computing operations especially for VoIP codecs, significantly higher data transfer rates or significantly more audio effects in real time: labCORE is ahead of many front ends in the market in terms of performance of voice and audio quality measurements.
The multi-channel front end is designed in such a way that it does not require a fan. It works absolutely noiseless. For mobile use, a HEADlab Power Box can be connected as an external battery. The front panel of the labCORE provides a 2.4 "LCD (320 x 240 pixels) for display purposes. With the help of four soft buttons, the display can be operated easily and intuitively.

Numerous interfaces in the basic configuration

Even the basic configuration of labCORE - without the optional modules presented above - offers numerous interfaces to users that underline the versatility and performance of the front end. Essential measurements can be carried out with these interfaces.

Basic interfaces at the front

- **Analog In/Out**: Two analog input and output connectors each on BNC connectors. The inputs have a typical THD+N of -102 dB and a typical S/N of -112 dB at 48 kHz sampling rate.
- **USB Host**: USB type C, for example, for measuring USB headsets.
- **Headphone**: 6.3 mm headphone jack. Analog 2-channel output for measuring as well as for monitoring purposes (130 mW per channel, ≥ 32 ohms).
- **ADAT/SPDIF/AES**: Optical interface on TOSLINK connector. Output format is selectable.
- **Ethernet**: The interface is prepared on the hardware side so that different VoIP network conditions can be monitored. Supports speeds of up to 1 Gbp/s.

Basic interfaces at the rear

- **Pulse/AES B**: Digital audio input and output at the rear is integrated into a HD-Sub-D 15-pin connector that supports sampling rates of up to 192 kHz. AES switchable to SPDIF.
- **AES A**: One additional AES digital audio input and output (XLR 3-pin) is provided for audio data exchange, e.g. with other front ends. AES switchable to SPDIF.
- **Digital Audio**: Programmable digital audio interface provides four digital inputs and outputs. The interfaces are programmable as, e.g. I²S, PCM, GPI/O for reference clock input and output. The maximum transmission rate of each input and output is 50 Mbits/s.
- **GPIO**: Two additional BNC sockets for general purpose input and output (GPIO) e.g. Pulse, Clocks etc.
- **HEADlink**: For connecting HEADlink devices such as the microphone arrangement MSA I.
3PASS flex and 3PASS lab

Accurate background noise simulation systems comply with ITU-T resp. ETSI standards

With 3PASS lab and 3PASS flex, HEAD acoustics has developed two highly accurate background noise simulation systems. The two playback and recording systems enable manufacturers to equalize any previously recorded sound field, that is representative of the typical use case, at distinctive microphone positions and to reproduce it exactly at these positions. Thus, 3PASS lab and 3PASS flex enables manufacturers to test how well their products perform in real life conditions. The two simulation systems are suitable for different applications.

3PASS flex is particularly suited for testing multi-microphone systems, microphone arrays or beamforming microphones where multi-point noise simulation with flexible microphone and loudspeaker arrays is required. More and more applications offer multi-microphone solutions: from car hands-free systems and In-Car Communication (ICC) systems to televisions, conference systems and even Internet of Things (IoT) devices and home automation systems.

The key features of 3PASS flex: The system...
- ...is highly adjustable: the number and location of microphones and loudspeakers can be freely adjusted in accordance with the rules of the thumb specified in the standards ITU-T P.1100, P.1110, P.1120 as well as P.1140.
- ...offers a fast and consistent, digital and fully automated equalization of the whole system via the user-friendly interface.
- ...ensures a highly accurate spatial sound-field reproduction including exact level and spectral reproduction of the pre-recorded noise field in the relevant microphone positions.

3PASS flex is always based on individual background noises. These recordings can be realized quickly and easily, for example, for mobile use with SQquadriga II.

3PASS flex complies with international standards such as ITU-T Recommendations P.1100, P.1110 and P.1120 (each Annex F), as well as P.1140 (Annex B).

3PASS lab: Sound field reproduction at one location in space

As with 3PASS flex, 3PASS lab is characterized by its excellent attributes regarding preservation and reproduction of the essential spatial characteristics of background noise scenarios in different test rooms. Unlike 3PASS flex, 3PASS lab measures at fixed microphone positions due to the use of the MSA I.

The advantage: The arrangement is always the same and therefore no individual recordings are required. Since 3PASS lab complies with the ETSI standard TS 103 224, users can employ the corresponding ETSI noise database.
3PASS lab is essential for the development of complex background noise reduction algorithms. Thus, especially mobile phone manufacturers can assess the real-life performance of their mobile. This applies in particular for mobile devices with multiple microphones, which is nowadays almost standard.

3PASS lab also offers fully automated digital equalization of the whole system. The lab version is optimized for three use cases: handset, hands-free, and desktop hands-free, especially in conjunction with the artificial head measurement system HMS and the motorized handset positioner HHP IV MotoMount.

Please find additional information to both background noise simulation systems 3PASS flex as well as 3PASS lab on our website: www.head-acoustics.de/eng/telecom_3PASS.htm

HD Voice+: Achieving logo certification with HEAD acoustics measurement equipment

The industry association GSM Association (GSMA) has published another HD Voice version with the HD Voice+ standard. HD Voice+ applies the EVS (Enhanced Voice Services) codec operated in super-wideband or fullband mode: Voice is transmitted at up to 20 kHz instead of 7 kHz, which improves voice quality. If manufacturers of LTE mobile terminals and network providers want to achieve HD Voice + logo certification for their devices, they must fulfill the minimum performance requirements. Due to newly defined requirements regarding e.g. jitter, packet loss and delay, HEAD acoustics’ measurement equipment is perfectly suited for achieving HD Voice+ logo certification for 4G setups.

Perfectly suited software and hardware solutions for appropriate tests

With the advanced communication quality analysis system ACQUA in combination with the measurement front end MFE VIII.1 as well as the software options for the EVS codec (Cod-EVS) and IP network impairments (MFE VIII.1-IMP), HEAD acoustics provides appropriate solutions for testing accordingly. In future, the multi-channel front end labCORE will be available for this purpose. Cod-EVS supports all specified bandwidths from narrowband to fullband and with all bit rates and modes (handset, handheld hands-free and headset). The option MFE VIII.1-IMP enables users to simulate different network impairments, such as delay, jitter, or packet loss, directly at the source of the signal and to apply it to the outgoing IP packets of the MFE VIII.1. The implementation of HEAD acoustics is globally unique: MFE VIII.1-IMP attaches the impairment information to the time signal before encoding. Thus, always the same impairments are applied to the same part of the time signal regardless of the DTX (Discontinuous Transmission) state. This way, network impairments are reproducible even with activated DTX.

The minimum performance requirements defined for HD Voice+ logo certification refer to Release 13 of the two standards TS 26.131 and TS 26.132 of 3GPP standardization body.
VoCAS – Fully automated measurements thanks to individual application solutions from HEAD acoustics

With VoCAS, HEAD acoustics has successfully established an efficient software for assessing speech recognition systems in the market. The fully automated analysis of these systems is an important aspect for manufacturers who rely on Automatic Speech Recognition (ASR) for their applications: from smartphones and tablets through home automation to the automotive sector. Now, HEAD acoustics provides its know-how to create a tailor-made automation solution for VoCAS, depending on the application.

The solutions are exactly tailored to the customer’s requirements and the ASR system to be tested. The automation is implemented via scripts based on the Python programming language. The scripts can easily be integrated into VoCAS. HEAD acoustics has already created completely automated test solutions for popular Automated Speech Recognition systems (e.g. Microsoft Cortana).

In one case study HEAD acoustics worked with an ASR manufacturer that uses a long, continuous reference file with consecutive utterances. During the test, these utterances are played concurrently with background noises. For a better use of the utterances in VoCAS, each sentence is saved in a separate audio file. With that, each statement or file is tagged accordingly. This information, which is essential for the subsequent assessment, is saved in an XML file.

In VoCAS, the user can define flexibly which test sequence should be played. Via attributes such as, e.g. “utterance”, “background noise”, “speaker” including the associated values, the user composes his individual test sequence. By means of serial connection as well as Python scripts, the VoCAS computer is started. Recording as well as playback of background noises begin simultaneously at the test object. This process is repeated for all selected files with utterances. The generated audio files are directly passed on to the VoCAS computer for analysis purposes.

A development tool creates a report into an XML file directly from the XML files and the audio files generated before. The reference utterances as well as the utterances recognized by the ASR system are apparent in the XML report. Via Python scripting, this XML file is imported into VoCAS so that the user can use the comfortable result presentation of the assessment software.

In VoCAS, the user can check fast and easily at which statement, speaker or background noise the test device passed or failed the test.

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HEAD acoustics GmbH establishes subsidiary in China

The newly founded subsidiary HEAD acoustics China Co., Ltd. strengthens the global presence and ensures first-hand service for the strategically important Chinese market. Thus, HEAD acoustics continues on his expansion course and underlines the trend set in the previous years.

“The foundation of our subsidiary in China was the logical consequence of the steadily growing demand for our products and services,” says Prof. Dr. Klaus Genuit, founder and shareholder of HEAD acoustics GmbH. „Customer proximity is key if we want to maintain our position as one of the world’s leading companies in the field of sound quality as well as voice and audio quality.”

Batch Processor series completely revised

Calculation tools offer more functionality and advantages for the user

New design, new features: HEAD acoustics Batch Processor has been fundamentally revised. Now, the efficient stand-alone tool for the automatic calculation of MOS values for 3QUEST, PESQ or POLQA offers users a completely modified user interface – clearer, easier and more intuitive. In addition to the previously available calculations 3QUEST, PESQ, POLQA, EQUEST, SNRI & TNLR, version 2.0.100 also supports TOSQA (ACOPT 10) as well as Speech-based Double Talk (ACOPT 32). With the upcoming version, even the 3QUEST-SWB/FB option will be supported.

The batch files to be used can optionally be in .ini or .txt format. The file import has been simplified: Drag and drop the audio files to be analyzed into the program. The user can now view all current results whenever he wants. In addition, the calculations can be paused and resumed at any time and furthermore the user can start calculations at a specific file and also set a starting point.

The export of results was also revised: In addition to the file formats .txt and .xls, the user can now export the results into an SQLite database and save even more detailed results. In addition, the user can choose which results he wants to export to a file.

Mr. Haifeng (Mike) Gong takes over the management of HEAD acoustics China Co., Ltd., located in the prestigious Shanghai International Ocean Shipping & Finance Center. Mr. Gong has long-standing and cultivated business contacts in China and is an established expert in the industry. He will constantly expand the team, allowing us to operate on a nationwide basis and to provide all the services across the entire country. After a transitional period, Mr. Gong and his team will be the contact for all questions concerning sales, consulting services, service and support as well as the complete product range of HEAD acoustics.
3QUEST-SWB/FB: Calculation method for SWB scenarios
Algorithm is approved as ETSI standard TS 103 281 (Model A)

With 3QUEST-SWB/FB HEAD acoustics now offers a completely new analysis method: 3QUEST, the optional calculation method for the objective speech quality evaluation of telecommunication terminals, has been successfully established in the market for many years and is now also available for super-wideband and fullband scenarios (audio in the frequency range up to 20 kHz). The 3QUEST-SWB/FB algorithm calculates the three MOS values (Mean Opinion Scores) S-MOS, N-MOS and G-MOS.

The calculation method finally enables manufacturers of mobile phones or hands-free kits, for example, to make a well-founded statement about the voice quality in the context of super-wideband and fullband, including background noises. 3QUEST-SWB/FB considers the influence of background noises in sending direction in the calculation and therefore offers a decisive advantage compared to other objective evaluation models such as TOSQA, PESQ or POLQA. Another advantage of 3QUEST-SWB/FB: The method no longer requires an unprocessed reference signal. In addition, the new calculation method is even more robust against unknown data than its predecessors, which are based on the standards ETSI EG 202 396-3 and TS 103 106.

3QUEST-SWB/FB was developed by HEAD acoustics in close collaboration with partners from the standardization as well as with the bodies ETSI and 3GPP. For this purpose, extensive listening tests with several hundred test persons have been conducted since 2015. An independent test laboratory has validated the calculation method. Since April 2017, the algorithm is approved as ETSI standard TS 103 281 (Model A). 3QUEST-SWB/FB is available for the communication quality analysis system ACQUA as well as for the stand-alone tool Batch Processor.

Did you know that..?
• ...the measurement standard “Skype” is the HEAD acoustics implementation of the „Skype for Business Audio Test Specification“ (Version 3.0, December 2016).
• ...the standard comprises tests for Skype™ for Business devices and accessories (e.g. headsets, headsets, hands-free speakerphones as well as conferencing devices).
• ...the implemented tests cover important aspects such as delay measurements in sending and receiving direction, echo tests, voice quality in the presence of background noise as well as during double talk.
• ...the ACQUA standard contains three different test suites which are applied depending on the test object: 1) if the device is using the built-in signal processing of Skype™ for Business, 2) if the test object is a personal device that has its own signal processing and 3) if the device is a USB speakerphone, conference system etc.

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